

INFORMATION DISCLOSURE STATEMENT BY APPLICANT (Not for submission under 37 CFR 1.99)	Application Number	13864249
	Filing Date	2013-04-17
	First Named Inventor	MILLINGTON; Nicholas A. J.
	Art Unit	2442
	Examiner Name	SURVILLO, Oleg
	Attorney Docket Number	13-0402 (MBHB 15-518-CON)

33	Microsoft Window's XP Reviewer's Guide (August 2001) (D+M_0402225-85) (61 pages)
34	Simple Network Time Protocol (SNTP), RFC 1361 (August 1992) (D+M_0397537-46) (10 pages)
35	Simple Network Time Protocol (SNTP), RFC 1769 (March 1995) (D+M_0397663-76) (14 pages)
36	Realtime Transport Protocol (RTP), RFC 3550 (July 2003) (D+M_0398235-323) (89 pages)
37	Real Time Control Protocol (RTCP) and Realtime Transfer Protocol (RTP), RFC 1889 (Jan. 1996) (D+M_0397810-84) (75 pages)
38	Realtime Streaming Protocol (RTSP), RFC 2326 (Apr. 1998) (D+M_0397945-8036) (92 pages)
39	Network Time Protocol (NTP), RFC 1305 (Mar. 1992) (D+M_0397417-536) (120 pages)
40	UPnP Design by Example, A Software Developers Guide to Universal Plug and Play Michael Jeronimo and Jack Weast, Intel Press (D+M_0401307-818) (April 2003) (511 pages)
41	Sonos, Inc. v. D&M Holdings (No. 14-1330-RGA), DI 226, Opinion Denying Inequitable Conduct Defenses (Feb. 6, 2017) (5 pages)
42	Sonos, Inc. v. D&M Holdings (No. 14-1330-RGA), DI 227, Order Denying Inequitable Conduct Defenses (Feb. 6, 2017) (1 page)
43	Sonos, Inc. v. D&M Holdings Inc. et al., Defendants' Opposition to Sonos's Motion to Strike Defendants' New Amended Answer Submitted with their Reply, provided October 3, 2016, 15 pages.

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44	Sonos, Inc. v. D&M Holdings Inc. et al., Exhibit A: Defendants' First Amended Answer to Plaintiffs' Third Amended Complaint, provided August 1, 2016, 26 pages.
45	Sonos, Inc. v. D&M Holdings Inc. et al., Exhibit A: Defendants' Second Amended Answer to Plaintiffs' Third Amended Complaint, provided October 12, 2016, 43 pages.
46	Sonos, Inc. v. D&M Holdings Inc. et al., Exhibit A: Defendants' Second Amended Answer to Plaintiffs' Third Amended Complaint, provided September 9, 2016, 88 pages.
47	Sonos, Inc. v. D&M Holdings Inc. et al., Exhibit B: Defendants' Second Amended Answer to Plaintiffs' Third Amended Complaint, provided October 12, 2016, 43 pages.
48	Sonos, Inc. v. D&M Holdings Inc. et al., Opening Brief in Support of Defendants' Motion for Leave to Amend Their Answer to Add the Defense of Inequitable Conduct, provided August 1, 2016, 11 pages.
49	Sonos, Inc. v. D&M Holdings Inc. et al., Order, provided October 7, 2016, 2 pages.
50	Sonos, Inc. v. D&M Holdings Inc. et al., Plaintiff's Opposition to Defendants' Motion for Leave to Amend Their Answer to Add the Defense of Inequitable Conduct, provided August 26, 2016, 25 pages.

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- Dawidowsky, Frank,
c/o Sony International (Europe)
70327 Stuttgart (DE)
- Hoefflinger, Jens,
c/o Sony International (Europe)
70327 Stuttgart (DE)

(71) Applicant: **Sony International (Europe) GmbH**
10785 Berlin (DE)

(74) Representative: **Müller - Hoffmann & Partner**
Patentanwälte,
Innere Wiener Strasse 17
81667 München (DE)

(72) Inventors:
• **Stadelmeier, Lothar,**
c/o Sony International(Europe
70327 Stuttgart (DE)

Remarks:
Amended claims in accordance with Rule 86 (2)
EPC.

(54) **Bandwidth oriented reconfiguration of wireless ad hoc networks**

(57) A first wireless ad hoc network with a certain available amount of bandwidth comprises a plurality of wireless terminals communicating via wireless connections with each other. Each wireless connection requires a certain amount of the available amount of bandwidth. A problem occurs, if a wireless terminal wants to set up a new wireless connection requiring bandwidth or requires more bandwidth for an existing wireless connection and the required amount of bandwidth exceeds the left amount of the available amount of bandwidth of the first wireless ad hoc network. In a state of the art wireless ad hoc network the requested bandwidth cannot be provided and a user application may not be possible. According to the invention, however, the required amount of bandwidth can be provided by splitting the first wireless ad hoc network such that a second wireless ad hoc network is spawned which operates in a different communication channel and thus provides additional bandwidth. The second wireless ad hoc network comprises wireless terminals of the first wireless ad hoc network and/or new wireless terminals.

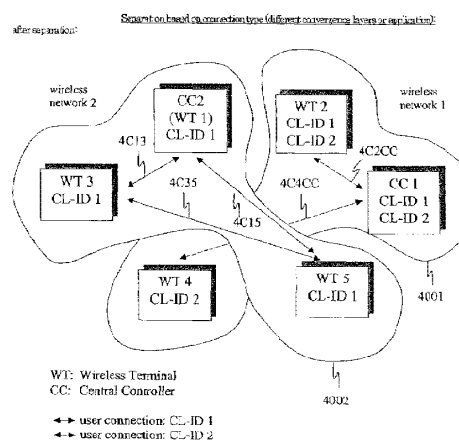


Figure 4B

Description

[0001] The invention is related to wireless ad hoc networks which comprise a central controller and a set of wireless terminals (also referred to as wireless devices). In particular, the invention is related to a method and means for providing additional bandwidth for a wireless ad hoc network if the available bandwidth is not sufficient.

[0002] A wireless ad hoc network is a collection of autonomous nodes or wireless terminals that communicate with each other by forming a radio network and maintaining connectivity in a decentralized manner. Since the nodes communicate over wireless connections, they have to contend with the effects of radio communication, such as noise, fading, and interference. In addition, the connections typically have less bandwidth than in a wired network. The network topology is in general dynamic, because the connectivity among the nodes may vary with time due to node departures, new node arrivals, and the possibility of having mobile nodes.

[0003] For a variety of home multimedia applications and business applications, it is important to establish networks, preferably wireless networks, for exchanging data and messages between different devices that are part of the network. In a typical business application scenario, a mobile terminal gets services over a fixed corporate or public infrastructure. In an exemplary home application scenario, a low-cost and flexible networking is supported to interconnect wireless digital consumer devices. Such devices can for example be multimedia devices which typically require high bit rates and are therefore using a lot of the available amount of bandwidth of a wireless ad hoc network.

[0004] The ETSI (European Telecommunications Standards Institute) Project BRAN (Broadband Radio Access Networks) has defined the standard HIPERLAN/2 (High Performance Radio Local Area Network), which provides high-speed multimedia communications between different broadband core networks and wireless (mobile) terminals. HIPERLAN/2 provides a flexible platform for a variety of business and home applications that can support a set of bit rates up to 54 Mbit/s. The HIPERLAN/2 standard is an example how data can be transmitted between different devices in a wireless ad hoc network. Another standardization body for wireless ad hoc networks is IEEE 802.11. The invention is not limited to wireless networks according to the HIPERLAN/2 and IEEE 802.11 standard, though.

[0005] A problem occurs within a state of the art wireless ad hoc network, if a wireless terminal requests more bandwidth than the left amount of bandwidth of the wireless ad hoc network. The terminal may request more bandwidth e.g. because it wants to setup a new wireless connection or requires more bandwidth for an existing wireless connection with another wireless terminal of the network. More bandwidth may also be required if a

new wireless terminal wants to join the wireless ad hoc network and wants to set up wireless connections. In such a case, if the wireless ad hoc network runs according to the above mentioned HIPERLAN/2 standard or IEEE 802.11 standard, a wireless terminal may not be provided with the requested amount of bandwidth. Therefore, a certain user application may not be possible.

[0006] Fig. 1 shows a state of the art wireless ad hoc network 100 comprising three wireless terminal-devices, i.e. a first wireless terminal-device WT10, a second wireless terminal-device WT20, and a third wireless terminal-device WT30, and a terminal-device central controller CC100 that controls the wireless terminal-devices. Further, there are three wireless device-connections, a first wireless device-connection C23 between the second wireless terminal-device WT20 and the third wireless terminal-device WT30, a second wireless device-connection C31 between the first wireless terminal-device WT10 and the third wireless terminal-device WT30 and a third device-connection C1CC between the first wireless terminal-device WT10 and the terminal-device central controller CC100. The wireless device-connections require a certain amount of the available bandwidth of the wireless ad hoc network 100. In the example the first wireless device-terminal WT10 wants to establish a new wireless device-connection C21 with the second wireless terminal-device WT20. However, the required bandwidth for this new wireless device-connection C21 exceeds the left amount of bandwidth of the total available bandwidth of the wireless ad hoc network. Therefore, in the state of the art wireless ad hoc network 100, if the network is operated according to HIPERLAN/2 standard, the setup of the new wireless device-connection C21 fails, or, if the network is operated according to IEEE 802.11 standard, an application on the new wireless device-connection C21 is not possible. A similar situation would occur if more bandwidth would be required for an existing wireless device-connection and the required amount of bandwidth exceeds the left available bandwidth of the wireless ad hoc network. If for example the second wireless terminal-device WT20 requires more bandwidth for the first wireless device-connection C23 and the required amount of bandwidth exceeds the left amount of the total available bandwidth, an application may not be possible. Moreover, data packets transmitted within the wireless ad hoc network may get lost.

[0007] Therefore, it is an object of the invention to provide a method and means for providing additional bandwidth for a wireless ad hoc network in case the amount of bandwidth of the wireless ad hoc network is not sufficient to establish new wireless connections or to provide more bandwidth for existing wireless connections.

[0008] The object of the invention is solved by a method for providing additional bandwidth for a wireless ad hoc network according to claim 1, by a wireless terminal according to claim 7, by a central controller according

to claim 9, and by a wireless ad hoc network according to claim 11. Preferred embodiments thereof are respectively defined in the respective following subclaims.

[0009] The method to provide additional bandwidth for a wireless ad hoc network comprising a plurality of wireless terminals operating in a certain communication channel (e.g. a RF channel at a certain frequency, an IR or optic channel) with a certain amount of available bandwidth according to the present invention comprises a step of splitting said wireless ad hoc network such that at least one new wireless ad hoc network is spawned, if more bandwidth than said certain amount of available bandwidth is required by said plurality of wireless terminals, wherein after the split of said wireless ad hoc network at least one wireless terminal of said wireless ad hoc network and/or one or more new wireless terminals belong(s) to said at least one new wireless ad hoc network, and said at least one new wireless ad hoc network is operating in a respective different communication channel.

[0010] In other words, if a first wireless ad hoc network requires more bandwidth, a second (new) wireless ad hoc network is spawned (generated). This second wireless ad hoc network is operated in a different communication channel than the first wireless ad hoc network, therefore providing additional bandwidth. Wireless terminals of the first wireless ad hoc network or new wireless terminals may be moved to the second wireless ad hoc network. It may also be possible, that new wireless terminals are moved to the first wireless ad hoc network and wireless terminals of the first wireless ad hoc network are moved to the second wireless ad hoc network. The splitting of a wireless ad hoc network (spawning of a new wireless ad hoc network) can also be seen as a bandwidth oriented reconfiguration of a wireless ad hoc network.

[0011] The wireless terminal of a wireless ad hoc network controlled by a central controller of said wireless ad hoc network according to the present invention comprises a receiving means adapted to receive a requesting command (SPAWN_NETWORK) from the central controller of the wireless ad hoc network, i.e. the first wireless ad hoc network, indicating certain operating conditions for the wireless terminal, a condition checking means to check if the wireless terminal can be operated under said certain conditions, and a sending means that sends out a confirmation command (SPAWN_NETWORK_ACK), if the condition checking means determines that the wireless terminal can be operated under said certain conditions, i.e. in the new (second) wireless ad hoc network.

[0012] The central controller of a (first) wireless ad hoc network comprising a plurality of wireless terminals according to the present invention comprises a splitting means that controls the splitting of said (first) wireless ad hoc network, wherein a new (second) wireless ad hoc network is spawned that comprises at least one of said plurality of wireless terminals and/or one or more new

wireless terminals.

[0013] The wireless ad hoc network according to the present invention comprises a plurality of wireless terminals according to the present invention and a central controller according to the present invention.

[0014] Preferably, said (first) wireless ad hoc network and the splitting of said wireless ad hoc network are controlled by a central controller of said (first) wireless ad hoc network that decides which wireless terminals of said (first) wireless ad hoc network and/or which new wireless terminals are moved to said at least one new wireless ad hoc network (second wireless ad hoc network), wherein the decision is based on certain separation criteria, and said central controller determines a new central controller for said at least one new wireless ad hoc network (second wireless ad hoc network).

[0015] Further, said (first) wireless ad hoc network and said at least one new wireless ad hoc network (second wireless ad hoc network) might be operated according to the IEEE802.11 or ETSI BRAN HIPERLAN/2 standard.

[0016] It is further advantageous, if said certain separation criteria assure that wireless terminals that have the same convergence layers, and/or are of the same application layer, and/or have the same connection, and/or provide the same or corresponding functions are not separated into different wireless ad hoc networks.

[0017] Preferably, said certain separation criteria further assure that wireless terminals with certain connections that should not be interrupted are not moved to said at least one new wireless ad hoc network.

[0018] Further, the invention provides advantageous new commands in order to spawn said at least one new wireless ad hoc network (second wireless ad hoc network), i.e. a requesting command (SPAWN_NETWORK) that is sent to a request wireless terminal to ask this request wireless terminal to move to said at least one new ad hoc wireless network (second wireless ad hoc network), and a confirmation command (SPAWN_NETWORK_ACK) that is used by a request wireless terminal to signal that it can move to said at least one new ad hoc wireless network (second wireless ad hoc network).

[0019] Preferably, said certain conditions define if said request wireless terminal can operate as a central controller of a wireless ad hoc network, if said request wireless terminal can operate in a certain communication channel, and a moment in time at which said request wireless terminal should operate in said certain communication channel.

[0020] Further, advantageously a wireless terminal stops using its entire wireless connections (radio link control - RLC) the moment it sent out said confirmation command (SPAWN_NETWORK_ACK), moves to one of said at least one new wireless ad hoc network, waits until it receives a start command (RLC_CC_START_OPERATION) sent out by a central controller, and then starts using its wireless connections according to the information

(e.g. time stamp) provided by said start command (RLC_CC_START_OPERATION). It may also be possible that within said requesting command (SPAWN_NETWORK) a time stamp is transmitted and a wireless terminal stops its entire connections at the moment in time indicated by this time stamp. Thus, all wireless terminals that are moved to said at least one new wireless ad hoc network (second ad hoc wireless network) stop their radio link control at the same time.

[0021] The invention and advantageous details thereof will be explained by way of an exemplary embodiment thereof in the following with reference to the accompanying drawings in which

Fig. 1 shows an example of a state of the art wireless ad hoc network where the setup of a new connection fails because there is not enough available bandwidth to establish the connection;

Fig. 2A shows an example of a wireless ad hoc network with different user connections wherein a wireless terminal requires more bandwidth and a wireless terminal wants to join the network;

Fig. 2B shows the splitting based on application level of the wireless ad hoc network of Fig. 2A according to the invention to provide more bandwidth;

Fig. 3 shows a flow chart illustrating the process of splitting a wireless ad hoc network;

Fig. 4A shows an example of a wireless ad hoc network with wireless connections with different convergence layers, i.e. different connection types, wherein a wireless terminal requires more bandwidth to set up a new wireless connection;

Fig. 4B shows the splitting based on connection type of the wireless ad hoc network of Fig. 4A according to the invention to provide more bandwidth; and

Fig. 5 shows a message sequence chart (MSC) for the ETSI BRAN HIPERLAN/2 standard that describes the process of splitting the wireless ad hoc network of Fig. 4A and 4B, wherein new commands according to the invention are used.

[0022] Figs. 2A and 2B show an example of a wireless ad hoc network according to the invention. In Fig. 2A, the first original wireless network 200 comprises four wireless terminals, i.e. a first wireless terminal WT1, a second wireless terminal WT2, a third wireless terminal

WT3, and a fourth wireless terminal WT4, and a central controller CC. There are three wireless connections (user connections), i.e. a first wireless connection 2C13 (connecting the first wireless terminal WT1 and the third wireless terminal WT3), a second wireless connection 2C12 (connecting the first wireless terminal WT1 and the second wireless terminal WT2), and a third wireless connection 2C4CC (connecting the fourth wireless terminal WT4 and the central controller CC). In the example, the second wireless terminal WT2 requests the setup of a fourth wireless connection 2C23 with the third wireless terminal WT3 requiring a certain amount of bandwidth. Further, a fifth wireless terminal WT5 wants to join the first original wireless network 200 requesting a fifth wireless connection 2C35 with the third wireless terminal WT3 requiring a certain amount bandwidth. The central controller CC therefore checks, if all conditions for establishing the new connections, i.e. the fourth wireless connection 2C23 and the fifth wireless connection 2C35, are met, i.e. especially the central controller CC checks, if there is enough bandwidth available. In the example, there is not enough bandwidth available to establish the new connections. If the network would run according to state of the art HIPERLAN/2 standard the required connections could not be established, or if the network would run according to IEEE 802.11 standard a desired user application would not be possible.

[0023] As can be seen in Fig. 2B, according to the invention, the first original wireless ad hoc network 200 is split up into two wireless ad hoc networks such that a first new wireless ad hoc network is spawned. In Fig. 2B, wireless ad hoc network 2001 corresponds to the first original wireless ad hoc network 200 of Fig. 2A, further, a first new wireless ad hoc network 2002 can be seen. Each wireless ad hoc network operates in a different RF channel. It should be mentioned, that the invention is not limited to RF channels. It may be used for a variety of different communication channels, e.g. IR or other optic channels. If the RF channels of the wireless ad hoc network 2001 and the first new wireless ad hoc network 2002 have the same available amount of bandwidth, the total amount of bandwidth available for the wireless terminals is doubled in comparison to the first original wireless ad hoc network 200 of Fig. 2A. Here, the wireless ad hoc network 2001 operates in the same RF channel as the wireless ad hoc network 200. However, it is also possible, that the RF channel of wireless ad hoc network 2001 is changed, wherein the channel is different from that of the first new wireless ad hoc network 2002.

[0024] As can be seen in Fig. 2B, each wireless ad hoc network, i.e. wireless network 2001 and the first new wireless ad hoc network 2002, has a central controller. Wireless ad hoc network 2001 has a first central controller CC1 and wireless ad hoc network 2002 has a second central controller CC2. The first central controller CC1 is the original central controller CC of the first original wireless ad hoc network 200 of Fig. 2A. The second

central controller CC2 is the original second wireless terminal WT2 of Fig. 2A, i.e. the second wireless terminal WT2 now has additional controlling functions to function as the second central controller CC2 of the first new wireless ad hoc network 2002.

[0025] A wireless terminal can function as a central controller. In this case the wireless terminal performs additional controlling functions. That means, a central controller can be seen as a wireless terminal with additional controlling functions. Controlling functions include the management of the network and its connections, for example the association of new wireless terminals. However, there may also be some wireless terminals that are not capable of performing additional controlling functions, i.e. they cannot be used as central controllers. If a network is split up the new central controller of the new wireless ad hoc network can therefore only be a wireless terminal with additional controlling functions. The original central controller can assign the new central controller, since the capabilities of the different wireless terminals with respect to the controlling functions are exchanged during the association process, i.e. the login of a wireless terminal to a network.

[0026] In Fig. 2B wireless ad hoc network 2001 comprises the fourth wireless terminal WT4 and the second central controller CC2. The second central controller CC2 is the central controller CC of the first original wireless ad hoc network 200 of Fig. 2A. Further, the wireless network 2001 comprises the third wireless connection 2C4CC between the fourth wireless terminal WT4 and the second central controller CC2.

[0027] In the following description, if the reference symbol of a wireless terminal, central controller, and/or a connection changes due to a split of a network, the corresponding reference symbol before the split will be given in parenthesis, e.g. the second central controller CC2 in Fig. 2B will be written as second central controller CC2 (CC).

[0028] In Fig. 2B the first new wireless ad hoc network 2002 comprises the first wireless terminal WT1, the third wireless terminal WT3, the fifth wireless terminal WT5 and the second central controller CC2 (WT2), which is the second wireless terminal WT2 of the first original wireless network 200 of Fig. 2A now functioning as a central controller. Further, the first new wireless network 2002 comprises the first wireless connection 2C13, the second wireless connection 2C12, the fourth wireless connection 2C23, and the fifth wireless connection 2C35. The fifth wireless connection 2C35 between the third wireless terminal WT3 and the fifth WT5 and the fourth wireless connection 2C23 between the second wireless terminal WT2 and the third wireless terminal WT3 are the requested connections that can now be established because there is now enough bandwidth available due to the splitting of the first original wireless network 200 of Fig. 2A.

[0029] In the example of Fig. 2B the criteria used for separating (splitting), i.e. the decision which wireless

terminals are placed in which wireless network, is based on the functionality of the different wireless terminals (functional or application level based separation). Examples for different functionalities of wireless terminals are video communication and audio communication on the one hand, and data transfer (PC to PC data communication) on the other hand. In the example of Figs. 2A and 2B, the first wireless terminal WT1, the second wireless terminal WT2, the third wireless terminal WT3 and the fifth wireless terminal WT5 are PC-type devices. The fourth wireless terminal WT4 and the first central controller CC1 are video and audio devices (VCR, TV, loudspeaker, ...). Therefore, the video and audio devices, i.e. the fourth wireless terminal WT4 and the first central controller CC1 are left in wireless ad hoc network 2001 and the PC-type devices, i.e. the first wireless terminal WT1, the second wireless terminal WT2, the third wireless terminal WT3 and the fifth wireless terminal WT5 are put into the first new wireless ad hoc network 2002. There is no connection between wireless ad hoc network 2001 and the first new wireless ad hoc network 2002.

[0030] There may be other criteria according to which wireless terminals should be separated during the splitting process. Examples for such criteria are:

- Connection type (supported convergence layers). In HIPERLAN/2 it is possible to have multiple convergence layers (e.g. Ethernet, IEEE 1394) in parallel on top of one wireless connection. It may be important not to separate wireless terminals connected by a wireless connection with the same convergence layer.
- Application layer specific criteria. In IEEE 802.11 and HIPERLAN/2 standard, an application layer may have specific requirements, e.g. communication on specific port numbers, applications, or protocols. These may be criteria to perform network separation, e.g. wireless terminals with application layers communicating on the same port number should not be separated.
- Communication on the same connection. It may be important not to separate wireless terminals that are communicating on the same connection(s).
- Functionality of devices (e.g. in IEEE 1394: Video devices, Audio devices, ...). It may be important that devices that can potentially communicate with each other are not separated. For example a video outputting device should maybe not be separated from a video reproducing device (cf. example of Fig. 2B).

[0031] Another separation criterion may be the interruption of connections. During the move of a wireless terminal to a new (spawned) wireless ad hoc network, the transfer of data may not be possible for a short mo-

ment. If the wireless terminal has an application requiring a certain quality of service, e.g. for a real time application, it should therefore eventually not be moved to the new wireless ad hoc network. If such a wireless terminal stays in the original wireless ad hoc network, the data transfer may not be interrupted, i.e. the quality of service can be assured even during a split of a network.

[0032] Fig. 3 shows a flow chart illustrating the process of splitting a wireless ad hoc network. In the original state S301 a wireless ad hoc network is established. In a following step S302 a wireless device (wireless terminal, WT) wants to set up a new connection. The wireless device therefore informs the central controller of the wireless ad hoc network about this desire to set up a new connection.

[0033] In the next step S303 the wireless terminal that wants to set up a new connection gets the feedback from the central controller that the left amount of bandwidth is insufficient to establish the new connection. The central controller therefore checks thereafter in step S304 if another channel is available. If this is not the case, the first final state S311 is reached which indicates that the application is not possible.

[0034] If another channel is available, in step S305, the central controller checks if a network split is possible. Therefore, it is checked which devices of the network need to remain in the same wireless ad hoc network and which can be moved to the new wireless network. This means the above mentioned criteria are checked and two sets of wireless terminals are formed such that wireless terminals that cannot be separated according to the respective criteria are kept within the same set, i.e. the same wireless network. It is then determined which wireless terminals are separated from the wireless ad hoc network to form a new wireless ad hoc network. (In the following these wireless terminals are referred to as splitting wireless terminals). If a network split is not possible, the first final state S311 is reached, i.e. the application is not possible. If, however, in step S305, the central controller determines that a network split is possible, in step S306, all splitting wireless terminals are informed about the split. The splitting wireless terminals are provided with the following information:

- Each splitting wireless terminal is informed whether its role in the new wireless ad hoc network is the role of a regular wireless terminal or the role of the central controller of the new network.
- The splitting devices are also informed about the splitting time, i.e. the moment in time at which the split of the network will be performed.
- Further, the RF frequency of the new wireless ad hoc network is transmitted to the splitting device such that a splitting device may restart its wireless connections after the split at the indicated frequency.

In a following step S307 the radio link control (RLC) of all affected devices, i.e. all splitting devices, is stopped.

[0035] Thereafter, in step S308 the central controller then transfers all connection information to the central controller of the new wireless ad hoc network, and the new central controller of the new wireless ad hoc network activates the RLC of all devices in the new network at the splitting time in step S309. In a second final state S310 there then exists a new wireless ad hoc network under control of the new central controller. In parallel there exists a wireless network (original wireless ad hoc network) that comprises all wireless terminals that have not been moved to the new wireless ad hoc network and the original central controller. It may also comprise new wireless terminals.

[0036] Fig. 4A and 4B show another example of a wireless ad hoc network that is split up according to the invention. Here, the separation of wireless terminals is based on the connection type of the different wireless terminals, i.e. wireless terminals with the same connection type are preferably not separated into different ad hoc networks. In the example here, a connection type refers to a convergence layer in the sense of the HIPERLAN/2 standard. This means, wireless terminals with the same convergence layers are not separated. As mentioned above in HIPERLAN/2 networks, it is possible to have different convergence layers (CL) on top of the data link control (DLC) layers and physical (PHY) layers. Specific convergence layers can be for example Ethernet, IEEE 1394, IP, ATM, or UMTS. This makes HIPERLAN/2 a multi-network air interface. There may be exceptions, i.e. it may be possible that wireless terminals with the same connection type are admitted to be separated. An exception is made, if the wireless terminals are not communicating with each other, i.e. they have no connections with each other.

[0037] Fig. 4A shows a second original wireless ad hoc network 400 before the separation according to the invention. It comprises the first wireless terminal WT1, the second wireless terminal WT2, the third wireless terminal WT3, the fourth wireless terminal WT4, and the fifth wireless terminal WT5, and the central controller CC. Further, it comprises a sixth wireless connection 4C13, a seventh wireless connection 4C2CC, an eighth wireless connection 4C35, and a ninth wireless connection 4C4CC. The first wireless terminal WT1, the third wireless terminal WT3, and the fifth wireless terminal WT5 have wireless connections with the same first convergence layer CL-ID1, i.e. the sixth wireless connection 4C13 and the eighth wireless connection 4C35 have the first convergence layer CL-ID1. The fourth wireless terminal WT4 has a wireless connection with a different convergence layer, i.e. a second convergence layer CL-ID2. The second wireless terminal WT2 and the central controller CC have the seventh wireless connection 4C2CC with both convergence layers, i.e. the first con-

vergence layer CL-ID1 and the second convergence layer CL-ID2. Now, the first wireless terminal WT1 wants to setup a tenth wireless connection 4C15 with the first convergence layer CL-ID1 with the fifth wireless terminal WT5 requiring a certain amount of bandwidth. However, the left amount of bandwidth within the wireless network is not sufficient. Therefore, according to the invention, the network is split up, i.e. a new wireless ad hoc network is spawned. It should be mentioned again, that if the network would run according to state of the art HIPERLAN/2 standard the setup of the tenth wireless connection 4C15 would fail, or if the network would run according to IEEE802.11 standard, an application would not be possible.,.

[0038] The central controller CC now determines, which wireless terminals are moved to the second new wireless ad hoc network and which remain in the second original network. The two separated wireless networks can be seen in Fig. 4B. The wireless ad hoc network 4001 in Fig. 4B corresponds to the second original wireless ad hoc network 400 of Fig. 4A, further a second new wireless ad hoc network 4002 is spawned during the splitting process. As mentioned above, in the example of Fig. 4A and 4B, the separation of wireless terminals is based on connection type, i.e. on the respective convergence layers. Since the first wireless terminal WT1, the third wireless terminal WT3, and the fifth wireless terminal WT5 are communicating over wireless connections with the same first convergence layer CL-ID1, these wireless terminals are not separated. Further, the second wireless terminal WT2, the fourth wireless terminal WT4 and the central controller CC have connections with each other with the second convergence layer CL-ID2 and are therefore not separated. As mentioned above, the second wireless terminal WT2 and the central controller CC additionally have the second convergence layer CL-ID1 on top of the seventh wireless connection 4C2CC. However, they have no connection with the first wireless terminal WT1, the third wireless terminal WT3, and the fifth wireless terminal WT5, and can therefore be separated from these wireless terminals.

[0039] As can be seen in Fig. 4B, the tenth wireless connection 4C15 between the fifth wireless terminal WT5 and the second new central controller CC2 (WT1) could now be established because of the split of the network there is now enough bandwidth available. The second new central controller CC2 is the original first wireless terminal WT1 of the second original wireless network in Fig. 4A.

[0040] Fig. 5 shows a message sequence chart (MSC) for the ETSI BRAN HIPERLAN/2 standard for the separation (splitting) process according to the invention. The MSC in Fig. 5 describes the separation process of the wireless ad hoc network in Figs. 4A and 4B. The MSC in Fig. 5 shows the first wireless terminal WT1, the second wireless terminal WT2, the third wireless terminal WT3, the fourth wireless terminal WT4, the fifth wire-

less terminal WT5, and the central controller CC of the second original wireless ad hoc network 400 of Fig. 4A.

[0041] In the initial state S501 the first wireless terminal WT1, the second wireless terminal WT2, the third wireless terminal WT3, the fourth wireless terminal WT4, and the fifth wireless terminal WT5 are associated to the central controller CC as shown in Fig. 4A. As mentioned above the first wireless terminal WT1 now requires more bandwidth to set up the tenth wireless connection 4C15 with the fifth wireless terminal WT5. Therefore, in step S502, the first wireless terminal WT1 sends the standard HIPERLAN/2 message "RLC_Setup" to the central controller CC indicating the required setup of the new connection requiring more bandwidth. In the following step S503, the central controller CC sends back the HIPERLAN/2 standard command "RLC_Release: lack of resources" to the first wireless terminal WT1 indicating that no bandwidth is available to set up a new connection with the required amount of bandwidth. The lack of bandwidth causes the central controller CC to check if parts of the network can be separated to a new network. This means the above mentioned criteria are checked. In the present example the criteria for the separation are the convergence layers of the wireless terminals associated to the central controller CC. As explained above with reference to the Figs. 4A and 4B, the first wireless terminal WT1, the third wireless terminal WT3, and the fifth wireless terminal WT5 have the same convergence layers and are therefore not separated and the second wireless terminal WT2, the fourth wireless terminal WT4, and the central controller CC are not separated.

[0042] Once, the central controller CC has determined that a split is possible and which wireless terminals are moved to the new wireless ad hoc networks (splitting wireless terminals), the central controller CC sends a command "Spawn_Network" to the splitting wireless terminals, here, the first wireless terminal WT1, the third wireless terminal WT3, and the fifth wireless terminal WT5. The command "Spwan_Network" is defined as follows:

SPAWN NETWORK

[0043] functionality: a central controller (CC) informs wireless terminals (WTs) to change to a new wireless network parameters:

channel	center frequency of the new network
CC_or_MT	defines role of the device in the new network (wireless terminal or new central controller)
start_mac_frame	defines point of time when devices belong to new network and are controlled by the new central controller CC.

[0044] A wireless terminal that receives a command "Spawn_Network" from a central controller sends back a command "Spawn_Network_ack" to the central controller indicating whether it can change to the new network or not. The command "Spawn_Network_ack" is defined as follows:

SPAWN_NETWORK_ACK

[0045] functionality: wireless terminals acknowledge SPAWN_NETWORK command parameters:

ACCEPT_ACTION indicates if the wireless terminal can change to the new network or rejects the separation. The rejection lets the central controller look for new separation possibilities or cancel the separation process.

[0046] As soon as a command "Spawn_Network_ack" is sent out by a wireless terminal indicating that the wireless terminal can change to a new network (the parameter **ACCEPT_ACTION** is set accordingly), the wireless terminal stops its radio link control (RLC). When the time indicated by the command "start_mac_frame" arrives, the wireless terminal moves to the new network, i.e. it sets all its parameters (e.g. frequency) according to the values indicated by the command "SPAWN_NETWORK". It then waits for a wake-up command "RLC_CC_START_OPERATION" sent out by the new central controller of the new wireless ad hoc network to actually start its RLC.

[0047] Now, in the example of Fig. 5, in step S504, the command "Spawn_Network" is sent to the first wireless terminal WT1. Since the first wireless terminal WT1 will become the second new central controller CC2, it is the designated central controller CC-CAND (central controller candidate) of the new wireless ad hoc network. Therefore, the parameter "CC_or_MT" of the command "Spawn_Network" indicates that the first wireless terminal WT1 becomes a central controller. In the following step S505 the first wireless terminal WT1 sends back the command "Spawn_Network_ack" to the central controller CC indicating that it can change to the new network and can be the central controller of the new network. Then, in step S506, the central controller sends the command "Spawn_Network" to the third wireless terminal WT3. Within this Spawn_Network command the parameter "CC_or_MT" indicates that the third wireless terminal WT3 remains a wireless terminal, i.e. does not become a central controller. Then, in step S507, the third wireless terminal WT3 sends back a command "Spawn_Network_ack" to the central controller CC indicating that it can change to the new network. In the following step, i.e. step S508, the central controller CC sends the command "Spawn_Network" to the fifth wireless terminal WT5. The parameter "CC_or_MT" again indicates that the fifth wireless terminal WT5 remains a

wireless terminal in the new network. The fifth wireless terminal WT5 sends back the command "Spawn_Network_ack" in step S509 to the central controller CC indicating that it can change to the new network.

[0048] As mentioned above, when a wireless terminal sends out a command "Spawn_Network_ack" it stops its RLC. Therefore, in the respective steps S505, S507, and S509 the first wireless terminal WT1, the third wireless terminal WT3, and the fifth wireless terminal WT5 stopped their RLC. Therefore, in the internal state S510 the RLC of the affected devices, i.e. the first wireless terminal WT1, the third wireless terminal WT3, and the fifth wireless terminal WT5 are stopped.

[0049] Thereafter, in step S511, the central controller CC sends out the HIPERLAN/2 standard command "RLC_TRANS_CC_DATA" to the central controller candidate CC-CAND of the second new wireless ad hoc network (the first wireless terminal WT1). With this command the central controller CC transmits all connection-relevant data to the first wireless terminal WT1 (central controller candidate CC-CAND of the second new network). It is therefore possible to maintain all direct link connections that shall be moved to the new network without the need of a new setup procedure. After the central controller candidate CC-CAND of the new network, here the first wireless terminal WT1, is aware of all network-relevant data it sends the standard HIPERLAN/2 command "RLC_CC_START_OPERATION" to all wireless terminals of the new network. Besides other information within the command "RLC_CC_START_OPERATION" also a time stamp is transmitted indicating at which time the wireless terminal that receives the command may start its RLC. This way, all wireless terminals of the new network start their RLC at the same moment in time.

[0050] In the next step, i.e. in step S512, the central controller candidate CC-CAND (WT1) sends a command "RLC_TRANS_DATA_ACK" of the HIPERLAN/2 standard to the central controller to acknowledge the reception of the command "RLC_TRANS_CC_DATA".

[0051] In the following step S513, the central controller candidate CC_CAND (WT1) sends the HIPERLAN/2 standard command "RLC_CC_START_OPERATION" to the third wireless terminal WT3. In step S514 the central controller candidate CC_CAND (WT1) then sends the HIPERLAN/2 standard command "RLC_CC_START_OPERATION" to the fifth wireless terminal WT5.

[0052] In step S515, when the time indicated by the time stamps of the commands "RLC_CC_START_OPERATION" transmitted to the third wireless terminal WT3 and the fifth wireless terminal WT5 in steps S513 and S514 arrives, the RLC of the third wireless terminal WT3 and the fifth wireless terminal WT5 and the RLC of the second new central controller CC2, i.e. CC-CAND (WT1), are enabled.

[0053] In the final state S516 the separation process is finished and the second new wireless ad hoc network

is running as shown in Fig. 4B.

[0054] In the following the invention is summarized:

[0055] A first wireless ad hoc network with a certain available amount of bandwidth comprises a plurality of wireless terminals communicating via wireless connections with each other. Each wireless connection requires a certain amount of the available amount of bandwidth. A problem occurs, if a wireless terminal wants to set up a new wireless connection requiring bandwidth or requires more bandwidth for an existing wireless connection and the required amount of bandwidth exceeds the left amount of the available amount of bandwidth of the first wireless ad hoc network. In a state of the art wireless ad hoc network the requested bandwidth cannot be provided and a user application may not be possible. According to the invention, however, the required amount of bandwidth can be provided by splitting the first wireless ad hoc network such that a second wireless ad hoc network is spawned which operates in a different communication channel and thus provides additional bandwidth. The second wireless ad hoc network comprises wireless terminals of the first wireless ad hoc network and/or new wireless terminals.

Claims

1. A method to provide additional bandwidth for a wireless ad hoc network operating in a certain communication channel with a certain amount of available bandwidth comprising a plurality of wireless terminals
characterized in that
said wireless ad hoc network is split up such that at least one new wireless ad hoc network is spawned, if more bandwidth than said certain amount of available bandwidth is required by said plurality of wireless terminals,
wherein after the split of said wireless ad hoc network at least one wireless terminal of said wireless ad hoc network and/or one or more new wireless terminals belong(s) to said at least one new wireless ad hoc network, and
said at least one new wireless ad hoc network is operating in a respective different communication channel.
2. The method according to claim 1,
characterized in that
said wireless ad hoc network and the splitting of said wireless ad hoc network are controlled by a central controller of said wireless ad hoc network that decides which wireless terminals of said wireless ad hoc network and/or which new wireless terminals are moved to said at least one new wireless ad hoc network,
wherein the decision is based on certain separation criteria, and

said central controller determines a new central controller for said at least one new wireless ad hoc network.

3. The method according to claim 1 or 2,
characterized in that
said wireless ad hoc network and said at least one new wireless ad hoc network are operated according to the IEEE802.11 or ETSI BRAN HIPERLAN/2 standard.
4. The method according to anyone of the preceding claims,
characterized in that
said certain separation criteria assure that wireless terminals that
 - have the same convergence layers, and/or
 - are of the same application layer, and/or
 - have the same connection, and/or
 - provide the same or corresponding functions
are not separated into different wireless ad hoc networks.
5. The method according to anyone of the preceding claims,
characterized in that
said certain separation criteria assure that wireless terminals with certain connections that should not be interrupted are not moved to said at least one new wireless ad hoc network.
6. The method according to anyone of the preceding claims,
characterized by
providing new commands in order to spawn said at least one new wireless ad hoc network, wherein a requesting command (SPAWN_NETWORK) is sent to a request wireless terminal to ask this request wireless terminal to move to said at least one new ad hoc wireless network, and a confirmation command (SPAWN_NETWORK_ACK) is used by a request wireless terminal to signal that it can move to said at least one new ad hoc wireless network.
7. The method according to claim 6,
characterized in that
a wireless terminal stops using its entire wireless connections the moment it sent out said confirmation command (SPAWN_NETWORK_ACK), moves to one of said at least one new wireless ad hoc network,
waits until it receives a start command (RLC_CC_START_OPERATION) sent out by a central controller, and
then starts using its wireless connections ac-

ording to the information provided by said start command (RLC_CC_START_OPERATION).

8. A wireless terminal of a wireless ad hoc network controlled by a central controller of said wireless ad hoc network

characterized by

a receiving means adapted to receive a requesting command (SPAWN_NETWORK) from the central controller indicating certain operating conditions for the wireless terminal,

a condition checking means to check if the wireless terminal can be operated under said certain conditions, and

a sending means that sends out a confirmation command (SPAWN_NETWORK_ACK), if the condition checking means determines that the wireless terminal can be operated under said certain conditions.

9. The wireless terminal according to claim 8

characterized in that

said certain conditions define

if said wireless terminal can operate as a central controller of a wireless ad hoc network,

a certain communication channel at which said wireless terminal is able to operate, and/or

a moment in time at which said wireless terminal shall operate in said certain communication channel and at which it may be controlled by a different central controller.

10. A central controller of a wireless ad hoc network comprising a plurality of wireless terminals,

characterized by

a splitting means that controls the splitting of said wireless ad hoc network, wherein a new wireless ad hoc network is spawned that comprises at least one of said plurality of wireless terminals and/or one or more new wireless terminals.

11. The central controller according to claim 10,

characterized in that

the splitting means comprises

a sending means that sends out requesting commands (SPAWN_NETWORK) to wireless terminals,

a receiving means that receives confirmation commands (SPAWN_NETWORK_ACK), and

an operating means that decides which of said plurality of wireless terminals and/or of said new wireless terminals may be moved to said new wireless ad hoc network and determines a wireless terminal of said plurality of wireless terminals and/or of said new wireless terminals that becomes the central controller of said new wireless ad hoc network.

12. A wireless ad hoc network comprising
a plurality of wireless terminals according to claim 7 or 8, and
a central controller according to claim 9 or 10.

Amended claims in accordance with Rule 86(2) EPC.

1. A method to provide additional bandwidth for a wireless ad hoc network operating in a certain communication channel with a certain amount of available bandwidth comprising a plurality of wireless terminals, **characterized in that**

- said wireless ad hoc network is split up such that at least one new wireless ad hoc network is spawned, if more bandwidth than said certain amount of available bandwidth is required by said plurality of wireless terminals,
- wherein after the split of said wireless ad hoc network at least one wireless terminal of said wireless ad hoc network and/or one or more new wireless terminals belong(s) to said at least one new wireless ad hoc network, and
- said at least one new wireless ad hoc network is operating in a respective different communication channel,
- wherein the decision which of said wireless terminals of said wireless ad hoc network and/or which of said at least one wireless terminal are moved to said at least one new wireless ad hoc network is based on certain separation criteria,
- which certain separation criteria assure that wireless terminals that
 - have the same convergence layers, and/or
 - are of the same application layer, and/or
 - have the same connection, and/or
 - provide the same or corresponding functions are not separated into different wireless ad hoc networks.

2. The method according to claim 1, **characterized in that**

said wireless ad hoc network and the splitting of said wireless ad hoc network are controlled by a central controller of said wireless ad hoc network that decides which wireless terminals of said wireless ad hoc network and/or which new wireless terminals are moved to said at least one new wireless ad hoc network,

wherein the decision is based on said certain separation criteria, and
said central controller determines a new central controller for said at least one new wireless ad hoc network.

3. The method according to claim 1 or 2, **characterized**

terized in that

said wireless ad hoc network and said at least one new wireless ad hoc network are operated according to the IEEE802.11 or ETSI BRAN HIPER-LAN/2 standard.

4. The method according to anyone of the preceding claims, characterized in that

said certain separation criteria assure that wireless terminals With certain connections that should not be interrupted are not moved to said at least one new wireless ad hoc network.

5. The method according to anyone of the preceding claims, characterized by

providing new commands in order to spawn said at least one new wireless ad hoc network,

wherein a requesting command (SPAWN_NETWORK) is sent to a request wireless terminal to ask this request wireless terminal to move to said at least one new ad hoc wireless network,

and a confirmation command (SPAWN_NETWORK_ACK) is used by a request wireless terminal to signal that it can move to said at least one new ad hoc wireless network.

6. The method according to claim 5, characterized in that

a wireless terminal stops using its entire wireless connections the moment it sent out said confirmation command (SPAWN_NETWORK_ACK),

moves to one of said at least one new wireless ad hoc network,

waits until it receives a start command (RLC_CC_START_OPERATION) sent out by a central controller, and

then starts using its wireless connections according to the information provided by said start command (RLC_CC_START_OPERATION).

7. A wireless terminal of a wireless ad hoc network controlled by a central controller of said wireless ad hoc network characterized by

a receiving means adapted to receive a requesting command (SPAWN_NETWORK) from the central controller indicating certain operating conditions for the wireless terminal,

a condition checking means to check if the wireless terminal can be operated under said certain conditions, and

a sending means that sends out a confirmation command (SPAWN_NETWORK_ACK), if the condition checking means determines that the wireless terminal can be operated under said certain conditions.

8. The wireless terminal according to claim 7, characterized in that

said certain conditions define

if said wireless terminal can operate as a central controller of a wireless ad hoc network,

a certain communication channel at which said wireless terminal is able to operate, and/or

a moment in time at which said wireless terminal shall operate in said certain communication channel and at which it may be controlled by a different central controller.

9. A central controller of a wireless ad hoc network comprising a plurality of wireless terminals, characterized by

a splitting means that controls the splitting of said wireless ad hoc network, wherein a new wireless ad hoc network is spawned that comprises at least one of said plurality of wireless terminals and/or one or more new wireless terminals.

10. The central controller according to claim 10, characterized in that

the splitting means comprises

a sending means that sends out requesting commands (SPAWN_NETWORK) to wireless terminals,

a receiving means that receives confirmation commands (SPAWN_NETWORK_ACK), and

an operating means that decides which of said plurality of wireless terminals and/or of said new wireless terminals may be moved to said new wireless ad hoc network and determines a wireless terminal of said plurality of wireless terminals and/or of said new wireless terminals that becomes the central controller of said new wireless ad hoc network.

11. A wireless ad hoc network comprising

a plurality of wireless terminals according to claim 7 or 8, and

a central controller according to claim 9 or 10.

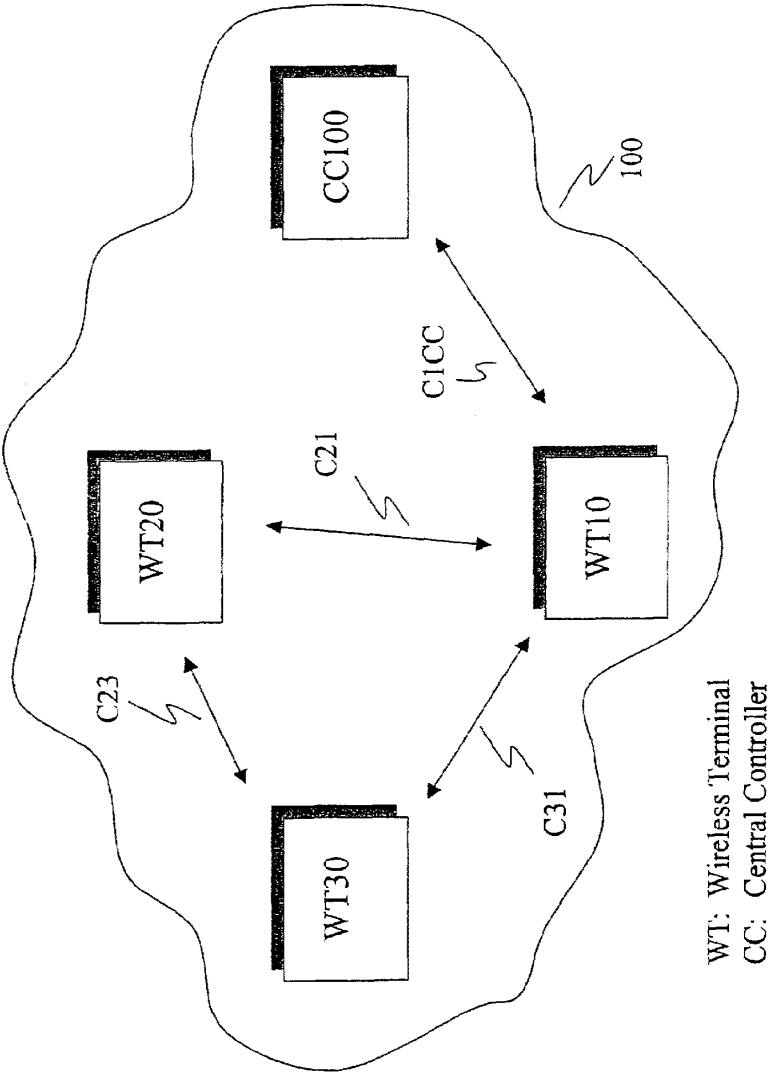


Figure 1

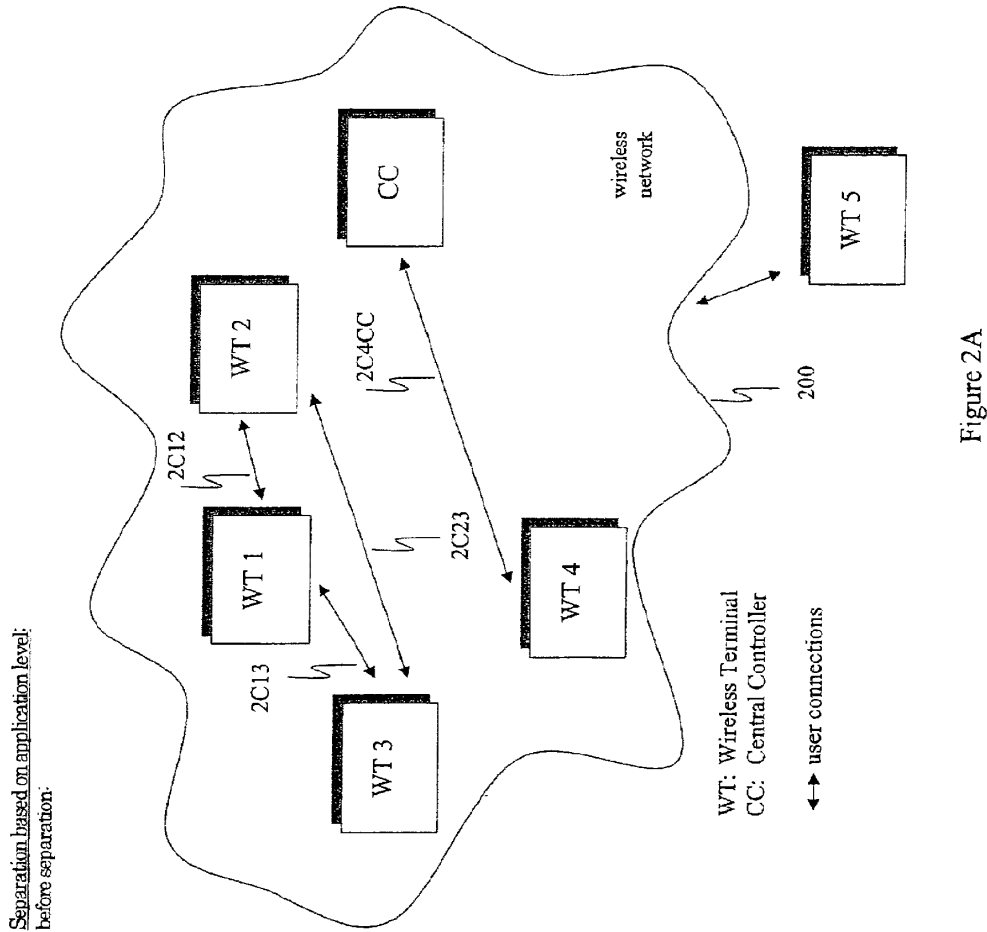


Figure 2A

Separation based on application level:
after separation:

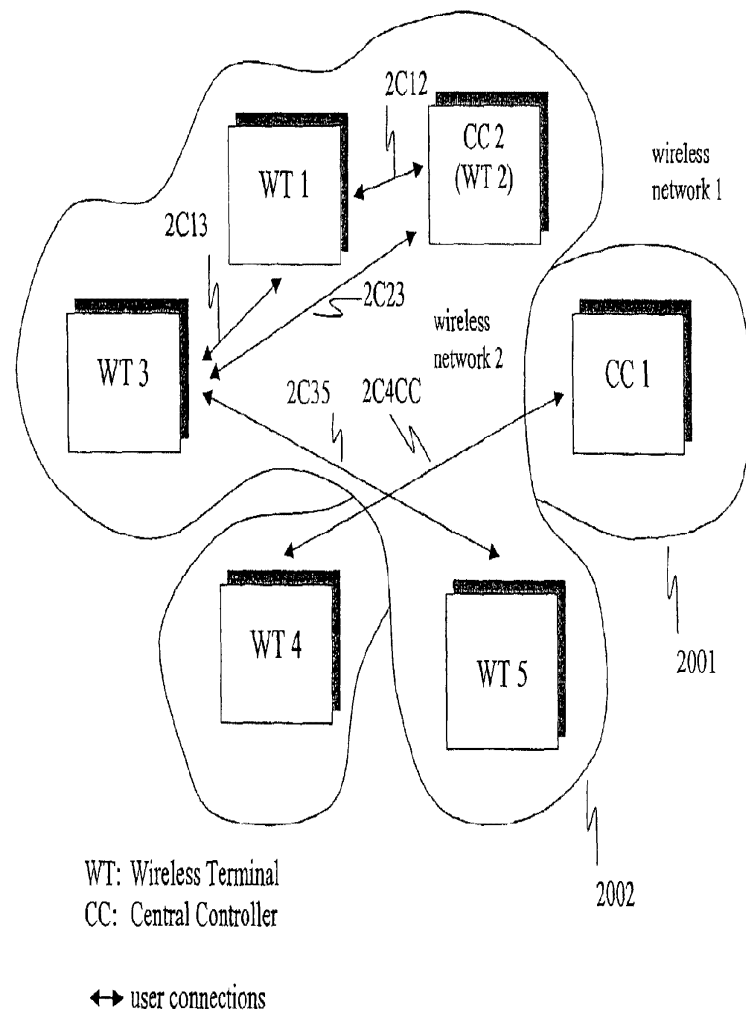


Figure 2B

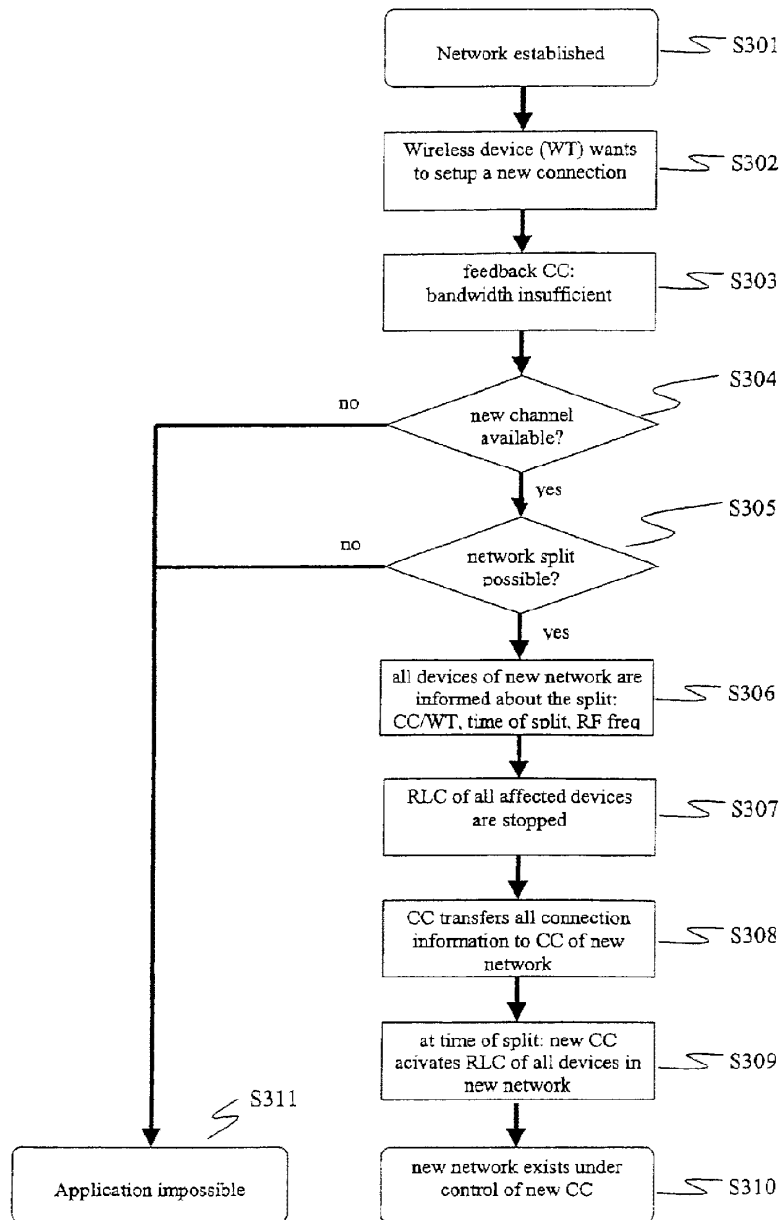


Figure 3

Separation based on connection type (different converges layers or application):

before separation:

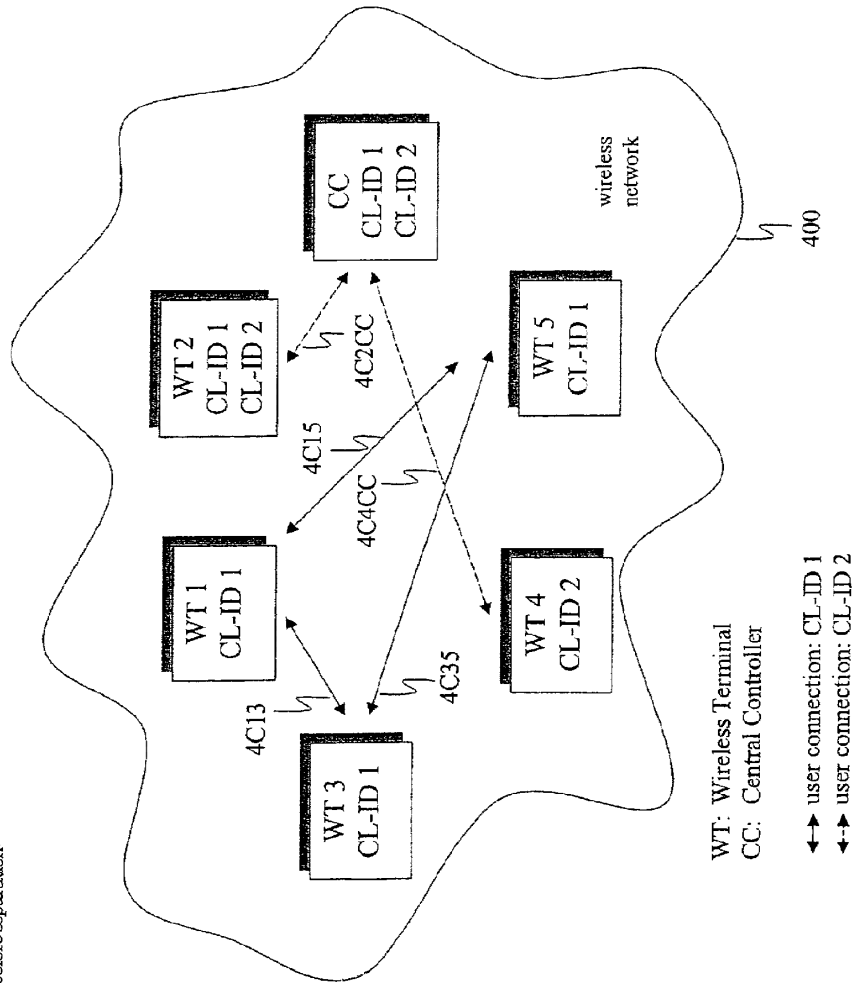


Figure 4A

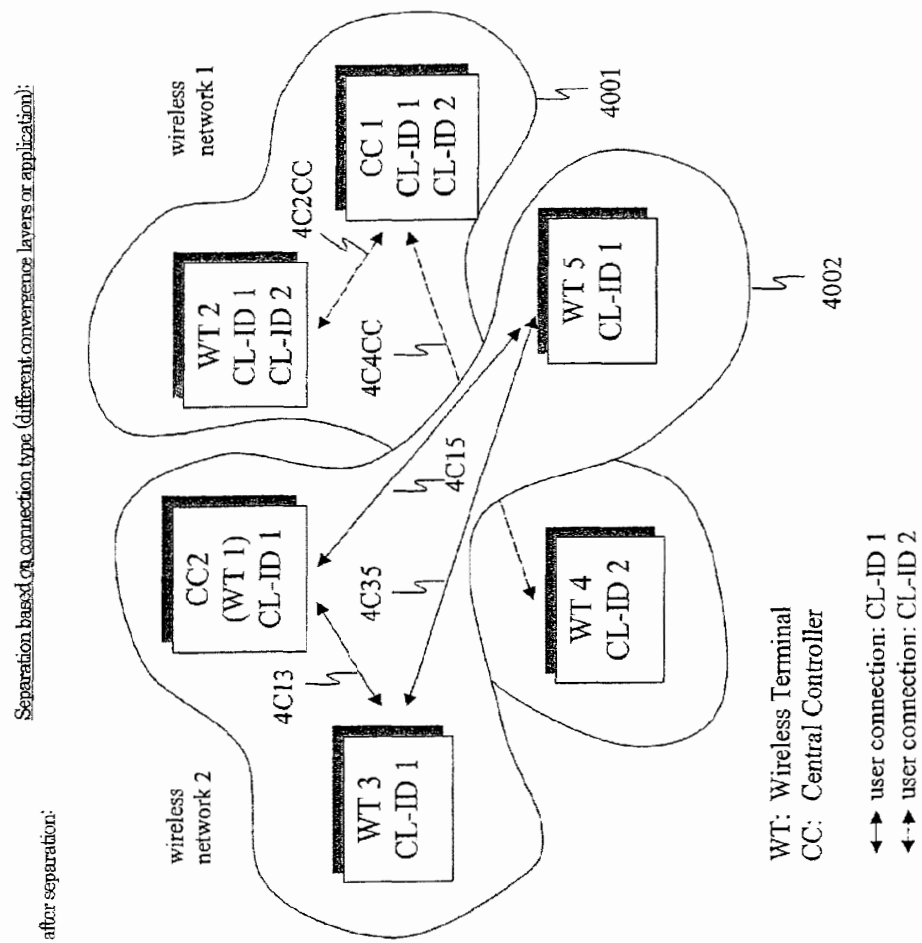
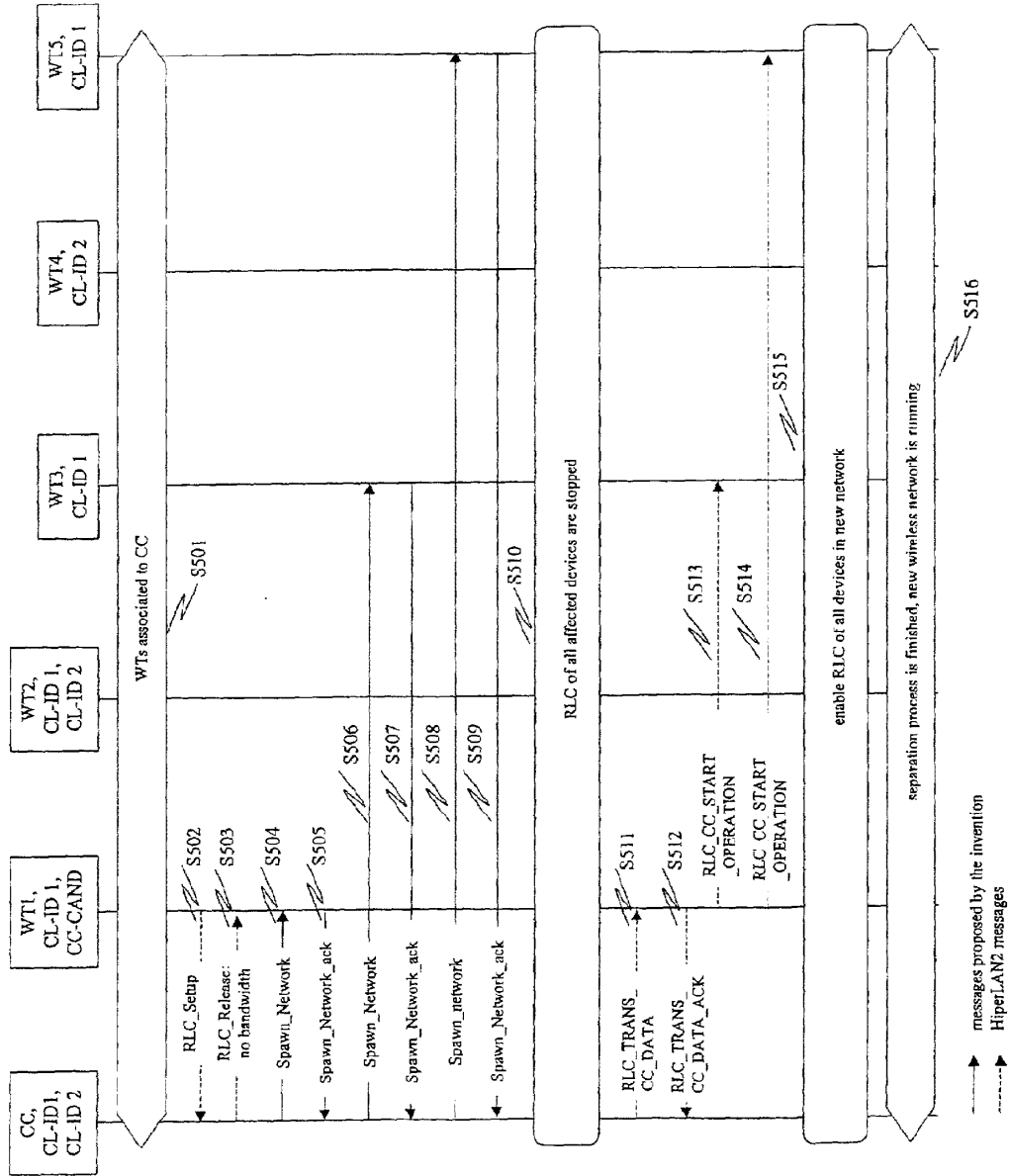


Figure 4B

Figure 5





European Patent
Office

EUROPEAN SEARCH REPORT

Application Number
EP 02 01 8385

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The present search report has been drawn up for all claims			TECHNICAL FIELDS SEARCHED (Int.Cl.7) H04L
Place of search THE HAGUE		Date of completion of the search 29 January 2003	Examiner Heinrich, D
<p>CATEGORY OF CITED DOCUMENTS</p> <p>X: particularly relevant if taken alone Y: particularly relevant if combined with another document of the same category A: technological background O: non-written disclosure P: intermediate document</p> <p>T: theory or principle underlying the invention E: earlier patent document, but published on, or after the filing date D: document cited in the application L: document cited for other reasons ... &: member of the same patent family, corresponding document</p>			

EPO FORM 1503 08 82 (P04C01)



European Patent
Office

EUROPEAN SEARCH REPORT

Application Number
EP 02 01 8385

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A	CAMPBELL A T ET AL: "SPAWNING NETWORKS" IEEE NETWORK, IEEE INC. NEW YORK, US, July 1999 (1999-07), pages 16-29, XP000875025 ISSN: 0890-8044 * abstract * * page 16, right-hand column, line 19 - page 17, right-hand column, line 34 * * page 18, right-hand column, line 53 - page 19, right-hand column, line 34 * * page 20, right-hand column, line 36 - page 21, left-hand column, line 40 * * page 24, right-hand column, line 12 - page 26, left-hand column, line 36 * * page 27, left-hand column, line 56 - right-hand column, line 2 * -----	1-12	
A	US 2002/018448 A1 (AMIS ALAN DEWAYNE ET AL) 14 February 2002 (2002-02-14) * page 1, paragraphs 2,4 * * page 3, paragraph 44 * * page 6, paragraph 108 * -----	1-12	TECHNICAL FIELDS SEARCHED (Int.Cl.7)
The present search report has been drawn up for all claims			
Place of search THE HAGUE		Date of completion of the search 29 January 2003	Examiner Heinrich, D
<p>CATEGORY OF CITED DOCUMENTS</p> <p>X : particularly relevant if taken alone Y : particularly relevant if combined with another document of the same category A : technological background O : non-written disclosure P : intermediate document</p> <p>T : theory or principle underlying the invention E : earlier patent document, but published on, or after the filing date D : document cited in the application L : document cited for other reasons & : member of the same patent family, corresponding document</p>			

EPO FORM 1503 03 82 (P4/C01)

**ANNEX TO THE EUROPEAN SEARCH REPORT
ON EUROPEAN PATENT APPLICATION NO.**

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This annex lists the patent family members relating to the patent documents cited in the above-mentioned European search report.
The members are as contained in the European Patent Office EDP file on
The European Patent Office is in no way liable for these particulars which are merely given for the purpose of information.

29-01-2003

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EPO FORM P0459

For more details about this annex : see Official Journal of the European Patent Office, No. 12/82



Espacenet

Bibliographic data: JP2009075540 (A) — 2009-04-09**DIGITAL DOWNLOAD JUKEBOX WITH EXPANDED COMMUNICATION FEATURE**

Inventor(s): DION DOMINIQUE; NATHAN GUY; BRILLON FRANCOIS;
GRATTON LOIC ± (DION DOMINIQUE, ; NATHAN GUY, ;
BRILLON FRANCOIS, ; GRATTON LOIC)

Applicant(s): TOUCHTUNES MUSIC CORP ± (TOUCHTUNES MUSIC CORP)

Classification: - international: G10K15/04
- cooperative: G06F3/04842; G06Q30/02; G06Q30/0207;
G06Q30/0235; G06Q30/06; G07F17/305;
G11B27/105

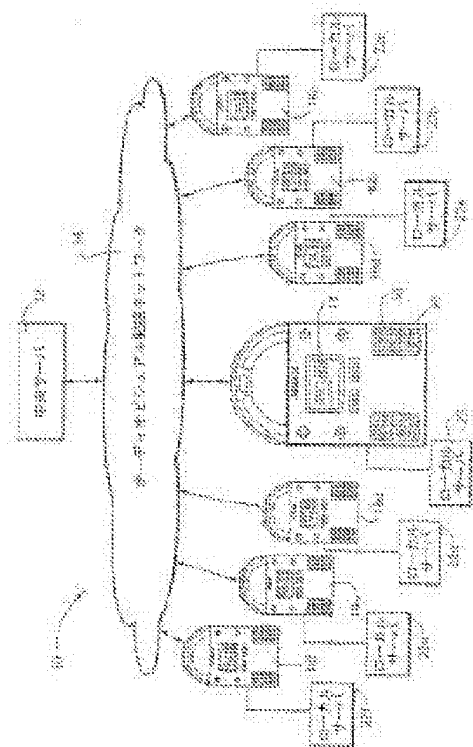
Application number: JP20080079311 20080325 Global Dossier

Priority number (s): US20070902658 20070924

Also published as: JP5795700 (B2) US2008086379 (A1) TW200915813 (A)
TWI499264 (B) MX2008004032 (A) more

Abstract of JP2009075540 (A)

PROBLEM TO BE SOLVED: To provide a system or the like for a digital download jukebox enhanced in a communication capacity to a registered user. ; SOLUTION: A user can get directly to the registered user in the jukebox. The registered user is allowed to receive a coupon for free replay, to import a replay list from a hardware device and/or a software application, to receive a specially set price when replaying a medium instance, to regenerate the replay list, to prepare and manage the replay list directly in the jukebox, and to establish and manage the connection with other registered user. Further, one or plurality of channels may be predefined and set up as to the jukebox. ; COPYRIGHT: (C)2009,JPO&INPIT



(19) 日本国特許庁(JP)

(12) 公開特許公報(A)

(11) 特許出願公開番号

特開2009-75540

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(43) 公開日 平成21年4月9日(2009. 4. 9)

(51) Int. Cl.

G10K 15/04 (2006.01)

F1

G10K 15/04 302F

テーマコード(参考)

5D108

審査請求 未請求 請求項の数 59 O L 外国語出願 (全 199 頁)

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 (31) 優先権主張番号 11/902, 658
 (32) 優先日 平成19年9月24日(2007. 9. 24)
 (33) 優先権主張国 米国(US)

特許法第64条第2項第4号の規定により明細書の一部
 または全部を不掲載とする。

特許法第64条第2項第4号の規定により図面の一部ま
 たは全部を不掲載とする。

(特許庁注：以下のものは登録商標)

1. Bluetooth

(71) 出願人 508089820
 タッチチューンズ・ミュージック・コーポ
 レーション

アメリカ合衆国・ニューヨーク・1000
 3・ニュー・ヨーク・ブロードウェイ・7
 40・イレブンス・フロア・#1102

(74) 代理人 100064908

弁理士 志賀 正武

(74) 代理人 100089037

弁理士 渡邊 隆

(74) 代理人 100108453

弁理士 村山 靖彦

(74) 代理人 100110364

弁理士 実広 信哉

最終頁に続く

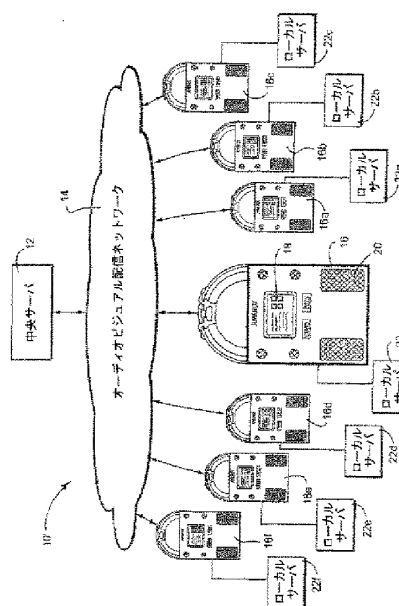
(54) 【発明の名称】 拡張された通信フィーチャを有するデジタルダウンロードジュークボックス

(57) 【要約】

【課題】登録ユーザに通信能力が強化されたデジタルダ
 ウンロードジュークボックスのシステム等を提供すること。

【解決手段】ユーザは、ジュークボックスにおいて直接
 に、登録されたユーザになることができる。登録された
 ユーザは、無料再生のためのクーポンを受け取ること、
 ハードウェアデバイス、ソフトウェアアプリケーション
 から再生リストをインポートすること、メディアインス
 タンスを再生する際に特別価格設定を受けること、再生
 リストを再生すること、ジュークボックスにおいて直接
 に再生リストを作成し管理すること、登録された他のユ
 ーザとの結び付きを確立して管理することなどもできる。
 さらに、1つまたは複数のチャンネルが、ジュークボッ
 クスに関して事前定義され、セットアップされることが
 可能である。

【選択図】図2



【特許請求の範囲】**【請求項1】**

オーディオビジュアル配信ネットワークに接続された1つまたは複数のデジタルジュークボックスに再生リストを配信する方法であって、

周辺デバイス上のコンピュータ可読記憶媒体の中に格納された周辺再生リストを有する該周辺デバイスを、コネクタを介してデジタルジュークボックスに接続するステップと、

前記デジタルジュークボックスのために前記周辺再生リストを取り出すステップと、

前記周辺再生リストをジュークボックス再生リストに変換するステップと、

前記ジュークボックス再生リストを、少なくとも前記ジュークボックス上で公開するステップとを含む方法。

【請求項2】

周辺デバイスは、ポータブル音楽プレーヤ、コンピュータ、および移動電話機の少なくとも1つである請求項1に記載の方法。

【請求項3】

前記周辺再生リストを、前記デジタルジュークボックスのコンピュータ可読記憶媒体に少なくとも一時的に格納するステップをさらに含む請求項1に記載の方法。

【請求項4】

前記ジュークボックス再生リストが、公開ジュークボックス再生リストと指定されるか、プライベートジュークボックス再生リストと指定されるかを決定するステップをさらに含む請求項1に記載の方法。

【請求項5】

前記ジュークボックス再生リストを、前記オーディオビジュアル配信ネットワークのサーバに送信するステップをさらに含む請求項1に記載の方法。

【請求項6】

前記ジュークボックス再生リストを、前記オーディオビジュアル配信ネットワークに接続されたジュークボックスに提供するステップをさらに含む請求項5に記載の方法。

【請求項7】

前記ジュークボックス再生リストを、前記オーディオビジュアル配信ネットワークに接続された遠隔デバイスに提供するステップをさらに含む請求項5に記載の方法。

【請求項8】

前記コネクタは、USB接続である請求項1に記載の方法。

【請求項9】

前記周辺再生リストのインポートが、可能にされるにはまず、前記ジュークボックスにユーザがログインすることを要求するステップをさらに含む請求項1に記載の方法。

【請求項10】

オーディオビジュアル配信ネットワークに接続された1つまたは複数のデジタルジュークボックスに再生リストを配信する方法であって、

前記デジタルジュークボックス上で使用するために、コンピュータデバイス上で使用されるソフトウェアアプリケーションに関連するソフトウェアアプリケーション再生リストを取り出すステップと、

前記ソフトウェアアプリケーション再生リストをジュークボックス再生リストに変換するステップと、

少なくとも前記ジュークボックス上で前記ジュークボックス再生リストを公開するステップとを含む方法。

【請求項11】

前記ソフトウェアアプリケーション再生リストを、前記コンピュータデバイスのコンピュータ可読記憶媒体に少なくとも一時的に格納するステップをさらに含む請求項10に記載の方法。

【請求項12】

前記ジュークボックス再生リストが、公開ジュークボックス再生リストと指定されるか、プライベートジュークボックス再生リストと指定されるかを決定するステップをさらに含む請求項10に記載の方法。

【請求項13】

前記ジュークボックス再生リストを、前記オーディオビジュアル配信ネットワークのサーバに送信するステップをさらに含む請求項10に記載の方法。

【請求項14】

前記ジュークボックス再生リストを、前記オーディオビジュアル配信ネットワークのサーバに送信するステップをさらに含む請求項10に記載の方法。

【請求項15】

前記ジュークボックス再生リストを、前記オーディオビジュアル配信ネットワークに接続されたジュークボックスに提供するステップをさらに含む請求項14に記載の方法。

【請求項16】

前記ジュークボックス再生リストを、前記オーディオビジュアル配信ネットワークに接続された遠隔デバイスに提供するステップをさらに含む請求項14に記載の方法。

【請求項17】

前記周辺再生リストのインポートが、可能にされるにはまず、前記ジュークボックスにユーザがログインすることを要求するステップをさらに含む請求項10に記載の方法。

【請求項18】

オーディオビジュアル配信ネットワークに接続された1つまたは複数の異なるデジタルジュークボックスに再生リストを配信するように構成されたコミュニケータと、

周辺デバイス上のコンピュータ可読記憶媒体の中に格納された周辺再生リストを有する該周辺デバイスを、前記デジタルジュークボックスに接続するように構成されたコネクタと、

前記デジタルジュークボックスのために前記周辺再生リストを取り出すように構成されたプロセッサと、

前記周辺再生リストをジュークボックス再生リストに変換するように構成された変換プログラミング論理回路とを含み、前記プロセッサは、少なくとも前記ジュークボックスにおいて前記ジュークボックス再生リストを公開するようにさらに構成されるデジタルダウンロードジュークボックス。

【請求項19】

周辺デバイスは、ポータブル音楽プレーヤ、コンピュータ、および携帯電話機の少なくとも1つである請求項18に記載のジュークボックス。

【請求項20】

前記周辺再生リストを少なくとも一時的に格納するように構成された前記デジタルジュークボックスに関連するコンピュータ可読記憶媒体をさらに含む請求項18に記載のジュークボックス。

【請求項21】

前記プロセッサは、前記ジュークボックス再生リストが、公開ジュークボックス再生リストと指定されるか、プライベートジュークボックス再生リストと指定されるかを決定するようにさらに構成される請求項18に記載のジュークボックス。

【請求項22】

前記コミュニケータは、前記オーディオビジュアル配信ネットワークのサーバに前記ジュークボックス再生リストを送信するように構成される請求項18に記載のジュークボックス。

【請求項23】

前記オーディオビジュアル配信ネットワーク上の前記サーバのモジュールは、前記オーディオビジュアル配信ネットワークに接続されたジュークボックスに前記ジュークボックス再生リストを提供するように構成される請求項22に記載のジュークボックス。

【請求項24】

前記オーディオビジュアル配信ネットワーク上の前記サーバのモジュールは、前記オーディオビジュアル配信ネットワークに接続された遠隔デバイスに前記ジュークボックス再生リストを提供するように構成される請求項22に記載のジュークボックス。

【請求項25】

前記コネクタは、USB接続である請求項18に記載のジュークボックス。

【請求項26】

前記コネクタは、独自のドッキングステーションである請求項18に記載のジュークボックス。

【請求項27】

前記プロセッサは、前記再生リストのインポートより前に前記ジュークボックスにログインするよう、ユーザに要求するようにさらに構成される請求項18に記載のジュークボックス。

【請求項28】

デジタルダウンロードジュークボックス上でメディアインスタンスを再生するための方法であって、

前記ジュークボックスにログインするよう、ユーザに要求するステップと、

前記ジュークボックス上で再生するために複数の選択可能なメディアインスタンスを提供するステップと、

前記ジュークボックスに対するユーザインターフェースを介して、前記複数の選択可能なメディアインスタンスからのメディアインスタンスのユーザ選択を可能にするステップと、

各ユーザ選択に関して、前記ジュークボックスのコンピュータ可読記憶媒体の中に格納された再生リストに前記メディアインスタンスを追加するステップと、

前記再生リストの中の前記メディアインスタンスを順に再生するステップとを含み、前記複数の選択可能なメディアインスタンスは、前記ユーザによるメディアインスタンスの前の再生に依存して、メディアインスタンスのマスタグループから選択される方法。

【請求項29】

ある所定の期間内に前記ユーザによって前に再生されたメディアインスタンスから前記複数の選択可能なメディアインスタンスを形成するステップをさらに含む請求項28に記載の方法。

【請求項30】

ある所定のロケーションにおいて前記ユーザによって前に再生されたメディアインスタンスから前記複数の選択可能なメディアインスタンスを形成するステップをさらに含む請求項28に記載の方法。

【請求項31】

前記ユーザに関連付けられた最近、再生されたメディアインスタンスのリストの中に格納されたメディアインスタンスから前記複数の選択可能なメディアインスタンスを形成するステップをさらに含む請求項28に記載の方法。

【請求項32】

前記ユーザが利用できるクレジット数を決定するステップと、

前記クレジット数が、前記再生リストの中の前記メディアインスタンスのすべてを再生するのに十分である場合、前記再生リストの中のすべてのメディアインスタンスを再生するステップと、

前記クレジット数が、前記再生リストの中の前記メディアインスタンスのすべてを再生するには不十分である場合、前記再生リストの中の前記メディアインスタンスの一部だけを再生するステップとをさらに含む請求項28に記載の方法。

【請求項33】

前記クレジット数が、前記再生リストの中の前記メディアインスタンスのすべてを再生するには不十分である場合、前記ユーザが、再生のために前記再生リストの中の前記メディアインスタンスの前記一部を選択することができるようにするステップをさらに含む

請求項32に記載の方法。

【請求項34】

前記クレジット数が、前記再生リストの中の前記メディアインスタンスのすべてを再生するには不十分である場合、前記クレジットが尽きるまで、前記再生リストの中の前記メディアインスタンスを順に再生するステップをさらに含む請求項32に記載の方法。

【請求項35】

デジタルダウンロードジュークボックス上でメディアインスタンスを再生する方法であって、

前記ジュークボックス上で複数のメディアインスタンスを提供するステップと、

前記ジュークボックスのユーザにインターフェースを提供して、前記ユーザが、前記ジュークボックス上で再生するためにある特定のメディアインスタンスを検索することができるようにするステップと、

前記メディアインスタンスが、前記ジュークボックスによるさらなる処理なしに、前記ジュークボックス上で即時の再生に利用できるかどうかを判定するステップと、

前記メディアインスタンスが、即時の再生のために利用できる場合、前記ユーザに所定の第1のクレジット数を請求して、前記メディアインスタンスを再生のためにキューに入れるステップと、

前記メディアインスタンスが、即時の再生に利用できない場合、前記ユーザが、検証されたユーザであるかどうかを判定するステップと、

前記ユーザが、検証されたユーザであるとき、前記メディアインスタンスが、前記ユーザに関連付けられた再生リストの中に存在する場合、第2の所定のクレジット数を請求するが、存在しない場合、第3の所定のクレジット数を請求するステップと、

前記メディアインスタンスを再生のためにキューに入れるステップとを含み、

前記第2の所定のクレジット数は、前記第3の所定のクレジット数より少なく、前記第3の所定のクレジット数は、所定のクレジットの前記第1の数より多い方法。

【請求項36】

前記ユーザに関連付けられた前記再生リストは、前記ユーザによって前に作成された再生リストである請求項35に記載の方法。

【請求項37】

前記ユーザに関連付けられた前記再生リストは、前記ユーザによって最近、再生された曲を含む再生リストである請求項35に記載の方法。

【請求項38】

前記ユーザに関連付けられた前記再生リストは、別のユーザによって作成されたが、前記ユーザによって少なくとも部分的に最近、再生された再生リストである請求項35に記載の方法。

【請求項39】

前記メディアインスタンスが、前記ジュークボックスによるさらなる処理なしに、前記ジュークボックス上で即時の再生に利用できるかどうかを前記判定するステップは、前記メディアインスタンスが、即時に再生可能なメディアインスタンスを含む前記ジュークボックスの専用のストレージ領域とは別個のストレージ領域上に存在するかどうかを判定するステップを含む請求項35に記載の方法。

【請求項40】

前記別個のストレージ領域は、遠隔サーバである請求項39に記載の方法。

【請求項41】

前記別個のストレージ領域は、前記ジュークボックスの前記専用のストレージ領域のパーティションである請求項39に記載の方法。

【請求項42】

前記別個のストレージ領域は、前記ジュークボックス内部の別個のストレージ領域である請求項39に記載の方法。

【請求項43】

前記メディアインスタンスが、前記ジュークボックスによるさらなる処理なしに、前記ジュークボックス上で即時の再生に利用できるかどうかを前記判定するステップは、前記インターフェースによって提供される前記ユーザの検索方法を特定するステップを含む請求項35に記載の方法。

【請求項44】

デジタルダウンロードジュークボックスを動作させる方法であって、

複数のチャネルを定義し、それぞれの前記チャネルが、該チャネルに関連付けられた複数のメディアインスタンスを有するステップと、

認定されたユーザが、前記複数のチャネルからあるチャネルを選択するステップと、

第1のモードにおける前記ジュークボックスの1回または複数回の利用客使用にตอบสนองして、前記使用の少なくとも数回は、前記ジュークボックス上で再生するためにキューにメディアインスタンスを追加して、前記キューに入れられたメディアインスタンスを再生するステップを含むステップと、

前記ジュークボックスが、利用客要求にตอบสนองしない場合、かつ/またはキューに入れられたメディアインスタンスを再生している場合、前記チャネルにおける前記メディアインスタンスを第2のモードで再生するステップとを含み、

前記第1のモードは、前記第2のモードとは異なり、前記第2のモードは、前記ジュークボックスの利用客が直接に制御可能ではない方法。

【請求項45】

前記メディアインスタンスは、認定されたユーザによってチャネルに関連付けられる請求項44に記載の方法。

【請求項46】

前記メディアインスタンスは、それぞれの前記メディアインスタンスに関連するテーマおよび/またはジャンルに従ってチャネルに関連付けられる請求項44に記載の方法。

【請求項47】

制御コードが、前記認定されたユーザから受け取られると、前記チャネルによって再生されているメディアインスタンスを飛ばすステップをさらに含む請求項44に記載の方法。

【請求項48】

前記認定されたユーザから受け取られた1つまたは複数の制御コードに依存して、あるメディアインスタンスが、あるチャネルにおいて再生される頻度を変更するステップをさらに含み、前記制御コードは、前記チャネルによって再生されているメディアインスタンスを飛ばすこと、前記チャネルによって再生されている、または再生されたばかりのメディアインスタンスをリプレーすること、および前記チャネルからメディアインスタンスを削除することの少なくとも1つに対応する請求項44に記載の方法。

【請求項49】

前記第2のモードは、バックグラウンドミュージックモードである請求項44に記載の方法。

【請求項50】

デジタルダウンロードジュークボックスのユーザにクーポンを届ける方法であって、

有効性確認コードを作成するステップと、

前記有効性確認コードに、前記ユーザのための無料クレジットの数を関連付けるステップと、

前記有効性確認コードを前記クーポンに添えるステップと、

前記有効性確認コードを前記ユーザに伝送するステップとを含む方法。

【請求項51】

前記有効性確認コードの中に、前記ユーザが前記クーポンを引き換えることができる少なくとも1つのロケーションを含めるステップをさらに含む請求項50に記載の方法。

【請求項52】

前記有効性確認コードの中に、前記ユーザが前記クーポンを引き換えることができる時間に関する制限を含めるステップをさらに含む請求項50に記載の方法。

【請求項53】

前記クーポンは、前記ユーザの移動電話機に伝送される請求項50に記載の方法。

【請求項54】

前記有効性確認コードは、符号化された英数字シーケンスである請求項50に記載の方法

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【請求項55】

前記有効性確認コードは、バーコードである請求項50に記載の方法。

【請求項56】

前記前記クーポンから前記有効性確認コードを獲得するステップと、

前記有効性確認コードが有効であるかどうかを判定するステップと、

前記有効性確認コードが有効である場合、前記有効性確認コードに関連付けられた無料クレジットの数に応じて、前記ユーザにある数のクレジットを与えるステップとを含む請求項50に記載の方法。

【請求項57】

前記有効性確認コードは、バーコードスキャナを介して前記ジュークボックスによって獲得される請求項56に記載の方法。

【請求項58】

前記有効性確認コードは、前記ジュークボックスに提供されたユーザインターフェースを介して、前記ユーザに、前記有効性確認コードを前記ジュークボックスに入力させることによって獲得される請求項56に記載の方法。

【請求項59】

前記有効性確認コードは、有効な有効性確認コードのデータベースに照らして有効性確認される請求項56に記載の方法。

【発明の詳細な説明】

【技術分野】

【0001】

本発明は、例えば、ジュークボックスシステムに関し、より詳細には、中央サーバと、印税計算および/またはコンテンツ更新のために中央サーバと通信する遠隔ジュークボックスデバイスとを通常、含むタイプのデジタルダウンロードジュークボックスシステムに関し、さらに詳細には、ユーザに合わせられた音楽管理、通信、およびジュークボックスシステムを使用する認識されたユーザのための他のツールを提供するジュークボックスシステムに関する。本発明の例示的な実施形態は、ジュークボックスシステムネットワークにおける各ジュークボックスデバイスのためのローカルサーバを提供することにより、そのようなシステムを改良する。ローカルサーバは、ジュークボックスデバイス上で再生するためにジュークボックスデバイスのユーザによって選択されることが可能なコンテンツ(例えば、オーディオデータおよび/またはビジュアルデータ)の第2のより広範な源を提供する。ローカルサーバは、中央サーバのミラーまたはサブセットを提供して、ジュークボックスデバイスの大容量記憶デバイス上で利用可能でない、要求されるコンテンツを中央サーバからダウンロードする必要性なしに、オーディオデータおよび/またはビジュアルデータのライブラリ全体、または少なくとも、要求されたほとんどのコンテンツが、各ジュークボックスデバイスに都合よく利用可能になることを可能にすることができる。また、ローカルサーバの共同グループが、各ジュークボックスデバイスを介して中央サーバによって、例えば、他のジュークボックスデバイスなどの他のデバイスにサービスを提供するように制御されることが可能な分散コンテンツサーバのネットワークの役割をすることも可能である。さらに、ジュークボックスデバイスおよびローカルサーバは、中央サーバの制御下で、ジュークボックスデバイスと一緒にあるロケーションに存在する様々なダウンロード可能な料金ベースのデバイスのための「中央ハブ」、つまり、管理デバイスとして動作することが可能である。

【0002】

例示的な実施形態は、さらに、例えば、ユーザが、1つまたは複数のジュークボックス上で取り出されることが可能な再生リストを作成して、管理すること、個人にメッセージ

を送信する、または個人の大きいグループにブロードキャストすることができる仲間リストを作成して、管理することを可能にすることなどにより、個々の認識されたユーザの体験を向上させる他の機構を提供することができる。つまり、一実施形態では、ジュークボックスは、ジュークボックスが、事前登録などを介して、特定のジュークボックスユーザを認識すること、およびユーザにユーザ特有のサービスを提供することを可能にするフィーチャを含む。例えば、ジュークボックスは、ジュークボックス上で再生するための再生リストの作成者によって指定されたユーザまたは他の登録されたユーザによって選択されることが可能なカスタム再生リストを、ジュークボックス上で、認識されたユーザが、作成する、編集する、かつ/または格納することを可能にすることができる。また、ジュークボックスは、登録されたユーザが、ジュークボックスを介して、他の登録されたユーザと通信することを可能にすることもできる。また、登録されたユーザは、ジュークボックスサービスを使用するために現金が必要ないように、ジュークボックス上でクレジットを与えられることも可能である。また、ジュークボックスは、本明細書で説明される独特の有利な仕方、従来のジュークボックスサービスに加えて、バックグラウンドミュージックサービスを提供するように使用されることが可能である。ジュークボックスは、例えば、インターネットを介するなどして、ジュークボックスから遠隔のロケーションでアクセス可能な対話型アプリケーションと対話することにより、所有感覚をさらに高めることができる。遠隔アプリケーションは、個人をオンラインコミュニティにつなげることができ、このコミュニティを介して、ユーザは、例えば、他のユーザと通信する、再生リストを構築する、再生リストを共有する、他の人々に曲を示唆する、プロモーションに参加する、ターゲット広告にリンクすることなどができる。さらに、ジュークボックスおよび/または遠隔アプリケーションは、ユーザが、例えば、セル電話機、パーソナルデジタルアシスタント、MP3プレーヤなどのポータブルデバイスに情報をダウンロードすることを可能にすることができる。一部の実施形態は、ジュークボックス上、および/または遠隔インターフェースを介するユーザのアクションは、そのユーザに特有の、かつ/またはそのユーザによってカスタマイズ可能なアバターによって表されることが可能である。

【背景技術】

【0003】

本出願は、「Digital Downloading Jukebox with Enhanced Communication Features」という名称の9月24日に出願した米国特許出願第11/902,658号の利益を主張する。

【0004】

ジュークボックスは、数十年にわたって存在しており、便利で有利な仕方、再生のために所望される曲を選択する能力をユーザに提供する。ジュークボックスは、レストランやバーなどの商業施設において、ある料金で、それらの施設の利用客に、所望される曲をオンデマンドで提供するために、都合よく提供されてきた。過去数年間に、すべての関係者に、ジュークボックスの運用の大幅な改良をもたらす新世代のジュークボックスデバイスが、利用可能になっている。より具体的には、従来の独立型のレコード盤ジュークボックスおよびCDジュークボックスが、中央サーバによって制御され、中央サーバと通信するデジタルダウンロードジュークボックスによって取って代わられている。この新世代ジュークボックスシステムの例が、参照により開示の全体が本明細書に組み込まれている、特許文献1において示される。この新世代のジュークボックスシステムの有力な販売業者が、（公序良俗違反につき、不掲載）である。

【0005】

図1は、デジタルダウンロードジュークボックスシステム10（以降、単に「ジュークボックスシステム」と呼ぶ）の例示的な実施形態の概略を示す。図1に示されるとおり、ジュークボックスシステム10は、中央サーバ12からダウンロードされることが可能な、オーディオコンテンツ（通常、音楽）のマスタライブラリとともに、または、代替として、オーディオビジュアルコンテンツ（通常、音楽、および関連するビデオまたはグラフィック）を含む中央サーバ12を含む。また、ジュークボックスシステムは、一連の遠隔ジュークボックスデバイス16、16a～16fも含む。これらのジュークボックスデバイスのそれぞれは、一般に

、バー、レストラン、クラブ、または所望される他のロケーションに配置され、硬貨、紙幣、クレジットカード/デビットカードなどの、ユーザからの支払いを受け取ること、および再生のためにユーザによって1つまたは複数の曲が選択されることに応答して、曲を再生するように動作可能である。代替の実施形態では、音楽サービスの代金は、ロケーションによって契約ベースで支払われ、選択された音楽は、エンドユーザには無料である。ジュークボックスデバイス16は、ユーザに情報を提示し、ユーザが、その情報から曲を選択することを可能にするスクリーン18と、選択された曲を再生するオーディオシステム20とを通常、含む。また、スクリーン18は、曲関連のビデオまたはグラフィックを表示するために使用されることも可能である。また、スクリーン18は、ジュークボックスに顧客を引き寄せるためにジュークボックス自体に関する広告を表示する、他のタイプの広告を表示する、かつ/または他の任意の所望される情報を表示するのに使用されることも可能である。

【0006】

ジュークボックスデバイス16(ときとして、本明細書で単に「ジュークボックス」と呼ばれる)は、例えば、インターネットなどの通信ネットワーク14を介して、中央サーバ12と通信するように動作可能である。ジュークボックス16は、ジュークボックス上で再生された特定の曲に関してサーバ12に情報を提供するために、サーバ12と定期的に通信する。すると、中央サーバは、この情報を、各ジュークボックス上で再生された曲に関して支払われるべき適切な印税および/または他の支払いを算出するために使用する。

【0007】

このため、この新世代のジュークボックスの1つの重要な利点は、サウンド再生の権利および/または他の該当する音楽の権利が、より正確で、信頼できる仕方で厳守されて、アーティストまたは音楽所有者に適切な印税が支払われることを確実にすることが可能であることである。また、中央サーバ12は、ジュークボックス上に、そのロケーションにおける特定の顧客に基づいて、適切な曲、または最も人気のある曲が保持されることを確実にするために、新たな曲をジュークボックス16に提供することもできる。このため、各ジュークボックス上で利用可能な曲は、各ジュークボックスロケーションにおいて顧客が一般に要求する曲および/または音楽タイプを提供するように、中央サーバとの通信を介してカスタマイズされることが可能である。前述した特許文献1において説明されるとおり、中央サーバは、例えば、新たなフィーチャ、または改良されたフィーチャを提供するようにするなど、受信ボードの動作を変換するために、ジュークボックス上のオペレーティングソフトウェアを更新するように有利に使用されることも可能である。このため、この新世代のジュークボックスの別の重要な利点は、曲(または他のオーディオコンテンツおよび/またはビジュアルコンテンツ)、ならびにジュークボックス自体の動作が、誰か(地域責任者など)が直に、ジュークボックスの改修を行うようにさせる必要なしに、所望に応じて、遠隔で変更されることが可能なことである。代わりに、そのような更新は、中央サーバ12を使用して行われることが可能である。

【0008】

前述したとおり、ジュークボックスデバイス16はそれぞれ、曲、および関連するビデオデータ/グラフィックデータ(存在する場合)、ならびにジュークボックス上の再生に関する所望される他の任意のグラフィック情報を格納する、ハードドライブなどの大容量記憶デバイスを含む。ジュークボックスの大容量記憶デバイスは、通常、中央サーバ12の記憶デバイスと比べて、限られた記憶容量を有する。その結果、任意の一時点において、中央サーバ上に格納された曲のほんの一部だけが、通常、ジュークボックスの大容量記憶デバイス上に格納される。限られた記憶容量をジュークボックス上に有すること、および/またはジュークボックス上に格納される曲の数を制限することに関して、データのセキュリティ、またはジュークボックス内部の限られたスペースなどの、他の理由も存在することが可能である。例えば、自立型のモデルと比べてサイズが小さいように設計されている、壁面取り付け型のジュークボックス上で、物理的スペースが限られている可能性がある。前述したとおり、ジュークボックス上の曲は、中央サーバとの通信を介して変更されるこ

とが可能であるが、通常、任意の1つのジュークボックスは、任意の一時点において、中央サーバによって保持されている完全な曲ライブラリの比較的小さいサブセットだけを格納する。

【0009】

ジュークボックスが生む収益を最大化するため、長期にわたって、最も所望される曲をジュークボックス上で利用可能にすることが、重要である。顧客が、顧客の好む曲をジュークボックス上で見つけることができない場合、ジュークボックスの使用は(そのため、生じる収益も)、劇的に低下する。他方、任意の特定のロケーションにおける顧客が、ジュークボックス上で何を再生することを所望するかを、前もって正確に予測することは、不可能である。実際、顧客が、中央サーバ上に存在するが、ジュークボックス上に現在、存在しないある曲を選択している多くの場合が、生じる可能性が高い。その結果、ジュークボックスは、最大限度まで享受されず、使用されない可能性がある。この問題に対処し、収益を増やすため、ジュークボックスシステムは、過去には、ユーザが、ジュークボックスから、中央サーバ上の曲を検索し、追加料金で、所望される曲の、中央サーバからジュークボックスへの即時のダウンロードを要求することができるようにするフィーチャが、提供されてきた。このフィーチャは、ユーザが、ジュークボックスを使用して、中央サーバによって保持される曲のマスタライブラリの中の任意の曲を、その特定の曲が、ジュークボックス自体の大容量ストレージの中に現在、格納されているか否かにかかわらず、再生することができるようにする。このため、ユーザはまず、ジュークボックスのローカルストレージ上で所望される曲を探し、次いで、所望される場合、中央サーバ上で、所望される曲をさらに探すことができる。ジュークボックスデバイスは、ジュークボックスのローカルストレージから直接の標準再生とは異なる、中央サーバからの曲の即時のダウンロードおよび再生に関して、追加料金(1クレジットの代わりに5クレジットなどの)を、通常、請求する。

【0010】

しかし、即時のダウンロードのフィーチャが抱える1つの問題は、実施するのに、中央サーバとの即時の高速の接続を有することが望ましいことである。さらに、中央サーバおよびネットワークは、このフィーチャが適切に機能するのに、そのような要求を信頼できる効率的な仕方では扱える準備ができており、かつ、そうすることができなければならない。これらの要求は、必ずしも満たされることは可能でなく、その結果、このフィーチャの実施は、限られている。例えば、ジュークボックスを有する多くのロケーションは、高速の接続(DSLなどの)を有さず、代わりに、ダイヤルアップモデム接続を使用する。ダイヤルアップ接続に依拠するジュークボックスは、一般に、サーバと定期的に通信し、ユーザが、曲を即時にダウンロードすることを許さないようにしか設計されていない。しかし、そのようなジュークボックスは、ダイヤルアップ接続が行われると、ある曲が、後の時点でダウンロードされるように、ユーザが投票することを可能にしていた。これは、もちろん、曲を即時にダウンロードすることができることほど、ユーザに満足の行くことではない。ネットワークまたはサーバが、トラフィック、動作不良などのために、現在、ダウンロードに利用できない場合、このダウンロードフィーチャに関連して、他の問題が生じる可能性がある。

【0011】

ほとんどのユーザに関して、ジュークボックスとの関係は、再生すべき曲の選択までしか及ばない。時たまのユーザは、迅速に選択を行い、かつ/または特定の曲を探し出すことを困難に感じる可能性がある。しかし、いつものユーザは、しばしば、同一の曲を選択し、同一の選択をいつも行わなければならないことが苛立たしくなる可能性がある。両方の問題とも、気後れ、苛立ちなどから、収益の損失をもたらす可能性がある。

【0012】

このため、例示的な実施形態のあるフィーチャは、ジュークボックスユーザのコミュニティ全体との関係を作り出すさらなるフィーチャを考慮して、ジュークボックスと利用客との間の関係を作り出すことである。そのようなシステムは、利用客が、利用客のサービ

スをカスタマイズし、利用客の友人と通信することができるようにすることにより、利用客の所有感覚も生じさせながら、ジュークボックスと利用客との間で信頼関係を確立することができる。そのような関係、および所有感覚は、気後れ、苛立ちなどが小さくなるようにしながら、ジュークボックス、および関連するサービスを使用することの便利さを大幅に高めることができる。

【0013】

本明細書で説明されるいくつかの例示的な実施形態において、ジュークボックスは、音楽サービス、ならびにユーザの体験とともに、ロケーションマネージャの収益も高める他のサービスのための中心にされることが可能である。例えば、ジュークボックスは、ユーザが、ユーザの友人と接触しようとすることができる所与のロケーションにおける接触ポイントとなることが可能である。例えば、ユーザは、ジュークボックスを使用して、集まりを呼びかけること、ジュークボックスシステムを介して音楽を共有すること、または混雑したクラブで離れ離れになった場合、ジュークボックスで落ち合うことができる。同様に、ジュークボックスは、例えば、所与のロケーションにおける、そのロケーションのマネージャが、顧客と接触しようとすることができる接触ポイントとなることが可能である。例えば、マネージャは、サービス品を宣伝すること、特別イベントについて常連客に知らせることなどができる。そのような通信は、ユーザによって、ロケーションマネージャによって、またはジュークボックス自体によって開始されることが可能である。

【0014】

ジュークボックスユーザが、ロケーションを離れると、その音楽とのユーザの結び付き、より詳細には、ジュークボックス自体との結び付きは、通常、終了する。ユーザは、例えば、いずれのユーザが、あるタイプの音楽、またはある曲を好むか、ユーザが、いつ音楽を聴くか、ユーザが、どれだけ頻繁に音楽を聴くかなどの、大量のデータをジュークボックスに提供する。従来のジュークボックスでは、この情報は、関与するユーザにさらなるフィーチャおよび/またはサービスを提供するように活用されない。このため、収集された情報は、通常、無駄になり、ユーザは、つかの間の所有感覚だけしか体験しない。ユーザは、例えば、ある特定の夜遊びの体験を再現することができない。

【0015】

また、所有経営者も、ジュークボックスユーザが、所有経営者の施設を離れると、ユーザとの結び付きが消えるため、損をする。このように、所有経営者は、顧客が戻るまで、顧客を失い、従来の、しばしば、無視される、大量広告の方法に頼ることを余儀なくされる。この広告形態は、通常、特定性に欠け、ユーザの個人化された帰属感覚を反映しない。さらに、所有経営者は、例えば、所有経営者が、特に収益性の高い夜を経験した特定のイベントを再現することもできない。

【0016】

オンラインコミュニティは、通常、例えば、インターネット上で、異種のユーザを仮想空間において結び付けようとして現れている。これらのコミュニティは、潜在的に、幅広いフィーチャおよびユーザへの絶え間ないアクセスを一様に提供する。しかし、従来のオンラインコミュニティは、単に仮想的であり、つまりユーザがコンピュータから離れると接続は、まさしく文字どおり切断される。このため、ユーザは、この幅広い、多数のフィーチャにアクセスし、ユーザの体験をカスタマイズし、例えば、習性 (habits) を購入すること、および見ることを備えたオンラインパーソナリティを構築することさえできるが、体験は、つかの間である。

【0017】

しかし、オンラインコミュニティを介してジュークボックスユーザを結び付けることにより、ジュークボックスに提供される情報を使用して、ユーザに様々なフィーチャおよびサービスが提供されることが可能である。ジュークボックスは、数千の集会ポイントに存在しているので、オンライン仮想コミュニティの他に数千の物理的ロケーションにおいて、結び付きが存在する。このため、ジュークボックスによって収集されたデータを活用すること、およびユーザをオンラインコミュニティに結び付けることにより、現実と仮想の

混合のコミュニティが、確立されて、例えば、長く続く所有感覚が生み出され、潜在的にすべてのユーザに関してサービスが個人化され、ユーザを互いに引き寄せ、音楽によって結び付けられた活気のある社会的に活動的なユーザコミュニティを確立することなどが可能である。所有経営者も同様に、例えば、顧客が、所有経営者のロケーションから離れている間に、顧客と結び付いて、顧客を所有経営者のロケーションに引き寄せることなどによって、さらなる商売へのパイプラインを実現することにより、利益を得ることができる。

【0018】

要するに、関連する遠隔アプリケーションを有するジュークボックスは、例えば、ユーザによるジュークボックスの認識を変えることができる。ジュークボックスは、バーにおいてだけアクセス可能な独立型のデバイスでなくてもよい。むしろ、本発明は、ユーザが、ジュークボックスを、例えば、メディアセンタ、集会ポイント、オンラインコミュニティへのポータルなどと考えることを可能にする。

【0019】

さらに、Web対応のポータブルデバイス(例えば、セル電話機、パーソナルデジタルアシスタントなどの)の登場とともに、ユーザは、どこであれ、ユーザが行くところに、仮想コミュニティを持っていくことさえできる。ユーザは、潜在的に、ポータブルデバイスを使用して、例えば、ユーザが、自宅、車内、バーなどで、ある曲を耳にした際などの、任意の時点で音楽を直接にダウンロードすることができる。

【0020】

いくつかの例示的な実施形態において、ジュークボックス上、および/または遠隔インターフェースを介するユーザのアクションは、そのユーザに特有の、かつ/またはそのユーザによってカスタマイズ可能なアバターによって表されることが可能である。ジュークボックスは、ユーザに代行してアクションを行い、かつ/またはユーザのアクションを表現するアバターアクションプログラミングされた論理回路(例えば、ハードウェア、ソフトウェアなどの任意の組合せ)を含むことが可能である。そのようなアクションは、オーディオおよび/またはビデオで曲を紹介すること、音楽と一緒に歌い、かつ/または踊ること、オンラインインタラクティブアクション(例えば、メッセージ、ホスティング、ファイル転送など)にマークを付けることを含むことが可能である。このため、ユーザアバターは、所有感覚および/または帰属感覚をさらに高めて、ユーザをジュークボックスに引き寄せることが可能である。さらに、アバターは、個人ユーザ、常連の利用客、および/またはある地域への新来者の楽しさをさらに高めることができる。

【特許文献1】米国特許第6,308,204号明細書

【発明の開示】

【発明が解決しようとする課題】

【0021】

ユーザ、ロケーションオーナー、および運用者の様々なニーズに応える改良されたジュークボックスシステムの必要性が、存在する。本発明は、以上、およびその他の問題に対処し、そのようなジュークボックスシステムのための強化された機能を提供するように設計される。

【課題を解決するための手段】

【0022】

上記目的を達成するため、本発明の一つは、オーディオビジュアル配信ネットワークに接続された1つまたは複数のデジタルジュークボックスに再生リストを配信する方法であって、周辺デバイス上のコンピュータ可読記憶媒体の中に格納された周辺再生リストを有する該周辺デバイスを、コネクタを介してデジタルジュークボックスに接続するステップと、前記デジタルジュークボックスのために前記周辺再生リストを取り出すステップと、前記周辺再生リストをジュークボックス再生リストに変換するステップと、前記ジュークボックス再生リストを、少なくとも前記ジュークボックス上で公開するステップとを含むことを特徴としている。

本発明の以上、およびその他の特徴、目的、および利点は、添付の図面と併せて読まれると、例示的な実施形態の以下の詳細な説明の精査により、さらに理解されよう。

【発明を実施するための最良の形態】

【0023】

次に図面を参照すると、図2は、改良されたジュークボックスシステム10'の例示的な好ましい実施形態のブロック図を示す。ジュークボックスシステム10'は、中央サーバ12、通信ネットワーク14、および遠隔ジュークボックスデバイス16、16a~16fを含め、図1に示され、前述したのと同様の要素を含む。しかし、ジュークボックスシステム10'は、ジュークボックスデバイス16、16a~16fのそれぞれに接続されたローカルサーバ22、22a~22fをさらに含む。中央サーバ12は、曲(および/または他のコンテンツ)のマスタライブラリを含む。ジュークボックスデバイスのそれぞれは、ジュークボックスのローカル記憶デバイス上にマスタライブラリのサブセットを含む。中央サーバを使用して、ジュークボックスデバイスの使用を最大化する意向で、ジュークボックスデバイスの各デバイス上の曲のサブセットの使用を監視し、更新することによって、ジュークボックスデバイスのコンテンツが個々に管理されることが可能である。中央サーバ12は、再生される曲に関する印税計算および支払いの目的で、ジュークボックスのそれぞれからデータを定期的に受信する。ジュークボックスデバイスは、ダイヤルアップモデムまたはブロードバンドモデム(例えば、DSL、ケーブル、無線ブロードバンド、または衛星)などの、任意の適切な仕方でネットワークに接続することができる。通信ネットワーク14は、中央サーバ12からジュークボックス16にデータ(例えば、オーディオビジュアルデータ)を配信することができ、データが、ジュークボックス16から中央サーバ12にアップロードされることを可能にする任意の適切なネットワークであることが可能である。

【0024】

曲(および/または他のデータ)は、好ましくは、知られている技術を使用して、セキュリティおよび帯域幅のために、ジュークボックスに曲を送信するのに先立って、中央サーバ12によってデジタル化され、圧縮され、暗号化される。その後、曲は、ジュークボックスによって、ジュークボックス上で格納し、再生するために、伸張され、復号化される。このため、ジュークボックスのそれぞれは、そのジュークボックス上で再生するためのデジタル化された曲のライブラリをデータベースの中に保持し、ライブラリは、中央サーバによる通信を介して変更される、または更新されることが可能である。また、ジュークボックスは、好ましくは、ジュークボックスデバイス16のディスプレイ18上で表示されることが可能なイメージ(例えば、静止イメージおよび/または動画ビデオイメージ、および/またはグラフィックイメージ)を構成するデータを受信して、格納することもできる。本発明の1つの例示的な実施形態では、ジュークボックスデバイスは、前述したにおいて説明したのと同様の構造および動作を有する。このため、ジュークボックスデバイス16はそれぞれ、好ましくは、メインCPUやオーディオDSPなどの1つまたは複数のマイクロプロセッサと、曲および/または他のコンテンツを格納するための、ハードドライブなどのメモリと、ビジュアルアイテムを表示するディスプレイと、オーディオを提供するためのオーディオ構成20と、ジュークボックスが、通信ネットワーク14を介して中央サーバ12と通信することができるようにするための通信システムと、ジュークボックスの動作を制御する、好ましくは、マルチタスクオペレーティングシステムを含むオペレーティングソフトウェアとを含む。また、オペレーティングソフトウェアは、好ましくは、例えば、前述した特許文献1において説明されるとおり、中央サーバ12との通信を介して更新可能である。ジュークボックス16は、顧客が、好都合な仕方でジュークボックスデバイスの使用の代金を支払うことができるようにするために、硬貨、紙幣、および/またはクレジットカードインプットデバイスなどの、1つまたは複数の支払いデバイスをさらに含む。スクリーン18は、好ましくは、ユーザが、スクリーンに触れることによって選択を入力することを可能にするタッチスクリーンである。

【0025】

各ジュークボックスデバイスは、一実施形態では、ジュークボックスデバイスによって

アクセスされることが可能なローカルサーバ22を有する。ローカルサーバは、イーサネット（登録商標）、または他のタイプのローカル接続を使用してジュークボックスデバイスに接続される。別の実施形態では、ローカルサーバは、別個のハードウェアデバイスではなく、単にジュークボックスのハードドライブの論理的拡張（例えば、パーティション、ディレクトリ、または領域）であってもよい。ローカルサーバ22はそれぞれ、好ましくは、中央サーバ12によって保持される音楽レコーディングのマスタライブラリのミラーコピーを含む。ローカルサーバ22には、ローカルサーバおよびジュークボックスデバイスをジュークボックス販売業者またはジュークボックス運用者に出荷するのに先立って、ジュークボックスネットワークを所有し、かつ/または管理するエンティティによって、マスタライブラリがロードされることが可能である。もちろん、しばらくすると、ローカルサーバは、中央サーバが、好ましくは、追加の曲、または新たな曲で絶えず更新されることから、中央サーバとはや、完全には一致しなくなる。このため、ローカルサーバ22も、好ましくは、定期的に更新されて、中央サーバ12上のライブラリとの一致を維持する。この更新は、例えば、ダイヤルアップモデムまたはブロードバンドモデムを使用してローカルサーバ22と接続されたジュークボックスデバイスとの通信を介して、例えば、中央サーバ12によって行われることが可能である。代替として、更新は、ローカルサーバのコンテンツを更新する目的で、地域責任者または他の個人によってジュークボックスまたはローカルサーバに直接に接続されることが可能な、更新ツールを使用して直に行われることも可能である。ポータブルツールは、将来の更新のためにジュークボックスシステムの所有者に戻されて、再使用されることが可能な、ハードドライブなどの取り外し可能な記憶媒体を含むことが可能である。ツール自体は、ジュークボックスシステムの所有者から更新された取り外し可能な記憶媒体を受け取ると、特定のジュークボックスを使用のために保守することを担当する運用者、または他の個人によって保持されることが可能である。

【0026】

セキュリティ上の理由で、ローカルサーバ22は、好ましくは、ローカルサーバ22上に格納される任意の1つの曲を構成するデジタルデータのすべては含まない。さらに、ローカルサーバ上にある曲の部分は、暗号化される。ジュークボックスデバイス16は、ローカルサーバ上の各曲の欠けている部分を含み、ジュークボックスが、ローカルサーバのコンテンツ、およびジュークボックスデバイス上のメモリに基づき、完全な曲を組み立てることを可能にする。ジュークボックス上に配置された、欠けているデータは、曲を復号化するために必要とされる。例えば、各曲に関するデータの単一のブロック（または他の小さい部分）が、ローカルサーバ上で欠けているが、ジュークボックスデバイス上に存在することが可能であり、暗号化は、その欠けているブロックに基づくことが可能であり、ブロックごとに進められることが可能である。このため、ブロックのいずれも、先行するブロックを獲得すること、および/または復号化することなしに、復号化され得ない。このフィーチャは、重要なセキュリティを提供し、ローカルサーバ上の曲の盗用、または他のタイプの許可のない使用または複製を防止または抑止する。このため、この実施形態では、各ローカルサーバは、前述した復号化が、適切に実行されることが可能であるように、ある特定のジュークボックスデバイスに限定して割り当てられなければならない。

【0027】

好ましい例示的な実施形態によれば、ローカルサーバはそれぞれ、中央サーバ12に個々に登録され、識別されて、中央サーバが、各ローカルサーバを個々に管理し、監視することができるようにすることも可能である。同じことが、ジュークボックスデバイス自体にも当てはまり、すなわち、ジュークボックスデバイスもやはり、好ましくは、中央サーバに登録されて、ジュークボックスデバイスも、中央サーバによって個々に監視され、管理されることが可能であるようにすることも可能である。以上の説明から理解されたとおり、ローカルサーバは、ローカルサーバのコンテンツが、ジュークボックスデバイスによってアクセスされて、ジュークボックスデバイス自体の上で利用可能でない追加のサービス（追加の曲を提供することなどの）が提供されることを可能にすることにより、ジュークボックスシステムの重要で、有利な部分となる。後段で説明されたとおり、中央サーバおよ

び/またはストレージ容量自体の曲ライブラリを有利に使用して、料金ベースの住宅向けジュークボックスおよび商用ジュークボックスなどの他のジュークボックス、および/または他の料金ベースの機器にサービスが提供されることが可能である。ローカルサーバの1つの好ましい用法は、ジュークボックスデバイスに即時の曲ダウンロードフィーチャを提供することであることが可能であり、図3～図6の例示的なスクリーンショットを参照して、以下に詳細に説明する。

【0028】

図3は、ジュークボックスデバイスのタッチディスプレイ上に表示された曲選択スクリーン30に関する例示的なスクリーンショットを示す。図3で見取ることができるように、好ましくは、顧客に表示される最初の選択スクリーンである、この選択スクリーンは、ジュークボックスデバイスのメモリの中に格納された曲に関する様々なアルバムカバーのグラフィック表現32を含む。アルバムカバーは、アルファベット順に示され、仮想スライドバー33を使用して、利用可能なアルバムの間でスクロールが行われることが可能である。また、上向き矢印と下向き矢印(34と35)が、利用可能なアルバムの間で1つずつ進むために提供される。また、「Now Playing」ボタン36が、ジュークボックス上で現在、再生されている曲(存在する場合)に関する情報を示すために提供される。また、「Top Ten」ボタン38が、ジュークボックス上の上位10の最も人気のある曲のリストを示すために提供される。また、「True Central」(タッチチューンミュージックコーポレーションの商標)ボタン39が、提供され、次に、このボタン39の機能を、図4に関連して詳細に説明する。

【0029】

ユーザは、アルバムカバーの表示の中で関心対象のアルバムが見当たらない場合、または何らかの理由で、ジュークボックスデバイス上に存在しない、利用可能な曲を検索することを所望する場合、ユーザは、「Tune Central」ボタン39を選択することができる。「Tune Central」ボタンが押されると、ジュークボックス上の表示は、図3の表示から図4の表示に変更される。図4の例示的なスクリーンショットは、ジュークボックスデバイスに接続されたローカルサーバ22上で検索が実行されることを可能にする検索スクリーン40を示す。このスクリーン40は、検索要求を入力する際に使用するための仮想キーボード42を提供する。検索は、関連するボタン47に基づき、アルバム、アーティスト、曲、またはジャンルもしくはテーマ(すなわち、ユーザが、ある特定の曲を見つけるのを助ける、好ましくは、人気に基づく、分類された曲リスト)によって行われることが可能である。検索が、タイプ入力されると、ユーザは、「Search」ボタン44に触れ、ローカルサーバのコンテンツの検索が開始される。仮想キーボードからの入力は、「Clear」ボタン46を使用してクリアされることが可能である。

【0030】

ジャンルは、例えば、「ポップ」、「ロック」、「カントリー」、「ラップ」、「R&B」、「ブルース」などを含むことが可能である。テーマは、例えば、「クリスマス」、「アイルランドの」、「誕生日」などを含むことが可能である。一部の例示的な実施形態では、これらのリストは、サーバから取り出されることが可能であるのに対して、他の例示的な実施形態では、これらのリストは、例えば、いくつかのパラメータに基づき、動的に構築されることが可能である。さらに別の実施形態では、サーバは、例えば、ロケーション、音楽タイプなどの基準に基づき、いずれのリストを使用可能にするか、またはジュークボックスに送信するかを選択することができる。システムは、ダウンロードされる音楽に関するジャンルおよび/またはテーマだけを表示することも可能である。しかし、システムは、すべてのジャンルおよび/またはすべてのテーマを表示して、ローカルジュークボックスに利用可能でない他の曲のダウンロードを可能にすることもできる。また、以下に詳細に説明する図4Cは、同一のスクリーン上でジャンルとテーマを示すが、他の実施形態は、ジャンルとテーマを別々に提示することが可能であることも理解されよう。

【0031】

図4Cは、1つの例示的な実施形態による、ユーザに利用可能なジャンルおよびテーマの非限定的なリストを示す例示的なスクリーンショットである。スクリーンの左側に配置さ

れたボタン450は、ユーザに利用可能なテーマに対応するのに対して、スクリーンの右側に配置されたボタン460は、ユーザに利用可能なジャンルに対応する。ボタン470は、この選択機能を終了する。これらのボタンの他の配置も利用可能であること、ならびに提供されるテーマおよびジャンルのリストは、単に非限定的な例示のためであることが理解されよう。

【0032】

ジャンルおよびテーマの検索と同様に、ユーザは、例えば、パーソナルミュージックアシスタントを使用して曲を検索することができ、これを行うための例示的なプロセスが、図4Aに示される。好ましくは、パーソナルアシスタントボタンを押した後(ステップ402)、ジュークボックスは、ユーザが既に識別されているのではない場合、ユーザを識別するある情報を求める(ステップ404)。そのような情報は、例えば、年齢(または生年月日)、好ましいスタイル、背景、出生地、またはユーザのプロファイルを生成するのに使用されることが可能な他の情報を含むことが可能である。次に、ジュークボックスは、好ましくは、そのプロファイル情報を、例えば、その特定のジュークボックス、その特定の施設、または全国データベースおよび推奨される曲からの類似したプロファイル(ステップ406)を有する他のユーザによって行われた選択と比較することができる(ステップ408)。例えば、ジュークボックスは、1960年に生まれたカリフォルニア州出身の男性ユーザに、「The Doors」による曲を示唆することが可能である。すると、ユーザは、そのリストから曲を選択する、または新たな検索を開始することができる(ステップ410)。

【0033】

さらに、図4Bに示されるとおり、識別子を入力する代わりに、パーソナルミュージックアシスタントは、例えば、クレジットカード、または事前プログラミングされたサイト特有のIDカードが、ジュークボックスのスロットを通された後などの、他の仕方ユーザを認識することも可能である(ステップ422)。好ましくは、パーソナルミュージックアシスタントは、ユーザによって行われた選択のリストを保持する。ユーザの選択リストは、例えば、ローカルジュークボックス端末装置上、あるサイトの中央ジュークボックスサーバ上、遠隔サーバ上、またはIDカード上に保持されることが可能である。パーソナルミュージックアシスタントは、ユーザを認識した後、例えば、特定のユーザが楽しむアーティストによる曲(ステップ426)、ユーザによって頻繁に再生される曲(428)、ユーザによって最近、聞かれていない曲(430)などにに基づき、曲を推奨することも可能である。

【0034】

さらに、例えば、好ましい顧客、または多数のクレジットを有する顧客を認識するパーソナルミュージックアシスタントが、ジュークボックスを、その特定のユーザが、より楽しむことができるジュークボックスに変形させることも可能である。関連するフィーチャにおいて、数名の利用客が、例えば、特別な行事のためにジュークボックスを「賃借り」することが可能である。したがって、利用客は、ある金額を支払って、ジュークボックスを、ある特定のタイプのジュークボックスに変形させることもできる。例えば、ある特定のロケーションにおいてパーティ(例えば、誕生パーティ)を開く利用客が、ジュークボックスを賃借りして、そのジュークボックスを、その利用客の好みの曲だけを許すジュークボックスに変形させることが可能である。別の関連するフィーチャにおいて、利用客が、ジュークボックスを排他的に賃借りして、例えば、別のデバイス(例えば、ジュークボックスシステムに動作上、接続されたWebまたは他の通信ネットワーク)を介して再生リストをプログラミングすること、またはパスワードなどを介してジュークボックスへのアクセスを制限することにより、他の人々が曲を選択するのを防止することも可能である。クレジットが、例えば、ユーザによって購入されること、または、例えば、施設において飲料または記念品を購入すること、または常連のリピーター客であることの報酬としてユーザに与えられることが可能である。このため、パーソナルミュージックアシスタントは、いずれの曲を選択すべきかを知るのにユーザにかかる直接のプレッシャーを取り除きながら、曲を選択することを、より楽しむことができる、動的で、応答性のあるプロセスにする仕方で実施されることが可能である。

【0035】

スクリーン40から検索が開始されると、スクリーンは、図5に示されるスクリーンに変わり、その検索の結果を表示する。図5に示されるとおり、検索の結果が、リストアップされる。より詳細には、この実施例では、検索要求を満たす曲のリストが、リストアップされる。また、リストは、検索が、アルバムベースである場合、アルバムによることも可能である。ユーザは、スライドバー53を使用して検索結果をスクロールすることができる。また、ユーザには、現在のクレジット数の表示55、およびローカルサーバからジュークボックスデバイスに曲をダウンロードするのに要求されるクレジット数の表示56も表示される。ユーザは、「Back」ボタン57に触れることにより、前のスクリーンに戻ることができる。ユーザが、検索リストからある曲を選択し、次に、「Get It Now」ボタン54に触れた場合、ジュークボックスは、ジュークボックス上で再生するためにローカルサーバからジュークボックスに、選択された曲を即時にダウンロードするように動作可能である。ダウンロードされた曲は、ジュークボックス上で再生されるように、選択されているが、再生されていない他の曲(存在する場合)と一緒にキューに入れられることが可能である。この実施例では、ダウンロードは、ジュークボックス自体のストレージからの通常の選択のように1クレジットではなく、5クレジットを要する。ダウンロードされた曲は、再生されると、ジュークボックスデバイスから(曲ダウンロードに関連してローカルサーバからやはりダウンロードされた、アルバムカバーグラフィックなどのグラフィックデータと一緒に)削除されることが可能である。しかし、好ましくは、曲は、「スマートキャッシュ」の中に保持されて、その曲が、削除されるべきかどうか、かつ/またはいつ削除されるべきかが決定される。例えば、曲は、ジュークボックスが、その曲の再生を監視する一方で、事前定義された期間(例えば、1日、1週間、2週間など)にわたってキャッシュされることが可能である。例えば、事前定義された期間にわたって、再生の回数が、ある閾値を超えている場合、その曲は、保持されることが可能であり、再生の回数が、その閾値を下回っている場合、その曲は、削除されることが可能である。その曲は、本発明の特定の形態に依存して、直ちに利用可能な曲のマスタリストに追加されることも、追加されないことも可能であり、例えば、「スマートキャッシュ」の中の曲は、実際には、例えば、保護された領域の中でジュークボックス上に格納されているながら、ユーザに直ちに利用可能でないように見えることが可能である。このように、ユーザは、「Tune Central」ボタンの使用を介して、中央サーバ12と連絡をとる必要なしに、レコーディングのマスタライブラリから任意の曲を、ジュークボックス上に一時的に獲得するオプションを有する。その結果、ジュークボックスは、ユーザに、より楽しむことができる体験を提供する一方で、それによって生じる収益も高める。

【0036】

やはり、ユーザに、楽しむことができる体験を提供するのが、好ましくは、リアルタイムで「ホットヒット」を識別する中央サーバの能力である。好ましくは、新たな曲が、マスタカタログの中で利用可能にされることが可能であり、つまり、新たな曲は、ローカルサーバ上、または拡張されたメディアストレージ上に存在しなくてもよい。その場合、所与の地域(例えば、単一のサイト、またはサイトのグループから、州または国、地球規模の接続にまで及ぶ)において頻繁に再生される曲が、人気があると識別されることが可能である。これらの曲、つまり、「ホットヒット」は、好ましくは、個々のジュークボックスによってダウンロードされる、または個々のジュークボックスに送信されることが可能である。個々のジュークボックスは、好ましくは、「ホットヒット」のリストをリアルタイムで保持して、ユーザが、任意の所与の時点で最も人気のある曲の中で検索することを可能にする。代替として、ジュークボックスは、人気のある曲をダウンロードすることなしに、「ホットヒット」のリストを保持して、ダウンロード時間およびリソースを潜在的に節約することも可能である。その結果、ジュークボックスは、最も人気のある曲への容易なアクセスを提供することにより、ユーザに、楽しむことができる体験を提供することができる。このため、ジュークボックスは、あるジュークボックスによって生成された、複数のジュークボックスによって生成された、運用者によって指定された、自動的に、か

つ/または使用可能にされたジュークボックスのために特別に作成された「ホットヒット」の「ホットリスト」にアクセスを有することが可能である。この「ホットリスト」は、テーマおよび/またはジャンルの代替の役割をすることが可能であり、かつ/またはテーマおよび/またはジャンルと一緒に機能して、所与のテーマ内、および/または所与のジャンル内で最も人気のある(「ホットな」)曲を提供することが可能である。同様に、ユーザは、最近、再生された曲のリストから、ある曲を選択することもできる。そのようなリストは、個別のジュークボックス上に保持される(例えば、格納され、更新される)ことが可能であり、あるいは、任意の適切に構成されたジュークボックスデバイスがアクセスすることができ、かつ/または更新することができるように、中央ロケーションに保持されてもよい。

【0037】

図5Aは、ブロードバンド接続を有するジュークボックス上に「ホットリスト」を保持するための例示的なプロセスを示す。同一のプロセスは、異なるタイプの接続を有するシステムに関しても適用されることが可能であり、ただし、より遅い接続を介して曲をダウンロードするのに、より多くの時間およびリソースが使用される可能性があることに留意されたい。ステップ502で、マスタカタログからの曲が、サイトの中央サーバによって受信される。もちろん、曲は、ローカルジュークボックスの記憶媒体に格納されることも可能であることに留意されたい。ステップ504で、ジュークボックス端末装置を使用するユーザが、「Hot List」ボタンを選択する。「ホットリスト」が表示された後(ステップ506)、ユーザは、ある特定の曲を選択する、または新たな検索を開始することができる(508)。

【0038】

図6は、図3のスクリーン30からのアルバムカバーグラフィックにユーザが触れた際に表示される、曲選択スクリーン60の別の例示的なスクリーンショットを示す。このため、このスクリーンは、曲を選択する代替の(または通常の)方法を示し、曲は、ジュークボックス自体(ローカルサーバではなく)の記憶デバイスから直接に利用可能な曲のサブセットから直接に選択される。この例では、Joe Cocker's Greatest Hitsが、図3のスクリーンから選択されている。図6に示されるとおり、もたらされるスクリーン表示60は、選択されたアルバムグラフィック61と、そのアルバムに関してジュークボックス上で利用可能な曲のリスト62を示す。ジュークボックスは、ある特定のアルバムに関する曲のすべてを含むことも、含まないことも可能である。利用可能な曲は、必要な場合、スクロールバー63aおよび63bを使用してスクロールされることが可能である。ユーザは、「Play」ボタン65を介して、ジュークボックス上で再生するためにリストから曲を選択するオプションを有する。また、「Play Now」ボタン66が、曲の優先再生をユーザが選択して、その曲に、「Play」ボタン65を使用して選択された曲よりも高い優先順位を与えることができるようにするために提供される。この優先フィーチャは、好ましくは、通常の再生よりも多くの再生クレジットを要求する。表示67は、ユーザに利用可能なクレジット数を示す。ボタン64は、61で示されるのと同じアーティストに関する他のアルバムを示して、ユーザが、ある特定のアーティストに関するアルバムの中で、所望される曲を容易に探すことができるようにする。

【0039】

やはり図6に示されるとおり、ユーザが、図4に関連して説明されるとおり、ローカルサーバ上で、この同一のアーティストによる曲を検索することができるようにする「Tune Central」ボタン68が、表示される。つまり、ボタン68は、ローカルサーバを検索するために、図4の検索スクリーン40にユーザを連れて行く。すると、ユーザは、図4および図5に関連して前述したとおり、ローカルサーバを検索し、所望される場合、ローカルサーバから曲を選択することができる。このため、前述したとおり、ユーザは、ジュークボックススクリーンと対話する際のユーザの所望に応じて、様々なスクリーンにおいて便利で効率的な仕方でもローカルサーバにアクセスすることができる。

【0040】

図3～図6から見て取ることができるとおり、ユーザには、ジュークボックスデバイス自体の上に存在する曲を再生するオプション、または、代替として、効率的で、信頼できる仕方でダウンロードして、再生するために、ローカルサーバから曲を選択するオプションが与えられて、その結果、ジュークボックスシステム、特に、中央サーバからオンデマンドで音楽のダウンロードを迅速に、容易に、または確実に受信することができないシステムの動作を大幅に向上させる。図3～図6のスクリーンショットは、単に例示的であり、任意の適切なスクリーン構成を使用して、本明細書で説明される機能が提供されることが可能であることに留意されたい。さらに、ジュークボックス運用者には、ある特定のロケーションにおいてエンドユーザのアクセスを制限し、望ましくない音楽が再生されるのを回避するため、運用者スクリーン(図示せず)を介して、音楽のジャンル別またはスタイル別にフィルタを設定する能力が与えられる。例えば、これらのフィーチャは、好ましくは、ジュークボックス運用者および/またはロケーションマネージャによって定義されたスケジュールを介して利用可能であり、例えば、ハードロックまたはヘビメタル音楽が、午前11時前に再生されることをロケーションマネージャが阻止することを可能にする。同一のフィーチャが、運用者またはロケーションマネージャによって、再生リストを構成し、特定の日付および/または時刻に関して利用可能な音楽選択を設定するのに使用されることが可能である。例えば、運用者またはロケーションマネージャが、聖パトリックの日にアイルランド音楽に、聖バレンタインの日にラブソングに、夕食中にフランクシナトラ音楽に、音楽選択を限定することを所望することが可能である。

【0041】

さらに他の例示的な実施形態では、ロケーションは、「仮想DJ」を提供することも可能である。そのようなシステムによれば、準備された再生リストが、ロケーションのジュークボックスに提供される。好ましくは、再生リストは、例えば、会場、テーマ、ターゲットの聴衆などの、ロケーションおよび/または行われるイベントの詳細に合わせられることが可能である。さらに、一部の例示的な再生リストは、利用客を引き寄せるために有名なDJによって作成されることも可能である。DJは、居合わせていなくても、再生リストにDJの名前を付け、様々なロケーションに自身を関連付けることができる。再生リストは、ロケーションのニーズに応じて、数曲にわたる、または数時間に及ぶことが可能である。特別再生リストを使用することのサービスに特別料金が関連付けられることが可能であるので、そのような仮想DJイベント中に、硬貨が集められなくてもよいことを理解されたい。また、DJは、1回限りの使用のために特別再生リストを準備することも、反復可能な特別再生リストを提供することも可能であることを理解されたい。これらの反復可能な再生リストは、例えば、ジュークボックスによってダウンロードされること、または任意の適切な仕方でジュークボックスに提供されることが可能である。このため、有料で、ジュークボックス上で使用するようにサードパーティの再生リストを利用可能にすることにより、ジュークボックス運用者、ロケーションオーナー、およびよく知られているDJなどの再生リスト作成者などにまったく新たな収入源がもたらされる。再生リストが活性にされると、ジュークボックスの通常の機能は、好ましくは、再生リスト上の曲の再生中に停止される。

【0042】

図7は、例示的な実施形態によるローカルサーバ22を定義する電子要素のブロック図を示す。図7に示されるとおり、ローカルサーバ22は、CPU72(例えば、AMD Elan 800MHz)と、BIOSおよびOSを含むフラッシュメモリ(例えば、8MB)と、マスタ/スレーブハードドライブのペア(それぞれ、82、84、および86、88)、ハードドライブペアそれぞれのためのIDEコントローラ78および80のペアと、RAM16(例えば、32MB)と、ジュークボックスデバイス16との通信を制御するためのイーサネット(登録商標)コントローラと、これらの様々な要素を互いに接続する適切なバスを含む。もちろん、ローカルサーバ22の他の構成または配置が、使用されてもよい。ローカルサーバが、ジュークボックスおよび/または中央サーバによって一意に識別されて、登録されることを可能にするように、一意識別子が、ローカルサーバにおいて提供されてもよい。識別子は、例えば、フラッシュメモリ74の中に

配置されることが可能である。

【0043】

以上の本発明の説明から理解されたとおり、ローカルサーバの追加は、ジュークボックスシステムの一部であるジュークボックスデバイスの動作を大幅に強化する。しかし、ローカルサーバは、次に説明される他の利点およびフィーチャも提供する。

【0044】

ローカルサーバ22の集合が、中央サーバ12によって、関連するジュークボックス16を介して他のデバイスに音楽サービスを提供するように制御されることが可能な分散サーバのネットワークとして使用されることが可能である。例えば、ローカルサーバ、および関連するジュークボックスを使用して、ローカルサーバが接続され、割り当てられている特定のジュークボックスに曲サービスを提供することに加えて、専用の住宅向けジュークボックスデバイスまたは商用ジュークボックスデバイス(または他の適切なジュークボックスデバイス)に、要求された曲が配信されることが可能である。このため、分散サーバのネットワークは、住宅ロケーションまたは商業ロケーションにおける再生および/または格納のために、ユーザが、適切な料金を曲をダウンロードすることを可能にするタイプの住宅向けジュークボックスおよび商用ジュークボックスを実施するためのサポートネットワークを提供することができる。その結果、ジュークボックスシステム運用者は、ジュークボックスシステムを介して商用ジュークボックスおよび住宅向けジュークボックスを提供し、制御することができる。この実施形態では、ジュークボックスデバイスおよび/またはローカルサーバは、ブロードバンドモデムを使用してインターネット(または他の適切なネットワーク)に接続され、中央サーバの制御下で任意の専用の住宅向けジュークボックスデバイス(やはりインターネットに接続可能な)に曲ファイルを選択的に配信することができるソフトウェアを備える。中央サーバは、住宅向けジュークボックスから要求を受信し、ネットワーク上のトラフィックを解析することにより、要求された曲ファイルを(選択されたジュークボックスデバイスのメモリから、またはローカルサーバから)、ある料金を、またはその住宅向けジュークボックスに関する契約プランの下で、住宅向けジュークボックスにダウンロードするよう、選択されたジュークボックスデバイスに命令を与える。いくつかの例示的な実施形態では、要求された曲は、ジュークボックスにストリーミングされることが可能である。ストリーミングされるメディアは、専用サーバを送信元とすること、ストリーミングサーバのネットワークを送信元とすること、あるジュークボックスまたは複数のジュークボックス(ピアツーピアダウンロードまたはマルチピアダウンロードなどの)から送出されることが可能であることを理解されたい。

【0045】

本発明の別の例示的な態様によれば、ローカルサーバおよびジュークボックスデバイスは、中央サーバの制御下で、ジュークボックスと同一の(または近傍に)設置された、ゲームデバイスなどの、他のタイプのコイン式機器または支払いによってトリガされる機器に管理サービスを提供するのに使用される。つまり、ジュークボックスシステムは、好ましくは、同一のロケーションにおける他のダウンロードデバイスの機能を更新し、かつ/またはそのようなデバイスを管理するのに使用される。その結果、ジュークボックスは、あるロケーションにおけるすべてのダウンロード機器に関する「中央ハブ」となる。このフィーチャは、一実施形態では、単一のロケーションにおけるダウンロードデバイスのすべてを、ジュークボックスおよびローカルファイルサーバと一緒にネットワーク化することによって実現される。すると、中央サーバは、いずれのデバイスが、どのようなデータおよび/またはソフトウェアで更新されるべきかについての情報を、ジュークボックスに対する命令と一緒に、ローカルサーバにダウンロードすることができる。また、ジュークボックスデバイスおよび中央サーバは、ジュークボックスデバイスが管理している他のダウンロードデバイスからの情報を収集して、その情報を中央サーバに、報告/会計の目的でアップロードするのに使用されることも可能である。このため、ジュークボックスシステムの所有者/運用者は、他のコイン機会社の機器を管理し、かつ/または更新する目的で、他のコイン機会社に対するサードパーティサービスプロバイダの役割をすることができる。

【0046】

ローカルサーバによって提供される大量のメモリと、ローカルサーバが、よく制御されたネットワークにわたる数千のロケーションにおいて提供され、アクセス可能であるという事実とは、ジュークボックスシステムを、コイン機業界における様々な機能を実行するのに使用されることが可能な、強力なツールにする。ますます多くのコイン機製造業者が、内部ハードドライブを介してソフトウェアアップグレード可能なゲームに向かって進んでいる。これらの更新は、定期的に行われるが、これらのデバイスが、増えるにつれ、遠隔ロケーションから更新を確実に、効率的に実行することができるシステムの必要性がますます高まる。本明細書で説明するジュークボックスシステムは、ジュークボックスロケーションにおけるすべての適切な電子コイン式デバイスが、そのロケーションにおけるジュークボックスおよびローカルサーバを使用して中央サーバによって管理されることを可能にすることにより、この必要性を満たす。中央サーバは、ソフトウェア更新またはデータ更新をダウンロードし、それらの更新をローカルサーバ上に格納し、次に、それらの更新を、施設における意図される機器ユニットにディスパッチすることができる。このため、ジュークボックスシステムは、コイン機ビジネスにおける他の会社に対するサードパーティプロバイダの役割をして、ジュークボックスシステムの機能を大幅に強化することができる。

【0047】

例として、現在、米国内に約140,000のメリット社のコイン式カウンタトップデバイスが存在し、これらのデバイスのそれぞれは、ユーザが、有料でゲームなどを行うことができるようにする。これらのデバイスの多くは、新たなソフトウェアでアップグレードされることが可能なハードドライブを使用して動作する。メリット社は、CD-ROMを運用者に出荷することによって、これを行い、運用者は、その後、各ロケーションに車で行き、各マシンを手動で更新する必要がある。しかし、本発明によれば、あるロケーションにおけるすべての適切なコイン式機器が、そのロケーションに割り当てられたローカルジュークボックスおよびローカルサーバと接続される(直接に、または間接的に)。このことは、中央サーバが、任意のデバイスに関する意図されるソフトウェア更新を、どのようなデバイスが、どのようなソフトウェアでアップグレードされるべきかを識別する情報と一緒に受信することを可能にする。アップグレードサービスは、好ましくは、有料であり、ジュークボックスシステムに関するさらなる収益の流れをもたらす。この場合、中央サーバは、適切なデバイスにアップグレードをさらにダウンロードするアップグレード命令と一緒に、ソフトウェアをローカルサーバにダウンロードする。

【0048】

前述したとおり、ローカルサーバは、中央サーバの制御下で、ローカルサーバが割り当てられている商用ジュークボックスに、または住宅向けジュークボックスに曲がダウンロードされることを可能にする。さらに、ローカルサーバは、他のコイン式デバイスを管理する構内のネットワーク化されたアプリケーションのために使用されることが可能である。本発明の、これらの様々なフィーチャが、図8に示される。

【0049】

図8は、例示的な実施形態によって企図される完全なジュークボックスシステムネットワークのブロック図を示す。前述したとおり、システムは、通信ネットワーク14に接続された中央サーバ12と、関連するローカル音楽ファイルサーバ22a、22b、および22cを有する一連の商用ジュークボックス16a、16b、および1.6cと、ブロードバンドデバイス102a、102b、102cを介してネットワークに接続された一連の住宅向けジュークボックス100a、100b、および100cと、図8の右側に示される構内ネットワークとを含む。この構内ネットワークは、ルータまたはネットワーク、ハブ110を介してローカルファイルサーバ22dに接続されたジュークボックスデバイス16dと、ダーツゲーム104、ゴルフゲーム106、およびカウンタトップビデオゲーム108などの、いくつかのさらなるコイン式機器と、このローカルネットワークを通信ネットワーク14に接続するブロードバンドモデム112とを含む。図8

に示される、この例示的な構成では、本明細書で説明される機能のすべてが、本発明のジュークボックスシステムを介して実施されることが可能である。

【0050】

図9は、本発明の一実施形態によるマルチゾーンジュークボックスシステムに関する例示的な施設レイアウトを上から見た図を示す。例示的な実施形態によれば、施設は、3つのゾーン121、123、125を有する。各ゾーンは、ジュークボックス133に動作上、接続された独自のスピーカセット127、129、131を備える。異なる音楽が、3つすべてのゾーン121、123、125において同時に再生されることが可能であり、すべての音楽は、単一のジュークボックス133から再生されることが可能である。ジュークボックス133は、必要に応じて、この実施形態を可能にする、さらなるハードウェアを備えることが可能である。

【0051】

代替として、ユーザは、ゾーン121、123、125の複数において同時に、またはそれらのゾーンの複数において異なる時点で、曲を再生させるように選択してもよい。ユーザは、これらのフィーチャのいずれかを実施するのに追加のクレジットを支払わなければならない可能性がある。マルチゾーンシステムの好ましい実施形態は、1998年9月28日に出願した米国特許出願09/161,584号の一部継続出願である「Wireless Digital Transmission System for Loudspeakers」という名称の、2004年12月29日に出願した米国特許出願第11/023,390号において説明されるシステムを使用して、異なるゾーンにおいて高い品質で音楽を再生することができる。両方の出願の内容全体が、参照により本明細書に組み込まれており、このシステムを使用して、例えば、ジュークボックスは、オーディオデータを圧縮して、AC電力線を介して、アドレス指定可能なロケーションに伝送することができ、そのロケーションにおいて、データは、受信され、伸張され、変換されて、再生されることが可能である。実際、本明細書のジュークボックス構成要素のいずれも、動作のための通信ネットワークとしてAC電力線を使用する仕方を実施されることが可能である。

【0052】

2つ以上のデバイス間でデータが送信される必要がある他の実施形態において他の目的で、無線デジタル伝送システムが、使用されることが可能であることを理解されたい。例えば、このシステムを使用して、ダミー端末装置が構成されることが可能である。そのような実施形態では、無線デジタル伝送システムは、例えば、変形させるかどうか、ジュークボックスのある特定の变形を所与として、どのような曲が適切であるか、選択された音楽が再生されるべきゾーン、最大音量レベルなどの情報を、スピーカシステムに音楽を送信することに加えて、送信するのに使用されることが可能である。また、運用者は、ゾーンにおける活動のタイプ、時間帯、または他の任意の適切な選択基準に基づき、所与のゾーンにおいてどのような種類の音楽が利用可能かを制限することもできる。例えば、図9において、ゾーン3 125は、レストランである。レストラン利用客は、図9におけるバーであるゾーン1 121、または玉突き場であるゾーン2 123にいる誰かと同一のタイプの音楽を聴くことを所望しない可能性がある。運用者は、このことを認識して、ゾーン3 125において再生されることが可能な音楽のタイプを制限することができる。代替として、運用者は、任意の所与のゾーンにおける音楽の音量を制限することができる。例えば、玉突き場123またはレストランセクション125の利用客は、バー121において所望されるほど大音量の音楽を望まない可能性がある。また、多分レストランセクション121は、玉突き場123よりも静かに保たれるべきである。オーナーは、任意の適切な基準に基づき、すべての適切な設定を調整し、制御して、ゾーンのそれぞれにおいて最も応用がきき、利用客に優しい環境を提供することができる。

【0053】

図10は、各ゾーンにおける選択端末装置を有するマルチゾーンジュークボックスシステムに関する例示的な施設レイアウトを上から見た図を示す。例示的な実施形態によれば、バーは、3つのゾーン121、123、125を有する。各ゾーンは、ジュークボックス133に動作上、接続された独自のスピーカセット127、129、131を備える。3つすべてのゾーンにおいて、異なる音楽が、同時に再生されることが可能であり、すべての音楽は、単一のジュー

クボックス133から再生されることが可能である。ジュークボックス133は、この実施形態を可能にする、さらなるハードウェアを備えることが可能である。

【0054】

図10の実施形態では、施設全体にわたって配置された1つまたは複数の「ダミー」端末装置137、139も存在する。例示的なダミー端末装置は、Xサーバ技術を使用することができる。独立型のデバイスであっても、デジタルディスプレイを有するゲームマシン上、または他の適切なデバイス上のインターフェースの一部として提供されてもよい、これらの端末装置137、139は、それらの端末装置137、139が配置されているゾーン(または、場合により、他のゾーン)のためのジュークボックス133からの曲の選択を可能にする。これらの端末装置137、139は、メインジュークボックスインターフェースに課されたゾーン制限、および選択基準をコピーする。端末装置137、139は、それぞれの端末装置が配置されているゾーンにおいて再生するための音楽の選択だけしか許さないように制限されても、異なる1つまたは複数のゾーンにおける再生のための選択を許してもよい。

【0055】

さらに、端末装置137、139のグラフィックインターフェースは、利用可能な選択、バーのテーマ、各端末装置が配置されている部屋のテーマ、または他の任意の適切な基準に従って変更されてもよい。

【0056】

図11は、マルチゾーンジュークボックスシステムに関するゾーン選択プロセスの例示的な実施を示す流れ図である。例示的な実施形態によれば、ジュークボックスはまず、ユーザとのトランザクションを開始する、141。ユーザは、曲を選択するように指示され、143、曲が再生されるべき1つまたは複数のゾーンを選択する、145。次に、ジュークボックスは、選択されたゾーンの数に基づいて価格を決定する、147。ジュークボックスは、ユーザから支払いを受け入れ、149、選択されたゾーン、または選択された複数のゾーンにおける再生のために曲をキューに入れる、151。次に、ジュークボックスは、ユーザが、別の曲を選択することを要望するかどうかを調べる、153。ユーザが、別の曲を要望する場合、プロセスは、曲選択ステップ143に戻り、そこから繰り返される。ユーザが、選択を行うことを終えた場合、プロセスは、終了する、155。

【0057】

図12は、マルチゾーンジュークボックスシステムに関するゾーン選択プロセスによる優先再生の例示的な実施を示す流れ図である。例示的な実施形態によれば、いくつかのジュークボックスシステムが、1つまたは複数のゾーンに対応する1つまたは複数の優先キューを備えることが可能である。優先再生が、あるゾーン、または複数のゾーンに関して提供される場合、ジュークボックスはまず、ユーザが、選択された曲に関して優先再生を選択するかどうかを調べる、161。優先再生が、選択された場合、ジュークボックスは次に、優先再生が行われるべきゾーンまたは複数のゾーンをユーザが選択するオプションを提供する、163。優先再生のために選択されたゾーンの数に基づき、ジュークボックスは、価格を決定し、ユーザからその価格の支払いを受け入れる、167。次に、ジュークボックスは、選択された各ゾーンに関する優先再生キューの中に曲を入れる、169。

【0058】

図13は、各ゾーンに関するキューのサブセットを伴う、優先キューおよび非優先キューのマルチゾーンセットの例示的な実施を示す。例示的な実施形態によれば、N個のゾーン171のそれぞれが、優先キュー175と、非優先キュー173とを含む独自のキューセットを備えることが可能である。再生のために選択された曲のリストが、各キュー173、175内に保持される。各キューの中の各曲は、識別子177、179を備えることが可能であり、この識別子177、179は、その曲、および/またはキューの中の、その曲の位置、および/または他の任意の適切なファクタを識別する。

【0059】

いくつかの例示的な実施形態では、キューの中の曲は、他の理由で並べ替えられることも可能である。例えば、曲は、曲の間の移行を滑らかにして、曲の、より良好な(例えば

、聴覚的に、より心地よい、それほど唐突でない、より滑らかなといったような)終わり-先頭間プレイバックを提供するように並べ替えられることが可能である。このため、いくつかの例示的な実施形態では、プレイバック曲の順序を変更するための方法が、提供される。そのようなプロセスにおいて、曲の「色」が、再生リストの中の2つ以上の曲に関して決定されることが可能である。曲の色は、曲の、例えば、音量、リズム、ビート、テンポなどの、曲の特性と関係する。色決定の後、色は、比較されることが可能であり、曲は、類似した(または嫌悪)曲をグループ化するように並べ替えられることが可能である。オブションとして、曲の色が、決定されることが可能であり、照明、ディスプレイなどを制御して、より対話的な体験をもたらすように使用されることも可能である。

【0060】

図14は、変形能力を有するジュークボックスに関する例示的な配信-初期化スキームを示す流れ図である。例示的な実施形態によれば、出荷時ドライブの内容が、製造地点で定義される、181。これと同一のドライブ(または拡張されたメディアストレージ)が、すべてのジュークボックスと一緒に出荷されることが可能であり、183。例えば、中央サーバ上で利用可能な総数の曲のサブセットだけしか含まないことが可能である。このフィーチャは、同一のドライブが、すべてのジュークボックスと一緒に出荷されることを可能にすることにより、製造工程を単純化する。このドライブを含むジュークボックスが、目的地に到着すると、運用者は、ドライブ上の曲のサブセットを、基本再生可能リストとして選択することができる、185。この選択は、施設のタイプ、施設利用客が、通常、選好する音楽のタイプ、または他の任意の適切な基準に基づいて行われることが可能である。また、運用者は、中央サーバが、基本再生可能リストを推奨することを許すこともできる。ドライブは、基本リスト上にないが、出荷時ドライブ上に存在する曲の選択を、追加料金で許すことも可能である、187。しかし、「代替の」曲の、このリストは、運用者が、施設のテーマに合わない曲へのアクセスを制限することを所望する可能性があるので、すべての曲は含まない可能性がある。例えば、カントリーバーオーナーは、ジュークボックス上でラップまたはヒップホップの曲が選択されるのを許すことを決して望まない可能性がある。いくつかの例示的な実施形態では、ジュークボックスは、例えば、ハードドライブ、フラッシュメディアなどの非常に大容量の記憶媒体上に多数の曲(例えば、25,000の曲)を有して出荷される。この実施形態によるジュークボックスは、運用者および/またはオーナーが、利用客に提供する、すべての曲のサブセットを選択するまで、「個性」を有さない。このように、運用者および/またはオーナーは、基本再生可能リストの定義を介して、運用者および/またはオーナーのジュークボックスの「個性」を定義することができる。

【0061】

ドライブ上の曲が、適切に分類されると(すなわち、基本再生可能リストが、定義されると)、ジュークボックスが、動作を開始する、189。新たな基本再生可能リストが、所望される191のでない限り、ジュークボックスは、現在の選択された基本再生可能リストで動作することを経る、189。基本再生可能リストが所望される場合、191、ジュークボックスは、「新たな」ジュークボックスに変形して、193、基本選択に関して異なる再生可能な曲サブセットを選択し、185、好ましくは、その変形によって規定されるとおり、ジュークボックスのさらなる特性(例えば、グラフィック、広告、ルックアンドフィールなど)を変化させる。このため、ジュークボックスを変形させる能力は、オーナーおよび/または運用者が、自動的に、事前定義されたイベント時/時点で、手動でといったように、切り替えられることが可能な、複数のトリガ可能な個性を有する単一のジュークボックスを保持することを可能にする。

【0062】

図15は、トリガイイベントに基づく自動的ジュークボックス変形開始プロセスの例示的な実施を示す流れ図である。例示的な実施形態によれば、ユーザは、あるイベント、例えば、あるテーマの夜、またはある時間帯を、ジュークボックス変形をトリガするトリガイイベントとして定義することができる、201。次に、ジュークボックスが、通常どおり動作し

、203、トリガイイベントが生じたかどうかを定期的に調べる、205。トリガイイベントが生じていない場合、ジュークボックスは、単に、動作すること続けるが、203、トリガイイベントが生じた場合、ジュークボックスは、「新たな」ジュークボックスに変形される。トリガイイベントは、1回限りのイベントであることも、毎週、毎日、毎月、生じるようにスケジュールされること、または他の任意の適切な基準に基づいてスケジュールされることも可能である。マルチゾーン構成において、異なるゾーンは、他のゾーンが変更されない一方で、変形されることが可能であることに留意されたい。例示的な実施形態の、このフィーチャは、例えば、他のゾーンが、時間帯、音楽を変更したいというオーナーの所望、またはユーザの要求などの、様々な要因に基づいて変更されることが可能である一方で、所与のゾーン、または所与の複数のゾーンが、ある種類の音楽に専用にされることを可能にする。

【0063】

図16は、ジュークボックス変形プロセスの例示的な実施を示す流れ図である。例示的な実施形態によれば、ジュークボックスは、変形を始めると、211、新たな曲サブセットを、基本再生可能リストになるように選択する、213。次に、ジュークボックスは、ジュークボックス上の残りの曲の一部、またはすべてが、より高い料金で選択されることを可能にする、215。残りの曲のいくつかは、何が変形をトリガしたかに基づいて制限されることが可能である。また、ジュークボックスの他の特性も、変化することが可能であり、217、例えば、ユーザインターフェースが、変更されることが可能であり、ジュークボックスの変形が合わせられている群衆の予測される嗜好と合致する様々な広告が、表示されることが可能である。また、他の適切な変更も行われることが可能である。好ましい実施形態の1つの例において、クラブオーナーが、水曜日に、午後9:00に始まり、午前4:00に終わるヒップホップの夜を催す。水曜日の午後9:00に、ジュークボックスは、適切な音楽の基本的な選択を伴って、ヒップホップジュークボックスに変形される。変形によれば、ジュークボックスは、カントリーミュージック、クラシックロック、ジャズ、ブルース、および懐メロなどの音楽のジャンルへのすべてのアクセスをブロックし、ジュークボックスは、ハードロック追加曲の利用可能な選択を、「ヒップホップ風の」ハードロック曲に限定する。ジュークボックス上のグラフィックは、先端的な、都会的グラフィックに変換され、広告が、それに応じて変化し、ヒップホップの群衆にアピールするはずの衣類、飲料、および商品などの製品を表示する。午前4:00に、ジュークボックスは、そのクラブに関する「標準の」ジュークボックスに戻るように、または他の任意の適切なジュークボックスに変形される。代替として、ジュークボックスは、次のトリガイイベントが生じるまで、ヒップホップモードに設定されたままであってもよい。この場合も、マルチゾーン構成において、他のゾーンは変化しない一方で、異なるゾーンが変形されることが可能であることに留意されたい。前述の例示的な非限定的な実施形態において、そのクラブに関する「標準の」音楽が、別の地域において再生されたままでありながら、システムは、その夜に関して、1つのゾーンにおいてヒップホップに変形されることが可能である。

【0064】

図16Aは、1つの好ましい実施形態において、ジュークボックスの変形が、どのように開始されることが可能であることを示す例示的なスクリーンショットである。地域160は、選択がまったく行われていないことを示すように空白である。ユーザは、この例示的な実施形態において、例えば、TouchTunes Masterボタン162、Dynamic Masterボタン164、またはClone a Locationボタン166を選択することにより、変形オプションを選択する。他の実施形態が、前述したボタンの代わりに、または前述したボタンに加えて、さらなる変形オプションを提供することが可能であることを理解されたい。この実施形態において、TouchTunes Masterボタン162を押すことは、TouchTunes音楽部門によって事前選択されたマスタ選択のリストで地域160をポピュレートする。これらのマスタ選択には、例えば、特定の曲のコレクション(例えば、英国のバンドによるすべての曲、ある特定のアーティストによるすべての曲)、特定のテーマまたはジャンルの比率(例えば、ある特定のダンスクラブの雰囲気のための75パーセントのヒップホップ、10パーセントのポップ、10パーセント

のロック、5パーセントのテクノの混合)などが含まれることが可能である。ユーザは、スライダバー170を使用して、このリストを詳細に調べることができる。ユーザが満足すると、Morph Itボタン168を押すことにより、ジュークボックスを変形させる。Morph Itボタン168を押すと、即時の変形をトリガすることができ、あるいは、変形を保存し、代わりに、その変形を事前定義されたトリガ時点で開始することができることを理解されたい。また、変形データが、選択が行われたジュークボックス、または任意の数の接続されたジュークボックス端末装置を変形させるために格納されることが可能であることも理解されたい。

【0065】

Dynamic Masterボタン164を押すことは、変形に対するより大きいユーザ支配を提供する1つの例示的な、限定的でない仕方である。図16Bは、1つの好ましい実施形態において、ジュークボックスの変形前に、ユーザが、どのように様々な基準を指定することができるかを示す別の例示的なスクリーンショットである。図16Aと図16Bにおいて同一の符号を有するボタンの説明は、省かれる。例えば、Rapラベル172などの、音楽のジャンルおよび/またはテーマの名称が、提供される。再生されるべき音楽の所与のパーセンテージを指定するのに、ユーザは、増加ボタン174または減少ボタン176を押すことができる。その特定のジャンルまたはテーマに関する、選択された音楽の量が、テーマ音楽量領域178内に現れるのに対して、選択された音楽の合計量についての詳細が、合計音楽量領域180内に現れる。好ましい実施形態において、ユーザは、再生すべきジャンルおよび/またはテーマのパーセンテージを指定することができる。例えば、ユーザは、カントリーの夜に関して、100パーセントのカントリーミュージックに限って選択することができる。代替として、別の非限定的な例として、ユーザは、国際的な雰囲気のためにワールド音楽とラテン音楽の混合を選択することができる。好ましい実施形態は、パーセンテージを使用するが、他の実施形態は、いずれのタイプの音楽のいくつが、再生されるべきかを指定するために、例えば、所与のカテゴリの曲の数、所与のテーマおよび/またはジャンルの再生時間などの、異なる目安を使用してもよいことを理解されたい。また、他の実施形態では、ユーザは、それらのボタンを使用して、音楽の量を増加または減少させなくてもよい、つまり、他の実施形態では、ユーザは、入力機構を介して音楽の量を直接に入力することができることも理解されたい。このため、変形させることは、ジュークボックスに動作上、接続された遠隔デバイスによって定義され、かつ/またはトリガされることが可能であることを理解されたい。

【0066】

ユーザが、Clone a Locationボタン(例えば、図16Aまたは図16Bからの)を押すと、ジュークボックスは、好ましくは、各ジュークボックスデバイスに関する一意識別子を含むジュークボックスデバイスのリストを表示する。しかし、他の実施形態において、リストは、例えば、現在のロケーションと同一のオーナーによって所有される他のロケーションを含むことが可能である。ジュークボックス端末装置が選択された後、現在のジュークボックスは、そのジュークボックスに変形されることが可能である。代替として、選択されたジュークボックスが、現在のジュークボックスに変形されて、例えば、遠隔変形を提供することも可能である。さらに代替として、リストは、例示的な実施形態によって選好されるデジタルジュークボックスに加えて存在する非デジタルジュークボックスについての情報を含むことも可能である。この変形フィーチャは、新たなジュークボックスが、従来のレコード、CD、またはデジタルジュークボックスなどの、置き換えられている、より古いジュークボックスに変形されることを可能にすることにより、ユーザが、既存のジュークボックスを迅速に、容易に置き換えることを可能にすることができる。さらに、複数のジュークボックスを保持する運用者は、ルートベースのクロージングおよび/または変形のプロセスを適用して、例えば、カスタムの、つまり、特殊化された曲セットを提供することができる。

【0067】

さらに、ユーザは、例えば、所与の日付に特有のロケーションのデータ、または知られ

ている最新のデータに基づき、ロケーションを変形させることができる。このことは、ユーザが、例えば、所与の変形の人気度に基づき、ジュークボックスを変形させることを可能にし、例えば、ユーザは、成功した変形に戻って、その夜のエンターテイメントを再現しようと試みることを選択することができる。また、このフィーチャは、ユーザが、ハードドライブクラッシュ、および/または他のジュークボックス障害から回復することも可能にする。例えば、ハードドライブが、古いハードドライブがクラッシュした後に置き換えられた場合、ユーザは、クラッシュより前の所与の日付および/または時刻における既存のロケーションを変形させて(またはクローニングして)、クラッシュより前に利用可能であった曲を復元することができる。というのは、好ましくは、サーバは、この機能のために十分な情報を保持するからである。このため、変形機能を使用することは、バックアップフィーチャおよび復元フィーチャも提供しながら、ユーザに、より大きい支配を与える。

【0068】

図17は、拡張されたメディアストレージを有するジュークボックスと、中央サーバとの間の関係を示す。例示的な実施形態によれば、中央サーバ221は、曲のマスタライブラリを含み、そのようなライブラリは、現在、ダウンロードされるように利用可能なすべての曲と、ジュークボックスハードドライブ上に現在、インストールされているすべての曲とを含む。中央サーバは、ローカルハードドライブ223を含む遠隔ジュークボックス225と通信することができる。222。ジュークボックス上のハードドライブ223は、ダウンロードのために利用可能なスペース227、事前ロードされた曲によって占められるスペース228、ならびにソフトウェアおよびオペレーティングシステム229を含み、いくつかのセクションを有することが可能である。さらなる適切なセクション、例えば、GUIを変更するための様々なピクチャを含むセクションが、追加されることが可能である。ジュークボックス225は、中央サーバ221と通信して、曲をダウンロードし、使用情報をアップロードし、ソフトウェアを更新し、他の任意の適切な機能を実行することができる。

【0069】

図18は、ある曲が、「標準の」利用可能な再生可能な曲リストの中に入っていない場合の曲選択プロセスに関する例示的なプロセスを示す流れ図である。例示的な実施形態によれば、ユーザがまず、曲を選択する、231。ジュークボックスは、その曲が、「非標準の」選択としてローカルハードドライブ上で利用可能であるかどうかを調べる、233。その曲が、ローカルハードドライブ上で利用可能である場合、ジュークボックスは、非標準の曲を獲得して、再生することの代金として設定された価格を顧客に請求し、235、その曲を再生する、237(または、適宜、その曲を再生リストに追加する)。

【0070】

その曲が、ローカルハードドライブ上で利用可能でない場合、ジュークボックスは、中央サーバへの高速の接続が、利用可能であるかどうかを調べる、239。高速の接続が存在しない場合、ジュークボックスは、その曲が、一時的に利用できないことをユーザに知らせ、241、ダウンロードのために、その曲を注文する、243。ジュークボックスは、その曲を注文することの代金として、追加の金額を請求しても、請求しなくてもよい。しかし、中央サーバへの利用可能な高速の接続が存在する場合、ジュークボックスは、その曲を即時に注文し、その高速の接続を使用して、直ちに、その曲をダウンロードし、再生のために、その曲をキューに入れる、245。次に、ジュークボックスは、非標準の選択の価格を顧客に請求する、247。いくつかの例示的な実施形態において、ジュークボックスは、ロケーションが閉店した後、またはロケーションが開店する前に、曲をオフラインで取り出すことができる。いくつかの例示的な実施形態では、ジュークボックスは、ダイヤルアップ接続を使用する専用ラインを介して、曲を即時にダウンロードすることができる。他のいくつかの例示的な実施形態では、曲は、中央の、つまり、限られたデータベースからではなく、別のジュークボックス(または他のジュークボックス)からダウンロードされて、ネットワークの負担が減らされることが可能である。いくつかの例示的な実施形態では、ジュークボックスは、所与の曲の諸部分が、複数の源からダウンロードされて、ターゲッ

トミュージックボックスのために再組み立てされることが可能な分散メディアサービスを介して、曲をダウンロードすることができる。そのようなピアツーピア(またはミュージックボックス-ミュージックボックス)またはマルチピア(いくつかのミュージックボックス-ミュージックボックス)のデジタルダウンロードネットワークは、曲ライセンス情報を追跡する必要があることを理解されたい。いくつかの例示的な実施形態では、ある曲が、あるミュージックボックス上で利用可能ではないが、他のバージョン、または他のカバーが、利用可能である場合、そのミュージックボックスは、これらの他の曲を、検索するユーザに推奨することができる。例えば、「How Do I Live」の、利用できないトリシャ・ヤウードバージョンを検索しているユーザに、ドリー・パートンおよび/またはリアン・ライムスによる利用可能なバージョンが推奨されることが可能である。好ましくは、推奨は、例えば、エヴァリー・ブラザーズの「Oh, Pretty Woman」やモトリークの「Pretty Woman」などの、完全に異なる、類似した名前の付けられた曲を無視するのに十分なだけスマートである。

【0071】

前述した出荷時ドライブが、変形能力と相俟って、前述したローカルサーバの必要性をなくす。というのは、出荷時ドライブは、別個のハードウェアデバイスの必要なしに、ローカルサーバと同一のサービスを証明するからである。つまり、本明細書の出荷時ドライブは、ミュージックボックスが、任意の技術の単一の大容量記憶デバイス(または単一の記憶デバイスとして作用する複数の技術および/または複数のデバイス)を有して出荷されながら、それでも、基本的な再生可能リストが定義されること、拡張された再生可能リストが定義されること、変形能力、ローカルサーバサービスが提供されること、ならびに本明細書で説明される他のすべてのフィーチャを可能にする。出荷される、出荷時ドライブのコンテンツは、好ましくは、顧客選好に関する履歴上の統計情報を使用して定義される。

【0072】

図19は、入札に基づく優先順位付け能力を有する優先再生キューに関する例示的なプロセスを示す流れ図である。例示的な実施形態によれば、ユーザはまず、ユーザが、優先再生を所望することを示す、251。すると、ミュージックボックスは、優先再生キューの現在のステータスを表示する、253。この表示は、どれだけの数の曲が、キューに入っているか、最高の入札はどれであるか、各曲に対して、どれだけの額が入札されているか、いずれの曲が「ロックイン」されているか、ならびに優先キューについての他の任意の適切な情報などの情報を含むことが可能である。次に、ミュージックボックスは、優先順位リスト上のある特定のスポットにユーザに曲を入れるのに、ユーザが、どれだけの追加の金額を支払うことを所望するかを、ユーザが選択することを許し、選択された金額の支払いを受け入れる、255。支払いを受け入れた後、255、ミュージックボックスは、ユーザから受け取られた追加の金額に対応する優先順位リスト上の位置に、その曲を入れる、257。

【0073】

代替として、例示的な実施形態の別の態様では、ユーザは、優先再生のために前に選択された他の曲が再生されるよりも前に、ある曲を再生させる権利について入札することができる。好ましい実施形態では、ユーザには、優先再生のために支払われる最高価格が示され、ユーザは、その価格よりも多くを支払って、入手可能な最高の優先順位を得ることができる。

【0074】

例示的な実施形態の別の態様は、他の人がどれだけの額を優先順位のために支払ったかが、ユーザに示されることを許さない。ユーザは、優先格付けを得るのにユーザが費やすことを所望するだけ、いくらでも支払うことができ、次いで、支払われた金額に基づく優先順位の格付けを受け取ることができる。

【0075】

例示的な実施形態のさらなる例示的な態様によれば、ユーザは、前述の例示的な態様に従って、優先格付けを得るのに、ユーザが費やすことを所望するだけ、いくらでも支払うことができ、次いで、ユーザに、その支払われた金額に基づいて得られた優先スポットが示されることが可能である。このスポットにユーザが満足ではない場合、ユーザは、さら

なる金額を支払って、優先格付けにおいて曲を上昇させることができ、ユーザには、支払われたさらなる金額に基づいて得られた、新たな優先格付けが示されることが可能である。ユーザは、所望される優先格付けが得られるまで、このプロセスを繰り返すことができる。また、ユーザは、さらなる金額を支払って、他のユーザが、入札タイプの状況においてリスト上の選択された優先スポットを先取りするのを、より困難にすることもできる。また、より高い優先順位に対する、より高い支払い額の他の任意の方法が実施されることも可能である。

【0076】

「ロックイン」フィーチャを提供することができる例示的な実施形態の例示的な態様によれば、ユーザは、事前選択された金額の支払いで優先格付けを「ロックイン」することができる。例えば、ユーザが、15クレジットを支払って、第3番の優先順位の格付けを獲得し、その第3番の格付けを保証することを所望する場合、ユーザは、例えば、さらに4クレジットを支払って、その格付けを「ロックイン」することができる。格付けをロックインすることは、ユーザより上のすべての格付けの「ロックイン」も要求するので、ユーザは、ある金額を支払って、ユーザの選択より上のすべての曲を「ロックイン」することを要求される可能性がある。1つのそのような状況において、ユーザは、「ロックイン」の代金として見積もられた価格を支払う、または、さらなる競り上げを防ごうとして、またはユーザの曲を優先リストの中でさらに上昇させようとして、同一の量の、もしくは異なる量のクレジットを支払うことを選択することができる。

【0077】

例示的な実施形態の別の例示的な態様によれば、前述した入札戦略のいずれかが、実施されることが可能であり、ユーザには、皆が、特定の格付けのためにどれだけの額を支払ったかが示されることが可能である。このことは、ユーザが、ある優先位置を得るのに、厳密にどれだけの額を支払わなければならないかを知ることを可能にする。「ロックイン」フィーチャが実施される場合、このことは、その曲を「ロックイン」する価格を支払うほうが安いのか、または優先リストで上昇するように支払うほうが安いかをユーザに知らせる。これらのオプションのすべては、運用者の、より高い収益をもたらす。

【0078】

前述の実施形態は、移動可能でないジュークボックスにメディアを配信するためのシステムを説明するが、類似したシステムを使用する代替の実施形態は、ポータブルジュークボックスデバイスにメディアを配信することができ、本発明によって企図され、本発明の範囲および趣旨に含まれることに留意されたい。ポータブルジュークボックスは、例えば、音楽を受信して、再生することができるPDA、セル電話機、または他の任意の移動可能なデバイスであることが可能である。さらに、メディアは、前述した方法(例えば、ブロードバンド接続、無線接続などを介する)、または例えば、Bluetooth技術を使用するなどの、特定のポータブルデバイスにより適した、他の任意の適切な方法を使用して、ポータブルジュークボックスに配信されることが可能である。さらに、前述したジュークボックスは、商業目的である。しかし、例えば、住宅向けメディアを再生するなどの、他の目的のジュークボックスも、本発明によって企図され、本発明の範囲および趣旨に含まれる。

【0079】

他の例示的な実施形態において、ジュークボックスを有するロケーションの従業員に、ジュークボックス上で顧客が支払う再生の回数に基づいて報酬が与えられることが可能である。1つのそのような例示的な実施形態では、システムは、顧客によって何回の再生の代金が支払われたかという回数を記録する。ある回数の再生の代金の支払いが行われるたびに、バー従業員は、1回の無料の再生を受け取る。一部の実施形態では、別個のカウンタが保持されるのに対して、他の実施形態では、利用可能な無料の再生の回数を計算するのに、1つだけのカウンタしか必要でない可能性がある。このため、いくつかの例示的な実施形態は、無料の再生で従業員に補償を行うことのより、ジュークボックスを奨励するよう従業員を促す。また、このことは、一日中、曲選択が類似していることを減らすこと、異なる交替勤務時間の従業員のために無料の再生を取っておくことなどにも役立つ。

【0080】

同様に、ジュークボックスユーザは、メディアの特定のインスタンスに投票して、再生リストにおけるユーザの優先順位を変更することができる。このようにして、ジュークボックスユーザは、例えば、ロケーション内、またはロケーション間で、ある特定のゾーン、または特定の複数のゾーンにおいて再生されるべき音楽に対する支配を「争う」ことができる。この投票/争いモードは、例えば、動的キュー、優先キュー、複数のキューなどを使用して実施されることが可能であることが理解されよう。また、ジュークボックスは、投票モードに自動的に入れられる(例えば、特定の時間帯に、かつ/または特定の曜日に)ことも、手動でトリガされることも可能であることが理解されよう。図19Aは、ジュークボックス投票プロセスの例示的な実施を示す流れ図である。従来のジュークボックス動作、またはジュークボックス入札モードとさえ異なり、ジュークボックスキューの中の曲のリストが、ステップS1902でユーザに表示される。このリストは、運用者、バーマネージャ、バーにおける利用客などによって自動的に生成されることが可能である。このリストは、例えば、ジュークボックス上で表示されることが可能であり、あるいは、より好ましくは、1つまたは複数のスタンドアロンのモニタ上で表示されることが可能である。さらに、いくつかの例示的な実施形態では、このリストは、複数の移動デバイスおよび/または移動端末装置によって閲覧されることも可能である。好ましくは、表示される情報は、来るべき曲の少なくともアーティストおよび題名、ならびに、それらの曲の優先順位に関連する番号を含む。このリストは、最も多くの「投票」を有する曲が、再生されるべき「次の」曲として表示され、その後、次に高位の曲が表示されるように、この番号によって並べ替えられなければならない。ステップS1904は、現在、再生されている曲が終わったかどうかを判定する。曲が、終わっている場合、ステップS1906が、その曲をキューから削除し、次の曲が、ステップS1908で再生される。別の実施形態では、ユーザは、現在、再生されている曲(またはメディアのインスタンス)を、例えば、ジュークボックスが再生することを始める前に、その曲が有していた投票の数を超えることにより、停止する/飛ばすことができる。

【0081】

ユーザは、来るべき曲を見るにつれ、ユーザが好む曲を押し上げて、それらの曲が、かつ/またはユーザが好まない曲を押し下げようように誘惑される。一般に、投票するユーザが多いほど、良好な曲の雰囲気が高まる。このため、ステップS1908の後、または曲が終わっていない場合、ジュークボックスは、ステップS1910で、特定の曲に対するユーザの投票を受け取る。投票は、クレジットに基づくこと(ユーザは、クレジットを購入するので)、またはユーザアカウントに結び付けられることが可能である。いくつかの例示的な実施形態では、ユーザは、投票するのにログインしなければならない。いくつかの例示的な実施形態では、ユーザは、限られた回数、投票することができる。好ましくは、ユーザは、ロケーション内の少なくとも2つの異なる場所から投票することができる。キューは、ステップS1912で、この投票に基づいて更新され、プロセスは、ステップS1902に戻り、表示されるリストが、リフレッシュされる。

【0082】

いくつかの例示的な実施形態では、キューは、特定の曲に対する投票の合計数に基づくことが可能である。他のいくつかの例示的な実施形態では、ユーザは、いくつかの曲に賛成する投票、および/または反対する投票を行うことができ、賛成の数と反対の数、または正味の結果だけを示す「正味の」情報が表示されることが可能である。ある曲に、賛成する投票よりも多くの、反対する投票が存在する場合、システムは、以下の機能の1つまたは複数を実行することができる。例えば、ジュークボックスは、負の数の投票を有する曲を、単にキューの中に保つことができる。代替として、ジュークボックスは、0の投票、または負の数の投票を有する曲をキューの中に保つが、例えば、常に、正味の投票が、少なくとも1に達するまで待つてから、その曲を再生してもよい。さらに代替として、ジュークボックスは、0の投票、または負の数の投票に達した曲をドロップしてもよい。

【0083】

図19B(1)は、投票キュー表示を示す例示的なスクリーンショットである。表示1900aは、現在、再生している曲1902、およびその曲の進行1904を示す。現在、再生されている曲1902の進行1904は、例えば、残り時間を示す数値、スライドバーなどの、いくつもの仕方に表示されることが可能であることが理解されよう。アーティストと曲のリスト1906が、好ましくは、投票領域1908aにおける投票の対応する数に基づいて表示される。オプションとして、ユーザ領域1910が、例えば、その曲をリストにのせたユーザ、その曲に投票した最後のユーザ、その曲を再生させるように「争っている」グループの名前などの、その曲に投票したユーザについての情報を示すことが可能である。

【0084】

図19B(2)は、投票キュー表示を示す別の例示的なスクリーンショットである。図19B(2)は、図19B(1)と同様であるが、投票領域1908bは、ユーザの「正味の」投票を反映する。つまり、例えば、図19B(1)では、表示は、ある特定の曲に賛成する投票と反対する投票の数、および正味の投票結果を示す。

【0085】

図19Cは、所与のロケーション内の投票機構を示す例示的なブロック図である。表示1900aは、図19B(1)に対応し、ただし、図19B(2)に対応する表示、ならびに同一の、もしくは同様の情報を示す他の任意の表示が、代用されることも可能であることが理解されよう。リストは、好ましくは、ジュークボックス16によって更新される。ユーザは、ジュークボックス16を介して、または、例えば、別のジュークボックス1912a、遠隔端末装置1912b、ラップトップ1912c、PDA1912d、セル電話機1912e、別の移動デバイス1912fなどの、ジュークボックス16にオプションとして接続される任意のデバイスによって、リストを更新するように投票する(例えば、曲に賛成して/反対して投票する)ことができる。

【0086】

図19Dは、所与のロケーション内の投票機構を示す別の例示的なブロック図である。図19Dは、オプションとして接続されたデバイス1912a～fから投票するユーザが、ジュークボックス16とインターフェースをとることなしに、投票リストに直接に影響を与えることができるという点で、図19Cとは異なる。

【0087】

代替のアプリケーションは、複数のロケーション(例えば、バー、レストラン、会場など)が、例えば、インターネットを介して一緒にネットワーク化される、投票機構を含む。これらのリンクされたロケーションは、より多くの利用客が、互いに「争う」ことを可能にする。この実施形態では、ネットワーク化されたジュークボックスは、同一の曲を再生しており、すべてのロケーションからのすべてのユーザが共同で、投票機構などを介して再生順序を決定する。このため、図19Eは、複数のロケーションで使用するための投票機構を示す例示的なブロック図である。第1のジュークボックス16(1)と通信する第1のロケーション内の第1の複数の移動デバイス1912a(1)～f(1)を有する第1のジュークボックス16(1)、ならびに第2のジュークボックス16(2)と通信する第2のロケーション内の第2の複数の移動デバイス1912a(2)～f(2)を有する第2のジュークボックス16(2)が、ジュークボックス16(1)および16(2)によってアクセス可能な再生リストに影響を与えることが可能である。

【0088】

図19Fは、複数のロケーションで使用するための投票機構を示す別の例示的なブロック図である。図19Fは、図19Eと同様であるが、第1の複数の移動デバイス1912a(1)～f(1)、および第2の複数の移動デバイス1912a(2)～f(2)が、投票リストに直接に影響を与えることが可能であることだけが異なる。

【0089】

この投票フィーチャの代替のアプリケーションは、ジュークボックスの「一時停止」モードを強化することに関する。ジュークボックスは、一時停止モードになっている間、通常、ユーザが、ジュークボックスから曲を選択することを防止する。例えば、バーテンダーは、特別イベント(例えば、フットボールの試合)が、そのロケーションで提示される場

合、およびそのロケーションのオーディオシステムが、そのマッチまたはイベントを流している場合、一時停止モードを使用することができる。投票フィーチャは、その特別イベントが終わった後に、いずれの曲が再生されるかを選択するのに使用されることが可能である。試合を観ている利用客が、試合が終わると、しばしばそのロケーションを去ることが広く見られている。再生されるべき曲を選択する、この代替の方法は、あるロケーションに留まり、かつ/またはジュークボックスなどを使用しつづけるよう、利用客に動機を与える可能性が高い。

【0090】

図20は、1つの例示的な実施形態による音楽、通信などを管理するためのユーザに合わせたシステムの利用に関する例示的なプロセスを示す流れ図である。パーソナルミュージックアシスタントは、この例示的な実施形態によるシステムの補完、構成要素などの役割をすることが可能であることを理解されたい。ステップS2000において、ユーザは、この場合、MyTouchTunesと名付けられたユーザに合わせたシステムにアクセスする。ユーザは、ジュークボックス端末装置などにおいて、例えば、インターネットフロントエンドを介するなどして、いくつかのロケーションにおけるシステムにアクセスすることができることを理解されたい。ユーザは、例えば、ユーザ名とパスワードの組合せをインターネットフロントエンドまたはジュークボックス端末装置に入力することによって、ジュークボックス端末装置においてクレジットカードをスロットに通すことによって、サイト特有の「スマートカード」またはフラッシュカードをスキャンすることによってなど、いくつかの仕方でシステムにアクセスすることができることをさらに理解されたい。いくつかの例示的な実施形態では、ユーザは、例えば、指紋、網膜スキャンなどのバイオメトリックデータを介して認識されてもよい。バイオメトリック認証は、独立に働いても、例えば、ユーザ名/パスワードの組合せ、スマートカードなどの、ユーザを識別する、より旧来の形態と一緒に働いてもよいことを理解されたい。システムは、ステップS2002で、システムが、ユーザを認識するかどうかを判定する。ユーザが認識されない場合、ステップS2004におけるとおり、ユーザは、アカウント/プロフィール情報を作成するように促されることが可能である。アカウント/プロフィール情報については、図21Aを参照して、後段でより詳細に説明する。

【0091】

ユーザが、ステップS2002で認識された場合、ステップS2006で、MyTouchTunesメインメニューが、ユーザに表示される。MyTouchTunesメインメニューから、この例示的な実施形態によるシステムを使用するユーザは、例えば、ステップS2010でアカウント/プロフィール情報を編集し、ステップS2020で音楽を管理し、ステップS2030で仲間を管理し、ステップS2040で他の人々と通信し、かつ/またはステップS2050で、システムによって提供されることが可能な様々な付加価値コンテンツにアクセスすることができる。これら、およびその他のフィーチャを、図21Aないし図21Bを参照して、以下により詳細に説明する。

【0092】

図21Aは、1つの例示的な実施形態におけるアカウント/プロフィール情報を編集するのに利用できる例示的なオプションを示す。前述したとおり、図21Aに示される例示的なステップは、初期アカウント/プロフィールセットアップに要求されるステップに対応することが可能であることを理解されたい。しかし、そのような初期セットアップの場合において、例示的なシステムは、メインメニューにユーザがアクセスすることを許すのに先立って、ユーザが、一度に、要求される情報のすべてを完成させることを要求することが可能である(図20のステップS2006)。また、そのようなアカウント/プロフィール情報は、例えば、中央ロケーションまたは中央データベースの中、ある特定の施設のサーバ上、特定のジュークボックス端末装置上、および/またはジュークボックスに挿入された際にユーザ、およびユーザの情報を識別する取り外し可能なカード上などの、所与の実施形態の必要に応じて、様々なロケーションに格納されることが可能であることも理解されたい。

【0093】

ユーザはまず、ステップ2012で個人データを入力する。そのような個人データには、例

えば、ユーザの電子メールアドレス、電話(好ましくは、移動電話)番号、および/または名前が含まれることが可能である。また、システムは、システムが、後の時点で、他のロケーションからユーザを識別することができるように、一意のユーザ名およびパスワードを作成するよう、ユーザに促すことも可能である。代替として、システムは、ユーザの電子メールアドレスをユーザ名として使用し、対応するパスワードだけを要求してもよい。さらに代替として、住宅向けジュークボックスの場合において、例えば、例示的なシステムは、パスワードをまったく要求しないことも可能である。個人データに基づき、例示的なシステムは、ログイン時にユーザを認識するのに十分な情報(例えば、オンラインアクセス、好ましいロケーション、所与の地域内の他のロケーション)を収集することができるとともに、ユーザに電子メールメッセージおよびテキストメッセージを送信することができる。これらの例示的なフィーチャを、以下により詳細に説明する。

【0094】

ステップS2014で、ユーザは、ユーザが、よく訪れる様々なロケーションを指定することができる。このフィーチャは、様々な施設、広告主などによるユーザ追跡を可能にすることができる。施設は、例えば、ある特定の地域内の常連客、またはその他の人々に特別オファーを提供して、来訪するよう、それらの人々を誘うことができる。広告主は、個人情報、ロケーション情報、および再生リストを観察して、ユーザの選好に基づいてアピールするように思われる可能性がある、例えば、コンサートチケット、CDに関する特別オファー、Tシャツ、および/または他の商品を提供することができる。さらに、そのようなアカウント/プロフィール情報が、様々な端末装置上にローカルで格納される場合、システムは、それらの他の端末装置と通信して、ユーザの訪問に備えて再生リストを提供することができる。ステップS2016で、ユーザは、クレジットを追加し、かつ/または支払い情報を指定することができる。ユーザは、例えば、施設に入る前に、または施設にいる間に、ユーザのクレジットカードを使用して、ユーザが有するクレジット数を増やすことができる。代替として、ユーザは、例えば、クレジットカード情報を提供して、ジュークボックスが、購入される将来のクレジットの代金を自動的にユーザのクレジットカードに請求することを許すことなどにより、「認定」されてもよい。ステップS2016は、自身のために、または、例えば、自分の仲間、施設の従業員などの、誰か他の人のためにクレジットを購入するのに使用されることが可能であることを理解されたい。また、ユーザは、例えば、Pay-Palアカウントなどのアカウントからジュークボックス特有のアカウントに金を振り込むことができ、あるいはユーザは、そのようなアカウントを使用して、音楽および/またはサービスを直接に購入することが可能であることも理解されたい。

【0095】

いくつかの例示的な実施形態では、ユーザは、ジュークボックスにおいてアカウントを作成し、例えば、移動電話番号および/または電子メールアドレスなどの検証連絡機構を提供することにより、登録することができる。ユーザが、アカウントを作成し、検証連絡機構を入力した後、メッセージが、そのロケーションに送信されることが可能である。例えば、ユーザは、テキストメッセージを受信することができる。メッセージが受信されると、ユーザは、ジュークボックスにあるコードを入力し、返信の電子メールメッセージまたはテキストメッセージなどを送信して、完全に登録され、かつ/または、例えば、本明細書で開示されるフィーチャの1つまたは複数などの、いくつかの他の、より高度なフィーチャを活用するよう、促されることが可能である。

【0096】

図21Bは、1つの例示的な実施形態における音楽を管理するのに利用できる例示的なオプションを示す。ステップS2022で、ユーザは、再生リストを作成する、または既存の再生リストを編集することができる。ユーザは、好ましくは、前述した方法を使用して曲の中で検索を行い、かつ/またはシステムによって生成された示唆を使用して、ユーザの再生リストを構築することができる。好ましくは、ユーザは、例えば、「Favorite Jams」、「80's Night」、「Smooth Love Songs」などの説明タイトルを、ユーザに再生リストに与えることもできる。ユーザは、ユーザの再生リストを「公開」または「プライベート」

と指定して、他のユーザが、ユーザの再生リストをブラウズするのを許す、または制限することがさらにできる。ユーザは、ステップS2024で、PickListフィーチャを介して、ユーザの曲および/または再生リストを格付けすることができる。また、PickListフィーチャは、所与の時点でユーザが最高の格付けを与えた曲を含むカスタム再生リストを生成することも可能である。代替として、PickListは、前述したHotHitsリストに基づくリストを含むことが可能である。ステップS2026で、ユーザは、例えば、これらの再生リストを使用することを伴って、または伴わずに、前述した方法のいずれかを使用して、ジュークボックス上で音楽を再生することができる。通常、ジュークボックス上で再生するために再生リストを選択することにより、それらの曲は、通常のキューに追加される。しかし、前述したとおり、ユーザは、追加料金を支払って、かつ/または前述した入札機構を使用して、ユーザの曲を即時に再生することができる。さらに、ジュークボックス上で再生するためにある再生リスト(または複数の再生リスト)を選択することにより、通常、その特定の再生リスト(またはそれらの特定の再生リスト)の中のすべての曲が再生される。しかし、ユーザは、いくつかの曲を選択解除して、それらの曲が再生されるのを防ぐこともできる。これにより、ユーザは、ユーザが、ユーザの音楽を再生する際に、再生リストを動的にさらにカスタマイズすることができるようになる。

【0097】

さらに、代替として、さらなる再生リストが、外部源からインポートされることが可能である。例えば、ソフトウェアデバイスおよび/またはハードウェアデバイスは、しばしば、ユーザによって定義された再生リストおよび/またはカスタム生成された(例えば、最も頻繁に再生される、最近、追加されたなどの)再生リストを格納する。例として、限定なしに、再生リストは、iPod、ポータブルMP3プレーヤ、および/または他のハードウェアデバイスから抽出されることが可能である。これは、ジュークボックスに提供される適切なハードウェアコネクタを使用して、ジュークボックスにおいて、かつ/または適切なハードウェアコネクタ(例えば、独自のドッキングポート、USB接続など)が、既に配置されている可能性がより高いジュークボックスから遠隔のデバイス(例えば、ホームコンピュータ上の)において行われることが可能である。いずれかの場合、または両方の場合において、再生リストは、再生リストが、ローカルデバイスにインポートされた後、中央オーディオビジュアル配信サーバにアップロードされることが可能である。同様に、例として、限定なしに、再生リストは、iTunes、Windows(登録商標) Media Player、および/または他のソフトウェアアプリケーションから抽出されることが可能である。多くの場合において、そのようなデバイスおよび/またはソフトウェアアプリケーションは、再生リストが、フラットテキストファイルと同様の仕方で読み取られることが可能であるように、暗号化されず、かつ/または符号化されない仕方で再生リストを格納する。このように、かつ/または他の仕方で、インポートされた再生リストは、オーディオビジュアル配信ネットワークに接続された1つまたは複数のジュークボックスに提供されることが可能である。もちろん、他のいくつかの場合において、例えば、ジュークボックス上で、またはジュークボックス関連のサービスに関連して使用するのに適した現時点で読み取り可能な再生リストを作成する処理に先立って、他のハードウェアデバイスおよび/またはソフトウェアアプリケーションの再生リストに対して、復号化、復号、および/またはその他の処理を実行することが必要である可能性がある。

【0098】

このフィーチャは、少なくとも、単一の再生リストが、デバイス上で定義されて、その後、後の時点で、同一の情報の相当量の再入力、再選択、および/または再グループ化なしに、ジュークボックス上で使用されることが可能である限りで、再生リストを作成し、既存の再生リストを後に再作成する必要性が小さくされることが可能であるという点で、ユーザに有利である可能性がある。また、そのような技術は、例えば、ユーザが、より快適に感じ、かつ/または、より慣れている、異なるインターフェースを有するなどといったデバイスおよび/またはソフトウェアアプリケーションを介して、再生リストのカスタマイズを可能にすることにより、ジュークボックスと対話し、ジュークボックスをユー

ザの選好にカスタマイズするさらなる手段を提供する。

【0099】

このため、オーディオビジュアル配信ネットワークに接続された1つまたは複数のデジタルジュークボックスに再生リストを配信するシステムおよび/または方法が、いくつかの例示的な実施形態において提供される。周辺デバイス上のコンピュータ可読記憶媒体の中に格納された周辺再生リストを有する周辺デバイスが、コネクタを介してデジタルジュークボックスに接続される。この周辺再生リストは、デジタルジュークボックスのために取り出される。これは、ジュークボックスに提供されるプログラミングされた論理回路(例えば、ハードウェア、ソフトウェア、ファームウェア、および/または類似物の任意の適切な組合せ)によって達せられることが可能である。周辺再生リストは、ジュークボックス再生リストに変換される。ジュークボックス再生リストは、少なくともジュークボックス上で公開される。同様に、コンピュータデバイス(例えば、PC、ラップトップ、PDA、または他の類似したポータブルコンピューティングデバイス)上で実行されるソフトウェアアプリケーションが、コンピュータによって読み取られ、オーディオビジュアル配信ネットワーク上のサーバに公開されることが可能である。周辺デバイスは、例えば、ポータブル音楽プレーヤ、コンピュータ、および/または携帯電話機の少なくとも1つであることが可能である。

【0100】

周辺再生リストは、デジタルジュークボックスのコンピュータ可読記憶媒体に、少なくとも一時的に格納されることが可能である。ジュークボックス再生リストが、公開ジュークボックス再生リストと指定されるべきか、またはプライベートジュークボックス再生リストと指定されるべきかが、判定されることが可能である。この判定に基づき、ジュークボックス再生リストが、一般に利用可能にされる、または再生リストをインポートする特定の登録されたユーザにプライベートに保たれることが可能である。

【0101】

ジュークボックス再生リストは、オーディオビジュアル配信ネットワークのサーバに送信されることが可能であり、ジュークボックス再生リストは、オーディオビジュアル配信ネットワークに接続されたジュークボックス、および/またはオーディオビジュアル配信ネットワークに接続された遠隔デバイスに提供されることが可能である。再生リストをインポートする個人は、再生リストをインポートすることを許されるにはまず、ジュークボックスに、またはジュークボックスへのポータルにログインしなければならない可能性がある。

【0102】

また、同一の機能および/または類似した機能を実現するジュークボックスデバイスが、提供されることも可能であり、その機能の少なくとも一部は、ジュークボックスに提供されるプロセッサの使用を介して可能にされる。

【0103】

いくつかの例示的な実施形態において、ユーザが、ジュークボックスインターフェース自体を使用して、ジュークボックスにおいて直接に再生リストを作成することを防止することが、有利である可能性がある。例えば、これにより、ジュークボックスの周囲の混雑の量、単独のユーザ、または少数のユーザによるジュークボックスの独占などが抑えられることが可能である。さらに、または代替として、ユーザは、ジュークボックスにおいて直接に再生リストを作成する、より小さい能力を有することも可能である。例えば、ユーザは、所定の曲セットから再生リストを作成することができることが可能である。その所定の曲セットは、その特定の夜に、その特定のロケーションにおけるジュークボックスにおいて、オーディオビジュアル配信ネットワークに接続された任意のジュークボックスにおいてといったように、ユーザによって再生される曲だけを含むことが可能である。このようにして、ジュークボックスにおいて直接に、本明細書で説明される再生リスト技術のカスタマイズの恩恵をユーザが受けることができるようにすることが、そのようにするのに要求される時間の短縮も行いながら、可能である。さらに、ユーザは、再生されており

、したがって、再び再生されることが可能である可能性が高い曲が追加された、再生リストを作成するように促されることが可能である。

【0104】

このため、いくつかの例示的な実施形態では、デジタルダウンロードミュージックボックス上でメディアインスタンスを再生するためのシステムおよび/または方法が、提供される。ユーザが、ミュージックボックスにログインするように要求される。複数の選択可能なメディアインスタンスが、ミュージックボックス上で再生されるように提供される。複数の選択可能なメディアインスタンスからのメディアインスタンスのユーザ選択は、ミュージックボックスへのユーザインターフェースを介して可能にされる。各ユーザ選択に関して、そのメディアインスタンスが、ミュージックボックスのコンピュータ可読記憶媒体の中に格納された再生リストに追加される。再生リストの中のメディアインスタンスは、順に再生される。複数の選択可能なメディアインスタンスは、ユーザによるメディアインスタンスの前の再生に応じて、メディアインスタンスのマスタグループから選択される。いくつかの例示的な実施形態において、複数の選択可能なメディアインスタンスは、所定の期間内にユーザによって前に再生されたメディアインスタンスから、および/または所定のロケーションにおいてユーザによって前に再生されたメディアインスタンスから形成される。いくつかの例示的な実施形態では、複数の選択可能なメディアインスタンスは、ユーザに関連付けられた最近、再生されたメディアインスタンスのリストの中に格納されたメディアインスタンスから形成される。

【0105】

曲が、再生リストに追加されると、再生リストは、全体が、または一部が再生されることが可能である。例えば、ユーザが、十分なクレジットを有する場合、再生リスト全体が、再生されることが可能である。しかし、ユーザが、十分なクレジットを有さない場合、ユーザは、さらなる支払いを入れて、再生リスト全体を再生するのに十分な数のクレジットを購入することも、曲の一部だけを再生することもできる。曲の一部だけを再生する場合、ユーザは、再生リストから再生のために曲のサブセットを選択するように促される。代替として、再生リストからの曲は、クレジットが尽きるまで、順に再生されることが可能である。オプションとして、適宜、追加の支払いを入れて再生リスト全体を再生しよう、ユーザに対するプロンプトが、提示されることが可能であり、このプロンプトは、十分な支払いが挿入されると、表示から削除されることが可能である。

【0106】

また、ユーザが、再生の代金として請求されることが可能な金額に関して、いくつかの例示的な実施形態において、登録され、かつ/または検証されたユーザに、再生に関する「価格破壊」が与えられることが可能であることが理解されよう。例えば、ユーザには、通常、ミュージックボックス上で現在、利用できない(例えば、オーディオビジュアル配信ネットワーク、または他の源からの、例えば、ダウンロードを要求する)曲を選択することに関して、割増料金が請求されることが可能である。いくつかの例示的な実施形態において、いくらかの好ましいユーザには、曲の源がいずれであるかにかかわらず、低減された量、または最小量のクレジット(例えば、1クレジットだけ)しか請求されないことが可能である。他のいくつかの例示的な実施形態において、好ましいユーザには、その曲が、その特定のユーザによって最近、再生されている場合、最小量のクレジットだけしか請求されないことが可能である。さらに他の例示的な実施形態では、好ましいユーザには、その曲が、そのユーザに関連付けられた再生リスト(例えば、そのユーザが作成した再生リスト、そのユーザが、全体または一部を再生した、別の人によって作成された再生リストなど)の中に存在する場合、低減された量、または最小量のクレジットだけしか請求されないことが可能である。このため、いくつかの例示的な実施形態において、それらのユーザの再生リストに関連するいくつかのメディアインスタンスに関して請求されるクレジットの数を低減することにより、検証されたユーザに報酬を与えることができることが可能である。

【0107】

このため、デジタルダウンロードジュークボックス上でメディアインスタンスを再生するシステムおよび/または方法が、いくつかの例示的な実施形態において提供される。ジュークボックス上で複数のメディアインスタンスが、提供される。ユーザが、ジュークボックス上で再生するために、ある特定のメディアインスタンスを検索することを可能にするインターフェースが、ジュークボックスのユーザに提供される。メディアインスタンスが、ジュークボックスによるさらなる処理なしに(例えば、再生のために標準であることを超えて、メディアインスタンスのさらなるダウンロードまたは取り出しの必要なしに、非標準の検索の必要なしにといったように)、ジュークボックス上の即時の再生に利用できるかどうか判定される。メディアインスタンスが、即時の再生に利用可能な場合、ユーザには、所定の第1のクレジット数が請求され、メディアインスタンスが、再生のためにキューに入れられる。メディアインスタンスが、即時の再生に利用できない場合、ユーザが、検証されたユーザであるかどうか判定される。ユーザが、検証されたユーザであるとき、メディアインスタンスが、ユーザに関連付けられた再生リストの中に存在する場合、第2の所定のクレジット数が、請求されるが、存在しない場合、第3の所定のクレジット数が、請求される。メディアのインスタンスが、再生のためにキューに入れられる。第2の所定のクレジット数は、第3の所定のクレジット数より少なく、第3の所定のクレジット数は、所定のクレジットの第1の数より多い。

【0108】

料金請求に関して、いくつかの例示的な実施形態では、通常の再生の価格は、ユーザが、登録されたユーザである場合、ダウンロードを要するプレミアム曲の価格と同一であることが可能である。いくつかの例示的な実施形態では、ユーザに関連付けられた再生リストは、そのユーザによって前に作成された再生リスト、そのユーザによって最近、再生された曲を含む再生リスト、および/または別のユーザによって作成されたが、そのユーザによって少なくとも部分的に最近、再生された再生リストである。

【0109】

メディアインスタンスが、ジュークボックスによるさらなる処理なしに、ジュークボックス上で即時の再生に利用できるかどうかの判定は、そのメディアインスタンスが、即時に再生可能なメディアインスタンスを含むジュークボックスの専用ストレージ領域とは別個のストレージ領域上に存在するかどうかを判定することを含むことが可能である。別個のストレージ領域は、遠隔サーバ、ジュークボックスの専用ストレージ領域のあるパーティション、ジュークボックス内の別個のストレージ領域などであることが可能である。メディアインスタンスが、ジュークボックスによるさらなる処理なしに、ジュークボックス上で即時の再生に利用できるかどうかの判定は、さらに、または代替として、インターフェースによって提供されるユーザの検索方法を判定することを含むことが可能である。例えば、タイトル検索またはアーティスト検索と比べて、「スーパー検索」に関して割増料金が、請求されることが可能である。もちろん、これらの例は、例として、限定なしに提供され、本明細書で説明される他の検索方法が、提供されることも可能であり、価格設定は、それらの方法だけに基づいて、かつ/または、それらの方法と、ユーザのタイプ、および/またはユーザに関連付けられた再生リストの中のメディアインスタンスの存在との組合せに基づいて、区別されることが可能である。

【0110】

この場合も、他の場合と同様に、プログラミングされた論理回路(例えば、ハードウェア、ソフトウェア、ファームウェア、および/またはこれらの任意の適切な組合せ)の任意の適切な組合せが、そのような機能を可能にすることが可能である。さらに、これを実行するための命令が、任意の適切なコンピュータ可読記憶媒体上に格納されることが可能である。

【0111】

図21Cは、1つの例示的な実施形態における仲間リストを管理するのに利用できる例示的なオプションを示す。ユーザは、仲間(例えば、友人、バーの常連客など)を常に把握しておくことができる。システムは、例えば、AOL Instant Messenger、MSN Messenger、Yahoo

ol Messenger、ICQ、カスタムのジュークボックス特有のプロトコルなどの任意のインスタントメッセージング標準を使用して、メッセージを送受信することができる。好ましくは、システムは、仲間のステータス(例えば、オンライン、非活性、ビジーなど)をユーザに知らせる。ステップS2032で、ユーザは、仲間リストを作成する、または既存の仲間リストを編集することができる。ユーザは、例えば、仲間の名前、電子メールアドレスなどを入力することにより、仲間を検索することができる。さらに、ユーザは、ステップS2038で、好ましくは、電子メール、または移動電話機へのSMS(Short Message Service)テキストメッセージによって、招待状を発行することにより、参加するよう、友人を招待することができる。

【0112】

また、ユーザは、S2034に示されるとおり、ユーザの仲間の音楽コレクション、再生リスト、PickListなどをブラウズすることもできる。次に、ユーザは、ユーザの友人の間で人気があるとユーザが認識したものに基き、ユーザ独自の再生リストおよび/またはPickListを開発することもできる。また、ユーザは、ステップS2036に示されるとおり、好ましくは、有料で、ユーザの仲間から音楽をコピーする、またはユーザの仲間に音楽を送信することもできる。図示されていないステップで、ユーザは、ユーザの友人が、ジュークボックスにログインした際に再生すべき特定の曲を、やはり、好ましくは、有料で、指定することもできる。仲間の間の通信に係る他のフィーチャを、図21Dを参照して、以下に詳細に説明する。

【0113】

図21Dは、1つの例示的な実施形態における他の人々と通信するのに利用できる非限定的な、例示的なオプションを示す。好ましくは、システムは、電子メールメッセージおよびSMSメッセージを移動電話機に送信することができる。システムは、例えば、MyTouchTunesシステムによってだけアクセス可能な電子メールメッセージおよび/または電子メール様のメッセージを介するなど、任意の様々な通信プロトコルを使用して通信することができる。また、パーマネージャが、メッセージを送信して(例えば、電子メール、SMSなどを介して)イベント、スペシャルなどを他の人々に知らせることもできることを理解されたい。ステップS2042で、ユーザが、ユーザの受信ボックスを確認することができる。ユーザは、例えば、ユーザの友人(または仲間)、所有経営者、および/または広告主などからメッセージを受信することができる。仲間は、例えば、曲、または集まる場所を示唆することができる。一部の例示的な実施形態では、仲間は、集まり、バー内のある特定のユーザの位置などを仲間に知らせるインスタントメッセージを移動電話機上で受信することができる。さらに、ユーザは、誰か他の人のログイン時に現れるように、他の人にメッセージを残すこともできる。所有経営者は、所有経営者がターゲットとすることを所望する常連客、または他の人々にメッセージを送信して、例えば、ドリンクスペシャル、来るべき特別イベントなどを、それらの人々に知らせることができる。広告主は、ユーザによって提供される人口統計上の情報に基づき、特別オファーを供与することができる。ステップS2044で、ユーザは、他の人々の電子メールアドレス、MyTouchTunes受信ボックス、移動電話機などにメッセージを送信することができる。ステップS2044の代替として、またはステップS2044に加えて、ユーザは、仲間、または他の人々のグループにメッセージをブロードキャストして、例えば、来るべきパーティなどについてある特定のグループに知らせることができる。このため、例示的なシステムは、ユーザが送信するメッセージを介して、友人のネットワークについて知ることができる。加えて、ジュークボックスは、前述したフィーチャのためのインターフェースを直接に提供すること、または、ジュークボックスのロケーション(またはその範囲内)に無線デバイスを携えてくることが可能なユーザに、それらのフィーチャへの無線アクセスを提供することができる。また、それらのフィーチャの一部、またはすべてが、例えば、ユーザのホームコンピュータを使用して、例えば、インターネットまたは他の適切な通信ネットワークを介してジュークボックスサービスにユーザがアクセスすることを可能にすることにより、遠隔でユーザに提供されることも可能である。いくつかの例示的な実施形態では、通信は

、個々のユーザ、ユーザの事前定義されたグループ(例えば、家族、同僚、チームメート、友人など)に行われることが可能である。また、カスタムメッセージが、作成されることが可能であり、あるいは、より標準のメッセージ(例えば、「Meet us at X Bar」、「What are you doing tonight?」などの)が、選択されることが可能である。メッセージが、電子メールを介して送信されるか、SMSを介して送信されるか、IMを介して送信されるかなどにかかわらず、グループメッセージおよび/または規格メッセージが、実施され、使用されることが可能である。

【0114】

同様に、いくつかの例示的な実施形態は、登録されたユーザの接続を確立し、管理するためのシステムおよび/または方法を提供する。登録されたユーザが、ジュークボックスにおける、またはジュークボックスから遠隔の別のユーザ、または見込みの連絡相手と結び付きを確立しようと努めることが可能である。いずれの場合においても、ユーザは、ログインするように促されることが可能である。ログインした後、ユーザは、見込みの接続について的情報を入力するように促される。この情報には、例えば、電子メールアドレス、移動電話番号、物理的地址、ジュークボックス特有のメッセージングサービス、および/または以上に類するものが含まれることが可能である。

【0115】

この情報が入力されると、確認メッセージが、見込みの連絡相手に送信されることが可能である(例えば、オーディオビジュアル配信ネットワークを介して)。確認メッセージは、登録されたユーザを識別し(例えば、名前、ログイン名で、ピクチャもしくはアバターを使用するなどして)、かつ/または登録されたユーザに関する連絡先情報(例えば、電話番号、電子メールアドレス、ジュークボックスログイン名など)を提供することが可能である。見込みの連絡相手は、その2人の間の結び付きが、確立されるべきかどうかを、例えば、その見込みの連絡相手が、登録されたユーザが誰であるかを本当に知っていることを検証することにより、判定することができる。

【0116】

見込みの連絡相手が、既にジュークボックスサービスの登録されたユーザであるのではない場合、見込みの連絡相手は、自身、ジュークボックスサービスの登録されたユーザになることを要求される可能性があり、一部のケースでは、必須とされる可能性がある。

【0117】

確認メッセージを受信した後、見込みの連絡相手は、登録されたユーザを接続として受け入れるかどうかを選択することができる。これは、例えば、受け入れコードを含むメッセージを送信すること、またはある特定のWebサイトまたはジュークボックスにログインすることにより、行われることが可能である。例えば、電子メールメッセージ、テキストメッセージなどが、確認として見込みの連絡相手から送信されることが可能である。受け入れコードは、登録されたユーザおよび見込みの連絡相手を一意に識別するある特定の英数字コードであることが可能であり、あるいは、単に、見込みの連絡相手の電子メールアドレスまたは移動電話番号であることが可能である(例えば、空白のメッセージ、「YES」メッセージ、英数字受け入れコードなどを有するSMSメッセージ、電子メールメッセージ、電話コールメッセージ、および/または他のメッセージを送信すること)。代替として、またはさらに、受け入れコードは、見込みの連絡相手がアクセスすることができるハイパーリンクに埋め込まれることが可能であり、かつ/またはコンピュータベースのインターフェース(例えば、Webサイト)またはジュークボックスインターフェースを介してアクセス可能であってもよい。さらに他の例示的な実施形態では、見込みの連絡相手は、埋め込まれた受け入れコードを有さない、実行されるハイパーリンクを使用せずに、Webサイトまたはジュークボックスにログインしなければならない可能性がある。

【0118】

受け入れコードは、見込みの連絡相手から送信されると、例えば、オーディオビジュアル配信ネットワーク内のサーバによって、有効性確認されることが可能である。有効性確認プロセスは、登録されたユーザが有効であるかどうか、見込みの連絡相手が有効である

かどうかなどを判定することを含むことが可能である。このプロセスに合格である場合、登録されたユーザと見込みの連絡相手の間の結び付きは、確立されることが可能である。

【0119】

2名の人々の間で結び付きが確立されると、その2名は、ジュークボックスおよび/またはジュークボックス関連のサービスを介して互いに通信することができる。例えば、その2名は、前述したとおり、メッセージを送信すること、クレジットを共有することなどができる。

【0120】

各ユーザは、どれだけ通信が受信されることが可能であるか、および/いつ通信が受信されることが可能であるかを決定することができる。例えば、ユーザは、ジュークボックス上で、またはジュークボックスから遠隔で提供されるインターフェースを使用して、単一の通信媒体へのメッセージを制限することができる。例えば、ユーザは、ある特定の電子メールアドレスでだけ(別のユーザが、代替の電子メールアドレスを提供しているものの)、メッセージを受信することを所望する可能性があり、その場合、メッセージは、その指定された電子メールアドレスに転送されることが可能である。別の例において、ユーザは、決して連絡を受けない、またはSMSメッセージによってだけ連絡を受けることを所望する可能性がある。さらに、ユーザは、メッセージが受信されることが可能な時間帯を制限して、例えば、メッセージが、あまりにも夜遅く、またはあまりにも朝早く送信されるのを防ぐことができる(例えば、そのことが、さもないければ、ユーザの迷惑になる可能性がある)。そのような場合において、それらのメッセージは、後の配信のためにジュークボックスまたはサーバにおいてキューに入れられることが可能である。ユーザは、アクセス制御リストの中で指定されていない限り、いずれの着信するコネクションも受け入れないことを指定することができる。このため、コネクションの間の通信が、管理され、かつ/またはフィルタリングされて、例えば、いつ、どこに、誰から通信が受信され得るかなどが指定されることが可能であることが理解されよう。すると、登録されたユーザは、コネクションリストを使用して、ジュークボックスを使用して、あるいはパーソナルコンピュータなどを使用してジュークボックスネットワークにログオンすることなどにより、そのネットワークにアクセスすることができる遠隔デバイスを使用して、そのリストの中の相手と通信することができる。

【0121】

図23Eは、1つの例示的な実施形態における付加価値コンテンツにアクセスするのに利用できる非限定的な、例示的なオプションを示す。ユーザは、いくつかの理由で、S2052に示されるとおり、外部Webサイト(好ましくは、パートナーの)にアクセスすることができる。例えば、広告主が、選択された特定の曲またはアルバムに関するバナー広告を配置して、関連する商品をユーザが注文することができるようにすることが可能である。広告主は、プロフィール情報を使用して、広告、電子メール、テキストなどを、潜在的にどこにでも(オンライン、ジュークボックス上など)に向けることができる。例えば、ジュークボックス使用が、1年にジュークボックスにある金額を超えて出費する個人に、いくつかの製品を向けるための「修飾子」として使用されることが可能である(目立つように、または目立たないように)。また、ユーザはある特定のバンドのWebサイトにリンクする外部Webサイトにアクセスして、シートミュージックまたはギター「タブ」もしくはベース「タブ」を購入する、歌詞を見ることなどもできる。また、ステップS2054で、ユーザは、リンクをたどって、iTunesまたはAmazon.comなどのサードパーティプロバイダから曲および/またはアルバムをダウンロードする/購入することができる。一部の例示的な実施形態では、このステップは、「Manage Music」スクリーンからリンクされることも可能である(ステップS2020)。さらに、ステップS2056において、ユーザは、好ましくは、有料で、カスタムCDを注文し、それらのCDを発送させることができる。ユーザは、これらの注文を、ユーザの再生リスト、ユーザのPickList、ユーザの仲間の再生リスト、前述したHotHitsなどに基づかせることもできる。

【0122】

前述した例示的なMyTouchTunesシステムは、任意の適合するシステム上でアカウントを利用可能にするための幅広い概念に関連して説明されてきたが、アカウントが、例えば、孤立して動作する単一のジュークボックスを含むジュークボックスの、より小さいサブセットに利用可能にされる他の例示的な実施形態も、企図されることを理解されたい。さらに、大きいグループが、孤立した1つのシステムであるかのように動作することが可能であり、あるいは、代替として、すべてのユーザアカウント/プロフィール、ロケーションなどの完全な知識を有することも可能である。

【0123】

図22は、1つの例示的な実施形態によるログインページの例示的なスクリーンショットである。ユーザが、電子メールアドレスをユーザ名領域2200に入力し、パスワードをパスワード領域2202に入力することが可能である。ショートカットとして、ユーザは、ドメインリスト2204から共通電子メールアドレスを選択することができる。この実施形態は、電子メールアドレスおよびパスワードを要求するが、他の実施形態は、電子メールアドレスであることを必要としないユーザ名を要求することが可能であることを理解されたい。同様に、ユーザ名およびパスワードは、例えば、ホームジュークボックスシステム上で要求されないことが可能であること、および、さらなる実施形態において、ユーザ名だけが、要求されることが可能であることも理解されたい。ユーザは、キーボード2206を使用して、要求される情報を入力することができるが、ホームコンピュータに接続されたキーボードなどの他の入力デバイスが、使用されてもよいことを理解されたい。また、図22に示されるログインスクリーンは、実際のジュークボックスデバイス上、コンピュータ上(例えば、Webインターフェース、カスタムソフトウェアなどを介して)、移動デバイス上などに現れることが可能であることも理解されたい。

【0124】

好ましくは、ログオンの成功により、ユーザにメインメニューが表示される。図23は、1つの例示的な実施形態による、認識されたユーザがログインした後に利用可能なフィーチャを示す例示的なスクリーンショットである。カスタマイズされたあいさつ(例えば、ユーザの名前)が、領域2300内に表示される。再生リスト2302、2304、および2306が、完全な再生リストを再生するのに要求されるクレジット数と一緒に、説明的な名前を伴って表示される。例えば、再生リスト2302を再生することは、7回の再生がかかる。好ましくは、再生リストは、公開であるものと考えられ、このことにより、他のユーザが、再生リストを見ることが許される。しかし、再生リスト、例えば、再生リスト2306は、プライベートであると指定されることも可能である。ユーザは、再生リストクリエータボタン2308を押すことにより、さらなる再生リストを作成することができ、再生リストマネージャボタン2310を押すことにより、既存の再生リストを編集することができる。再生リストマネージャは、ユーザが、例えば、曲を追加すること、および/または削除すること、再生リストを完全に削除することなどによって、既存の再生リストを変更することができるようにすることが可能である。

【0125】

ユーザは、仲間追加ボタン2314を押すことによって仲間を追加することにより、仲間リストを管理することができる。代替として、ユーザは、仲間削除ボタン2316を押すことにより、ある仲間を削除することができる。また、ユーザは、メッセージ送信ボタン2318を押すことによってユーザの仲間にメッセージを送信することにより、通信することもできる。このフィーチャは、いくつかの実施形態では、例えば、移動デバイスにSMSメッセージを送信すること、電子メールを送信すること、受信側ユーザが、次回にログインした際に現れるメッセージを送信することなどができる。さらに他の実施形態では、ユーザは、例えば、このフィーチャを使用して、曲を送信すること、クレジットを寄付することなどができる。インジケータ領域2320は、現在のユーザが、1つのメッセージを受信していることを示し、送信者を識別する。しかし、他の実施形態では、他の情報が、表示されることが可能であり、さらに完全な受信ボックス(例えば、フォルダ、転送などを有する)が、提供されることなどが可能である。クレジットインジケータ領域2322は、アカウントステ

ータスを提供する。この実施形態では、領域232は、ユーザが、20回の再生をオンラインで以前に購入しており、さらに5回の再生のために、つまり、総計で25回の可能な再生のために十分な金額をローカルジュークボックス端末装置に投入していることを示す。

【0126】

次に、バックグラウンドミュージックの改良と関係する本発明の別の態様を説明する。本明細書で説明されるローカルサーバおよびTune Centralフィーチャが、ジュークボックス上のバックグラウンドミュージックに関してまったく新しい世界を開く。現在、「代金の支払われた音楽なし」が、キューに入れられている場合、ジュークボックスは、DMXまたは衛星ラジオなどの、ジュークボックスに接続された補助システムの助けを借りて、バックグラウンド(アンビエント)ミュージックを再生することができる。代替として、ジュークボックスが、ジュークボックス独自の音楽でバックグラウンドミュージックを再生する場合、ジュークボックスは、通常、約2000曲の、再生に利用できる曲に限定される。また、これらの曲は、ジュークボックス利用客によって選択されるように利用可能でもある。従来のデジタルジュークボックスは、通常、より広範な音楽にアクセスするためにインターネットを介して遠隔ネットワークに接続される。したがって、これらのアプリケーションは、ローカルで利用可能である音楽以外のバックグラウンドミュージックサービスを提供するのに、よく適していない。というのは、そのようなバックグラウンドミュージックサービスを提供することは、バックグラウンドミュージックのために遠隔サーバからの絶え間ないダウンロードまたはストリーミングを要求するからである。

【0127】

しかし、外部サーバまたは大きい音楽パーティション(本明細書で説明されるTune Centralなどの)を備えたジュークボックスは、曲の、その大きいプールから再生すべきバックグラウンドミュージックを選択することを、その曲が、標準のインターフェースを介して再生に利用できない場合でも、行うことができる。このことは、ジュークボックスを使用するバックグラウンドミュージックに関して広大な音楽バラエティを可能にする。バックグラウンドミュージック選択は、ランダムに、またはジャンル、テーマ、時代、または他の基準に基づいて音楽を選択する高度なアルゴリズムを使用して、ジュークボックスによって自動的に行われることが可能である。また、バックグラウンドミュージックリストが、ディスクジョッキーなどによって事前プログラミングされて(バックグラウンド再生リスト)、中央サーバからジュークボックスに送信されることも可能である。曲の権利は、ペイパープレイモードにおいて利用可能な曲に関するのと、バックグラウンドミュージックに関するのとでは異なるので、本発明のある実施形態では、適切な曲の権利の利用可能性に基づき、ジュークボックスが、バックグラウンドミュージック(通常のジュークボックス再生と対比される)として再生されることが可能な曲の間で区別を行うことを可能にし、その結果、適切な印税計算も行うことができるようにする、各曲、または曲の集合(例えば、ABCレコード会社のすべての曲)に関する指標が、存在する。本発明のある実施形態において、ジュークボックスは、ビルボードヒット、クリスマス音楽、再生リスト(例えば、DJ John-summer 2005)などのジャンル、テーマ、指標(フラグ)に基づき、どのようにバックグラウンドミュージックが実行されるかを運用者が構成することを可能にするための高度なオプションを提供する。本発明の別の実施形態において、運用者は、バーテンダーが、従来のバックグラウンドミュージックシステムにおいてラジオ局を選択するのと同じように、例えば、リモートコントロールを介して、バーテンダーが、バックグラウンドミュージックモードをさらに構成することを可能にするように、ジュークボックスを構成する。このモードは、バックグラウンドミュージック機能を強化するために、ジャンルベースの、またはテーマベースの音楽または再生リストを提供する。運用者は、例えば、いずれのジャンル、テーマ、または再生リストが、バーテンダーに利用可能であるかを構成することができる。さらに、適切なソフトウェアと組み合わされたプログラマブルキーを有するリモートコントロールが提供されて、バーテンダー(または他の個人)が(例えば、ある特定のゾーンに関して)、あるバックグラウンドミュージック再生リスト、または、例えば、バーゾーンにおけるフットボールの試合のような特別なイベントのためのTVオ

オーディオ入力を選択することができるようにすることが可能であり、その結果、ジュークボックスデバイスを介して、便利で、効果的なバックグラウンドミュージックサービス、およびその他のサービスが提供される。

【0128】

このため、バックグラウンドミュージックサービスと同様に、音楽の「チャンネル」が、提供されることが可能である。ロケーションの所有経営者が、再生されるべきチャンネルを選択することができる。チャンネルのリストが、ロケーションによって定義されることが可能であり、かつ/またはコンテンツプロバイダ(例えば、ジュークボックスプロバイダ)によって定義されることが可能である。いくつかの例示的な実施形態では、チャンネルは、ジュークボックスに既に提供されている音楽のテーマまたはジャンル(例えば、ラップ、ロック、カントリーなど)に対応することが可能である。さらに、または代替として、ロケーションの所有経営者が、曲と一緒にグループ化して、ある特定のロケーションまたはロケーションのグループのための、またはそのようなロケーションにおけるカスタムチャンネル、および/または多数の無関係または無関連である可能性があるジュークボックスの間で共有可能なカスタムチャンネルを(例えば、一種の公共チャンネルとして)形成することもできる。ジュークボックスに提供される数千の曲は、例えば、このチャンネルフィーチャを介して、バックグラウンドミュージックとしての一貫性のあるグループの中で再生されることが可能である。例えば、ジュークボックスにまったく金が挿入されずに、および/またはキューの中のすべての曲が再生されてから所定の時間が経った後、ある音楽チャンネル(例えば、所有経営者、マネージャなどによって選択された)が、再生されることが可能である。チャンネルは、この例示的なケースにおいて、例えば、実質的に直接のユーザ入力なしに、活性化されることが可能であり、したがって、通常のジュークボックス動作からバックグラウンドモードへの切り替えは、通常の利用客および/または認定されたユーザ(例えば、マネージャ、所有経営者など)に実質的にトランスペアレントである。もちろん、ジュークボックスの認定されたユーザは、ジュークボックスが、通常の動作モードからチャンネルモードに切り替わるようにする、チャンネルモードから通常の動作モードに切り替わるようにする、チャンネルの切り替えを制御する、チャンネルの間で選択を行うなどのこともできることが理解されよう。

【0129】

多数の利点が、このフィーチャによって提供される。例えば、ジュークボックスに提供されるメディアインスタンスを再生するのに、補助メディア源は、まったく要求されず、その結果、例えば、ハードウェアの点、および月々の契約料金の点でさらなる費用を、一部は要求する可能性がある、衛星ラジオ、ブロードキャストラジオ、CDプレーヤ、および/またはその他の機器(例えば、ジュークボックスに加えて、またはジュークボックスの代わりに)の必要性が、低減される。それどころか、ジュークボックスチャンネルは、ジュークボックスから直接に再生可能な、数百、または数千さえものメディアインスタンスを提供することができる。このようにして、バックグラウンドミュージックの多数の異なるチャンネルが、ジュークボックスの中に格納された、またはネットワーク上で利用可能な音楽を使用して、ジュークボックス上で提供されることが可能である。すると、利用可能なチャンネルは、例えば、ジュークボックスと一緒に提供されるリモートコントロールを使用して、認定された個人によって選択されることが可能である。また、それらの様々なチャンネルは、要求された際に、ジュークボックス上で表示されることも可能であり、したがって、あるチャンネルが、リストから選択されることが可能である。

【0130】

さらに、従来のブロードキャストラジオチャンネル、および拡張された衛星ラジオチャンネルとは異なり、ジュークボックス利用客は、新たなチャンネルに単に合わせる以上のことによって、ある特定のチャンネルのコンテンツに支配力を働かせることができる。例えば、認定されたユーザ(例えば、所有経営者、マネージャなど)が、ジュークボックスリモートコントロールを使用して、曲を飛ばす、曲をリプレーする、チャンネルから曲を削除する、チャンネルに曲を追加するなどのことができる。現在、この機能は、ブロードキャストラジオ

チャンネルまたは衛星ラジオチャンネルに関連して可能ではない。ジュークボックスチャンネルは、オプションとして、この制御の振舞いから学習し、格付けシステムと同様に、人気のある曲をより頻繁に再生する、飛ばされた曲を再生する頻度を落とすなどといったように、チャンネルを適合させることができる。

【0131】

さらに、ジュークボックスチャンネルは、従来のブロードキャストラジオおよび衛星ラジオに関連するのと同じ種類の物理的問題を被らない。例えば、衛星ラジオは、悪い気象条件中に動作不能である可能性があるのに対して、ジュークボックスチャンネルは、外部条件とは実質的に独立に機能することができる。同様に、ジュークボックスチャンネルは、例えば、ジュークボックスチャンネルが、ジュークボックスが存在する場所ではどこであれ、再生されることが可能であるという点で、従来のブロードキャストチャンネルのアンテナに関連するのと同じ種類の地理的制約を被らない。実際、ジュークボックスチャンネルは、永続的な、または確立可能なインターネット接続の存在さえ要求せず、ただし、インターネット接続は、ストリーミングメディア、更新可能なチャンネルなどを有することが望ましい、いくつかの例示的な実施形態に関連して、有利に使用されてもよいことが理解されよう。

【0132】

このため、いくつかの例示的な実施形態は、デジタルダウンロードジュークボックスを動作させるシステムおよび/または方法を提供する。複数のチャンネルが、定義され、それぞれの前記チャンネルは、そのチャンネルに関連付けられた複数のメディアインスタンスを有する。それらの複数のチャンネルからあるチャンネルが、認定されたユーザによって選択される。ジュークボックスの1回または複数回の利用客による使用に、第1のモードにおいて応答が行われ、前記使用の少なくとも数回は、ジュークボックス上で再生するようにキューにメディアインスタンスを追加すること、および前記キューに入れられたメディアインスタンスを再生することを含む。ジュークボックスが、利用客の要求に応答しておらず、かつ/またはキューに入れられたメディアインスタンスを再生している場合、そのチャンネルにおけるメディアインスタンスは、第2のモードで再生される。第1のモードは、第2のモードとは異なり、第2のモードは、ジュークボックスの利用客によって直接に制御可能ではない。いくつかの例示的な実施形態では、第2のモードは、バックグラウンドミュージックモードであることが可能である(したがって、音楽は、より低い音量で再生されるなどする)。

【0133】

いくつかの例示的な実施形態では、メディアインスタンスは、認定されたユーザによってチャンネルに関連付けられることが可能であり、加えて、または代替として、メディアインスタンスは、それぞれの前記メディアインスタンスに関連するテーマおよび/またはジャンル(例えば、ラップ、ロック、カントリーなど)に応じて、チャンネルに関連付けられてもよい。

【0134】

あるチャンネルにおいてあるメディアインスタンスが再生される頻度は、認定されたユーザから受け取られた1つまたは複数の制御コードに依存して、変更されることが可能である。制御コードは、チャンネルによって再生されているメディアインスタンスを飛ばすこと、チャンネルによって再生されている、または再生されたばかりのメディアインスタンスをリプレーすること、およびチャンネルからあるメディアインスタンスを削除することの少なくとも1つに対応することが可能である。

【0135】

この場合も、他の場合と同様に、プログラミングされた論理回路(例えば、ハードウェア、ソフトウェア、ファームウェア、および/またはこれらの任意の適切な組合せ)の任意の適切な組合せが、そのような機能を可能にすることが可能である。さらに、これを実行するための命令が、任意の適切なコンピュータ可読記憶媒体上に格納されることが可能である。

【0136】

図24は、1つの例示的な実施形態における遠隔でシステムにアクセスするための例示的なプロセスを示す流れ図である。ステップS2402で、ユーザが、バーにローカルで収容されたジュークボックスとは異なるロケーションおよび/またはデバイスから、ジュークボックスシステムにログインする。好ましい実施形態では、ユーザは、Webブラウザを介してシステムにアクセスして、インターネットポータルへのアクセスを得る。しかし、例えば、移動デバイス(例えば、セル電話機、PDAなど)、遠隔コンピュータ上で実行されるべきスタンドアロンのアプリケーション、適切に構成されたジュークボックス、その他を介するなどの、システムにアクセスするための他の機構も可能であることが理解されよう。さらに、システムによって提供されるフィーチャおよび/またはサービスは、システムにアクセスするのに用いられる機構および/またはデバイスに依存することが可能であることが理解されよう。例えば、セル電話機を介して利用可能ではない一部のフィーチャが、パーソナルコンピュータ上のWebブラウザを使用する例示的な実施形態に利用可能である可能性があり、パーソナルコンピュータ上のWebブラウザを使用する例示的な実施形態に利用可能でない一部のフィーチャが、セル電話機を介して利用可能である可能性がある。この例に即して、ただし、やはり、限定的ではない例示的な目的で、パーソナルコンピュータ上のWebブラウザを使用する例示的な実施形態は、完全なフィーチャセットおよび/またはサービスセット(例えば、音楽ダウンロード、フォーラム、チャットルームへの参加など)へのアクセスを有することが可能であるものの、セル電話機を使用する例示的な実施形態は、音楽をダウンロードすること、およびテキストメッセージを受信することだけしかできないことが可能である。もちろん、様々なデバイスが、互いに、より結び付けられ、より強力になるにつれ、そのような区別は、もはや成立しなくなる可能性があることが理解されよう。

【0137】

ステップS2404は、ユーザが、既存のユーザとしてシステムによって認識されるかどうかを判定する。システムによって認識されないユーザは、好ましくは、ステップS2408におけるとおり、アカウントを作成するように促される。ユーザは、例えば、ユーザの名前、年齢、電子メールアドレスなどを供給するように要求されることが可能である。好ましくは、ユーザは、例えば、名前および/またはニックネーム、ユーザの年齢、ユーザのロケーション、および関連するイメージを含むプロフィールを作成することができる。これらのプロフィールは、システムの他のユーザによって閲覧可能であってもよい。一部の例示的な実施形態では、ユーザは、例えば、システムの一部のフィーチャに参加するのに先立って、登録し、かつ/またはプロフィールを作成するように要求されることが可能である。代替として、他の例示的な実施形態では、ユーザは、ユーザが、ユーザの情報、またはユーザの情報のいくつかの部分が、プライベートのままであることを要望することを指定することができる。アカウント情報を編集するための限定的でない例示的なスクリーンショットが、図26Cに提供され、図26Cの付随する説明は、後段で与えられる。ユーザが、ユーザのアカウントを作成した後、好ましくは、中央アカウントデータベースが、ステップS2410で更新される。他の例示的な実施形態では、中央アカウントデータベースは、保持されなくてもよく、アカウント情報は、例えば、ジュークボックス上、ユーザ自身のパーソナルコンピュータ上などの、別の場所に格納されることが可能である。

【0138】

ユーザが、ステップS2404で認識された場合、ユーザは、図25および図26Aに関連して、より詳細に説明されるステップS2406において、あるフィーチャおよび/またはサービスを選択することができる。一部の例示的な実施形態では、ユーザは、ユーザが契約するサービスによって限定されたシステムへのアクセスを有することが可能である。例えば、一部の例示的な実施形態において、ユーザは、パーソナルコンピュータからオンラインシステムへの無料で、無制限のアクセスを有することが可能であるが、PDAを介してサービスにアクセスするのに、割増料金を支払わなければならないことが可能である。このため、図示されていないステップにおいて、既存のユーザが、認識された場合でも、そのユーザが

、許可されていない仕方システムにアクセスしようと試みた場合、そのユーザは、システムにアクセスすることを許されない。このフィーチャは、一部の例示的な実施形態では、システム、および/またはシステムの基礎をなすコード、曲などに不適切にアクセスしようと試みる、登録された、または未登録のユーザによる許可のないアクセスをブロックすることもできる。

【0139】

図25は、1つの例示的な実施形態によるシステムにおける非限定的なリストのサービスおよび/またはフィーチャの1つまたは複数を選択するための例示的なプロセスを示す流れ図である。以下に説明されるフィーチャおよび/またはサービスに加えて、そのようなフィーチャおよび/またはサービスと組み合わせられた他のフィーチャおよび/またはサービスが、可能であり、本明細書で企図されている。図25は、図24に関連して前述したとおり、ステップS2406から始まる。好ましい実施形態では、以下に説明されるステップは、何らかの特定の順序で選択される必要はないことが理解されよう。例えば、ユーザは、他のユーザと通信する前に、音楽を検索しなくてもよい。また、以下に説明されるステップは、一部の例示的な実施形態は、単独で実行されても、組合せて実行されてもよいことが理解されよう。例えば、ユーザは、他のユーザと通信する前に、音楽を検索することを所望することが可能である。

【0140】

ステップS2502で、ユーザは、音楽を検索することができる。好ましくは、ユーザは、例えば、アーティスト、曲、アルバム、ジャンルなどの検索フィールドを単独で、または組合せて指定することができる。ある検索の結果を示す例示的なスクリーンショットが、図26Eに提供され、後段で説明される。音楽を検索した後、ユーザは、次に、例えば、見つかった曲を再生リストに追加する、それらの曲を購入する、またはそれらの曲を他の人々に推奨することができる。例えば、通常の使用は、ユーザが、バーにおける気を引く曲を憶えているが、その曲が、ダミアンマリー(Damien Marley)の最新アルバムからの「Zion」と関係があるレゲエの曲であったことだけしか憶えていないことを含む可能性がある。したがって、ユーザは、ステップS2502で、以下のパラメータを使用して検索することが可能である。すなわち、アーティスト、「Damien Marley」、曲、「Zion」、およびジャンル、「reggae」である。検索が実行された後、ユーザは、結果リストから「Road to Zion」を選択することができる。その曲を見つけた後、ユーザは、次回にユーザがバーを訪れた際、便利のように、その曲を再生リストに追加することができ、ユーザは、それらの曲を友人に推奨することができる。また、ユーザは、シングルを購入し、かつ/またはデジタルコピーをダウンロードすることもできる。特に大胆なユーザは、「Welcome to Jam rock」アルバム全体を購入することも可能である。

【0141】

ステップS2504で、ユーザは、ユーザの再生リストを管理することができる。ユーザが、どのように再生リストを管理することができるかを示す例示的なスクリーンショットが、図26Dに提供され、後段で説明される。簡単に述べると、ユーザは、例えば、新たな再生リストを作成すること、既存の再生リストを削除すること、および/または既存の再生リストを編集することができる。好ましい実施形態では、ユーザは、例えば、友人または一般の人々と再生リストを共有することも可能であり、ユーザは、一般に利用可能な再生リストを検索することもできる。好ましくは、再生リストは、一意の名前(例えば、「Joe's All-Time Favorites」、「Ghetto Grooves」、「Maximum Chill-Out」など)を有し、好ましくは、再生リストは、その再生リストに関連する複数の曲を有する。再生リストは、1つの曲だけしか含まないことも可能であることが理解されよう。さらに、一部の実施形態では、ユーザ、専門のDJ、またはその他の人々が、再生リストを作成して、販売することも可能であり、ユーザは、システムを介して、そのような再生リストの代金を支払い、ダウンロードすることもできることが理解されよう。

【0142】

ユーザは、ステップS2506で音楽を購入することができる。1つの例示的な実施形態では

、ユーザは、単一の曲、複数の曲、アルバム全体、または複数のアルバムを注文することができる。一部の例示的な実施形態では、ユーザは、ユーザの注文品を発送させることが可能であるのに対して、他の例示的な実施形態では、許可されたデジタルコピーが提供されることが可能である。さらに他の例示的な実施形態では、音楽を購入し、メディア再生デバイス(例えば、適切に構成されたPDA、セル電話機など)に、その音楽を直接にダウンロードすることができることが可能である。

【0143】

好ましくは、ユーザは、ステップS2508で、電子メールメッセージおよび/またはテキストメッセージを介して他の人々と通信することもできる。また、ステップS2508は、好ましくは、ユーザが、オンラインフォーラムおよびライブのチャットルームに参加することも可能にする。このステップを介して送信されるメッセージは、従来の電子メール受信ボックス、セル電話機などに配信されることが可能である。さらに、メッセージは、例えば、受信側ユーザが、最初にログインした際、メッセージが、送信された際、および/または受信された際などに、ジュークボックス上で表示されることが可能である。オンラインフォーラムおよびチャットルームは、例えば、ニュースグループ、NNTP(ネットワークニュース転送プロトコル)、Usenet、IRC(インターネットリレーチャット)などのプロトコルを使用しても、カスタムプロトコルを使用してもよいことが理解されよう。

【0144】

ときどき、所有経営者、またはシステムのマネージャは、コンテスト、プロモーション、または大安売りを開催することが可能である。ユーザは、ステップS2510を介して、そのような活動に参加することができる。例えば、所有経営者は、クーポンを印刷した誰もが、カバーチャージを回避することを許すことができる。別の例として、最良の再生リストの競争が存在することが可能である。これらの例示的なプロモーションは、単に例示を目的とし、他のコンテスト、プロモーション、大安売りなども可能であり得ることが理解されよう。

【0145】

ステップS2512で、ユーザは、そのユーザのプロファイルを編集することができる。前述の図24からのステップS2408および後段の図26Cの説明が、ユーザプロファイルに好ましくは、関連付けられる情報のタイプに関するさらなる詳細を提供する。ユーザは、ステップS2514においてシステムからログアウトすることができる。いくつかの例示的な実施形態において、ユーザは、事前定義された非活動期間の後、自動的にシステムからログアウトされてもよい。

【0146】

サンプルスクリーンショットを提供する図26Aないし図26Eは、例えば、単に例示を目的とする。これらの特定のフィールド、ボタン、グラフィックなど、およびこれらの位置は、限定することを意図するものではまったくない。それどころか、図26Aないし図26Eは、本発明の1つの好ましい実施形態だけを提示する。他の構成、および他の組合せも可能であり、それらの構成および組合せは、本明細書で企図されていることが理解されよう。また、ユーザは、いくつかの仕方では以下のスクリーンの間で移動することができ、「押す」という用語の使用は、1つのそのような実施形態を示すに過ぎないことも理解されよう。例えば、ユーザは、マウス、キーボード、タッチスクリーン、スタイラスペンなどを使用して、スクリーンの間で移動することができる。好ましくは、ユーザインターフェースは、限られた量の広告、または他の非中核の材料を有して、非常にスムーズであり、簡素化される。したがって、広告、調査などが、ユーザインターフェースを介してアクセス可能である場合、それらの広告、調査などは、ジュークボックス機能にリンクされ、かつ/またはそれらのリンクに結び付けられた特典にリンクされる。例えば、広告主が、広告主の広告の1つをクリックするユーザに、クレジット、無料の再生、競争へのエントリーなどで報酬を与えることが可能である。同様に報酬は、調査に参加するユーザに与えられることが可能である。図26Aは、1つの例示的な実施形態による、システムにユーザがログインする前に利用可能なフィーチャを示す例示的なスクリーンショットである。ユーザは、ユー

ザ名フィールド2600およびパスワードフィールド2602にそれぞれ、ユーザ名およびパスワードを入力し、次に、ログインボタン2604を押すことにより、システムにログインする。代替として、未登録のユーザは、登録ボタン2606を押すことにより、アカウントを作成してもよい。ナビゲーションボタン2608は、ユーザが、システムにおいて迅速に移動するのに役立つ。この例示的な実施形態において、ナビゲーションボタン2608には、ホームボタン、アバウトボタン、FAQボタン、Find a jukeボタンが含まれる。ホームボタンは、ユーザを導入スクリーンに戻す。アバウトボタンは、好ましくは、例えば、ヘルプドキュメンテーション、使用条件などを含む、システムについての情報を表示する。FAQボタンは、好ましくは、アバウトボタンによって与えられる情報を補足する情報を提供し、システム、ならびに関連するサービスおよびフィーチャについて最も頻繁に尋ねられる質問に関する情報を提供するようにカスタマイズされる。

【0147】

Find a jukeボタンは、好ましくは、ユーザが、ジュークボックスを有するロケーションを検索することを可能にする。ユーザは、例えば、ユーザが旅行している可能性があるある特定の都市における、あるアドレス近くのロケーション、特別なフィーチャ(例えば、ドリンクスペシャル、ゲストDJなど)を提供するロケーション、ユーザが好む音楽を再生する可能性が高いロケーション(例えば、ラインダンスのあるバーが、カントリーミュージックファンに適切である可能性があるといったように)などを検索することができる。また、ソーシャルネットワークが、郵便番号、学校(例えば、高校、単科大学、総合大学など)などに基づいて構築されることも可能である。好ましい実施形態では、ユーザが、ある特定のロケーションを見つけると、ユーザは、カスタム情報、例えば、道順、カスタム地図、営業時間、駐車受信側ホストなどを表示させることができる。多くのロケーションに配置されたジュークボックスが、好ましくは、システムに接続され、ユーザの再生リストに、好ましくは、システムがアクセスすることができるため、ユーザは、ユーザの嗜好に合う可能性が高い推奨を受信することができることが理解されよう。例えば、システムは、ある特定のタイプの音楽を好むことが知られている同一の町におけるある年齢のユーザに、ある特定のバーを示唆することができる。いくつかの例示的な実施形態では、ジュークボックスは、ユーザらが、ある結び付きを形成することを推奨することができる。利用客(したがって、潜在的な「仲間」)の間の、これらの示唆される接続は、好ましくは、他のユーザらには必ずしも知られていないものの、ジュークボックスには知られている情報に基づくことが可能である。そのような情報は、例えば、よく訪れる会場、音楽の嗜好、年齢などを含むことが可能である。したがって、いくつかの例示的な実施形態では、ユーザらは、必ずしも、ユーザらが、どこ出身かではなく、どこに行くかに基づいて結び付けられることが可能である。

【0148】

同様に、ジュークボックスロケータサービスが、提供されることが可能であり、そのサービスは、ジュークボックス上、Webページを介して、MyTouchTunes Webインターフェースおよび/またはジュークボックスインターフェースなどを介して存在することが可能である。例えば、ユーザは、アドレス、またはアドレスの一部(例えば、都市、州、および/または郵便番号)を指定することができ、名前もしくはキーワードによって検索を行う、または特別に能力を与えられたジュークボックスを検索することができる。地図が、表示されて、マッチする各ロケーションが、ハイライトされることが可能である。さらなる情報が、所与のロケーションに関して表示されることが可能である。そのようなさらなる情報は、例えば、ロケーションの名前およびアドレス、および/またはロケーションについての他の任意の詳細を含むことが可能である。例えば、最も頻繁に再生される曲が、表示されることが可能であり、いくつかのプロモーションのオファー、紹介されるDJなどが表示されることも可能である。

【0149】

さらなる検索条件が、最初に、または検索を絞るように入力されることが可能である。そのようなさらなる検索条件は、ユーザの年齢、音楽の嗜好、選好されるテーマ(例えば

、アイリッシュバー、カントリーラインダンスなど)などを含むことが可能である。代替として、ジュークボックスロケータサービスは、ユーザによって入力された基準だけを使用して基本検索を行ってもよく、次に、例えば、前述のさらなる検索情報などの、例えば、ユーザについて既に知られている情報に基づいて、推奨されるロケーションをハイライトしてもよい。

【0150】

図26Fは、ジュークボックスロケータサービスを使用した検索の結果を示す例示的なスクリーンショットである。図26Fにおいて、検索基準は、検索ブロック2601の中に入力されることが可能である。検索は、例示的な検索基準2603a～dを使用して制約されることが可能である。地域の地図2605が、マッチするロケーション2607が、フラグを付けられて、またはそれ以外で区別されて、表示されることが可能である。ユーザは、フラグを選択することができる。この例では、フラグ2607'が、選択されると、選択されたロケーションに関する情報を含む情報ブロック2609が、表示される。前述したとおり、情報ブロック2609は、ロケーションに関する前述の情報、または他の情報を提示することができる。

【0151】

ユーザは、検索フィールド2610の中に検索用語を入力することができる。さらに、ユーザは、検索パラメタリスト2612の中でリストアップされたカテゴリの1つにおいてだけ検索を行うことにより、ユーザの検索をさらに限定することができる。検索パラメタのフィールド2612は、単に例示を目的とし、例えば、年、人気度などの他のフィールドが、リストアップされるフィールドに加えて、またはそれらのフィールドの代わりに提示されてもよいことが理解されよう。ユーザは、検索ボタン2614を押すことにより、検索を行うことができる。ユーザは、代替として、検索フィールド2616に記入し、検索パラメタドロップダウンリスト2618の中の検索パラメタのリストから検索パラメタを選択することにより、検索を行ってもよい。検索の結果は、後段で説明される図26Eに従って表示されることが可能である。

【0152】

限定的ではない例示的なジャンルリストが、ジャンルリスト2620の中で与えられる。他のジャンルが、ジャンルリスト2620の中で提示されることが可能であり、また、この例示的な実施形態では、さらなるジャンルに、ジャンルリスト2620の中の「More...」ボタンをクリックすることによってアクセスが行われることが可能であることも理解されよう。ある特定のジャンルを押したユーザが、そのジャンルに合わせられたコンテンツを受信することが可能である。例えば、「Alternative」のジャンルを押すことにより、最も人気のあるオルターナティブのアルバム、オルターナティブミュージックの議論を専門とするフォーラムのリストなどが提示されることが可能である。

【0153】

最も人気のあるアルバムリスト2622は、最も人気のあるアルバムと関係する情報を含み、好ましくは、アルバムカバー、アーティスト名、およびアルバムの名前を表示する。最も人気のあるアルバムリスト2622は、いくつもの仕方で生成されることが可能である。例えば、最も人気のあるアルバムリスト2622は、システムのユーザの選好、アルバム売上高に関する商業データ、ある特定の地理区分内のアルバム売上高などに基づくことが可能である。好ましくは、最も人気のあるアルバムリスト2622は、ジャンルリスト2620から選択されたジャンルに基づいて変更されることが可能である。

【0154】

ユーザは、通信方法2624からある特定の通信方法を選択することにより、通信固有のコンテンツにアクセスすることができる。この例示的な実施形態において、ユーザは、グループで、かつ/またはフォーラムにおいて通信することができる。グループおよび/またはフォーラムへの参加は、登録を要求することが可能であり、そのグループ、および/またはそのフォーラムは、公開またはプライベートであると指定されることが可能であることが理解されよう。好ましくは、グループは、近くのロケーションにおける、例えば、類似

した関心を有するユーザを含む。例えば、いわゆる「デッドヘッド(Dead Heads)」のためのグループが存在して、グレートフルデッドの来るべきツアーのために集合地点、交通手段などを「デッドヘッド」が計画することができるようにすることが可能である。フォーラムは、好ましくは、より静的であるが、同様に機能して、ユーザ間のスレッド化されたメッセージを可能にしてもよい。

【0155】

ブレッダクラムトレール2626が、システムにおいて移動するための便利な仕方を提供する。具体的には、ブレッダクラムトレール2626は、あるページにアクセスする際にユーザがたどる経路の跡をたどる。ブレッダクラムトレール2626からのある特定のブレッダクラム部分を押すことにより、好ましくは、ユーザは、対応するロケーションに連れて行かれる。

【0156】

情報領域2628および情報領域2630は、さらなる情報をユーザに提供するようにカスタマイズされることが可能である。この例示的な実施形態において、情報領域2628は、特定のフィーチャ(この場合、カスタム再生リストを作成することと関係する)を使用する理由を示唆し、情報領域2630は、役に立つ助言(この場合、カスタム再生リストをどのように作成すべきかと関係する)を提供する。情報領域2628および情報領域2630は、例えば、ユーザ、アクセスの時刻、所与のページにユーザがアクセスした回数などに依存して、変更されることが可能であることが理解されよう。さらに、情報領域2628および2630は、例えば、広告、プロモーションなどの、システムの使用と関係のない様々な情報を含んでもよいことが理解されよう。

【0157】

図26Bは、1つの例示的な実施形態による、システムにユーザがログインした後に利用可能なフィーチャを示す例示的なスクリーンショットである。ユーザのプロファイル情報に基づき、ユーザ情報2632およびユーザアイコン2634が、表示されることが可能である。この例示的な実施形態では、ユーザ情報2632は、ユーザ名と、ロケーションとを含み、ユーザアイコン2634は、ペタの正方形である。さらなる情報、および/または代替の情報が、表示されることが可能であることが理解されるときに、ユーザによって提供された情報だけが、表示されてもよいことも理解されよう。ユーザは、プロフィール編集ボタン2636を押すことにより、ユーザ情報2632およびユーザアイコン2634を含め、ユーザのプロファイルを編集することができる。ユーザは、ログアウトボタン2638を押すことにより、システムからログアウトすることができる。いくつかの例示的な実施形態において、ユーザは、事前定義された非活動期間の後、自動的にログアウトされてもよい。ユーザ再生リスト2640は、ユーザが作成した再生リスト、および/またはインポートした再生リストを表示する。ユーザ再生リスト2640は、ユーザが、再生リストをまったく作成していない、またはインポートしていない場合、空であることが理解されよう。好ましくは、例示的な実施形態は、再生リストのそれぞれに関してユーザによって指定された一意の名前を表示する。ユーザは、ユーザ再生リスト2640から1つまたは複数の特定の再生リストを展開して、その再生リストに関連する曲を示すことができる。図26Bに示される例では、「Alternative」再生リストが、展開され、4つの曲2642が、「Alternative」再生リストに関連するものとして示されている。ユーザは、複数の再生リストを展開すること、および/またはリストが展開された後、リストを折りたたむことができることが理解されよう。

【0158】

好ましくは、システムは、少なくとも1つの推奨2644を表示する。推奨は、好ましくは、アルバムカバー、曲名、アーティスト情報、および以上に関連するキーワードを含む。推奨は、好ましくは、自動的に生成され、様々なデータに基づくことが可能である。推奨を生成するためのデータには、例えば、ユーザ人口統計データ(例えば、年齢、性別、人種など)、ユーザの再生リストの中の曲の数およびタイプ、システムを介してユーザが購入した曲、人気のある曲などが含まれることが可能である。好ましくは、システムは、ユーザが、追加ボタン2646を押すことにより、推奨される曲を再生リストに迅速に追加するこ

とを可能にする。

【0159】

図26Cは、ユーザが、どのようにアカウント情報を更新することが可能かを示す例示的なスクリーンショットである。統計ボックス2648が、一般に利用可能な基本情報を表示する。この例示的な実施形態では、この情報には、ユーザ名、本当の名前、性別、ロケーション、および年齢が含まれる。しかし、他の実施形態は、さらなる情報、および/または代替の情報を要求することが可能であることが理解されよう。また、ユーザは、一部の情報(例えば、本当の名前、年齢など)をプライベートに保つことを選択してもよいことが理解されよう。

【0160】

初期アカウントセットアップにおいて選択されたユーザ名が、ユーザ名フィールド2670として表示される。好ましくは、ユーザは、初期セットアップの後、ユーザ名フィールド2670を変更することができず、好ましくは、各ユーザ名は、固有である。ユーザは、電子メールアドレスフィールド2652の中に電子メールアドレスを入力することができる。好ましい実施形態では、システムは、重要な電子メール(例えば、契約条件情報、大安売りの知らせ、請求書など)を、この電子メールアドレスに送信する。本当の名前は、名前フィールド2654の中で指定されることが可能である。この例示的な実施形態は、名前フィールド2654を単一のフィールドとして示すが、複数のフィールド(例えば、ファーストネームおよびラストネームのための複数のフィールド、ファーストネーム、ミドルネーム、およびラストネームのための複数のフィールドなど)が、同一の情報を代わりにキャプチャしてもよいことが理解されよう。ユーザは、性別オプション2656から性別を選択することができる。

【0161】

ロケーションが、ロケーションフィールド2658の中で指定されることが可能である。この例示的な実施形態は、ロケーションフィールド2658を単一のフィールドとして示すが、代わりに複数のフィールド(例えば、都市および州のための複数のフィールド、都市、州、および郵便番号のための複数のフィールド、都市、郡、および国のための複数のフィールドなど)が、同一の情報、またはさらなる情報をキャプチャしてもよいことが理解されよう。生年月日が、生年月日フィールド2660の中で指定されることが可能である。この例示的な実施形態は、生年月日フィールド2660を3つのフィールドとして示すが、単一のフィールド、または異なる順序の複数のフィールド(例えば、月、日、および年のための1つのフィールド、月および年だけのための1つのフィールド、月、日、および年のための1つのフィールドなど)が、同一の情報、またはさらなる情報を代わりにキャプチャしてもよいことが理解されよう。ユーザは、自由な形式の経歴フィールド2662に個人情報、または他の情報を入力することができる。さらに、ユーザは、フォーラムシグネチャフィールド2662の中でシグネチャラインまたはタグラインを指定することができる。好ましくは、システムは、グループおよび/またはフォーラムへのユーザの投稿に、フォーラムシグネチャフィールド2662の内容を自動的に付加する。ユーザは、ピクチャフィールド2666の中にファイルロケーションを入力することにより、ピクチャまたはアイコンをアップロードすることができる。好ましくは、このピクチャは、ログイン時に、かつ/またはユーザが、メッセージをグループまたはフォーラムに投稿した際に、表示される。

【0162】

好ましい実施形態では、必須のフィールドは、例えば、アスタリスク(*)を付けられる、ハイライトされる、またはそれ以外で指定されることが可能である。他の実施形態において、ユーザは、例えば、クレジットカード情報、料金請求アドレス、および発送アドレスなどの購入情報をさらに格納することができる。この情報は、例えば、ユーザが、曲、アルバム、または他の商品を注文する、またはダウンロードすることを決定した場合、アカウントの借方に自動的に記入するのにも使用されることも可能であり、したがって、ユーザは、バーおよび/またはクラブなどにおいてジュークボックスを使用しながら、「今、再生し、後に支払う」こともできる。

【0163】

再生リストマネージャボタン2668が、図26Dに関連して以下にさらに詳細に説明される、再生リストマネージャフォームへの迅速なアクセスを提供する。折りたたみ-展開ボタン2670が、この例示的な実施形態では、表示の左端に示される再生リスト情報を、ユーザが迅速に変更することを可能にする。推奨2672が、好ましくは、前述した仕方で生成され、1つまたは複数の推奨をクリックすることが、好ましくは、ユーザが、それらの推奨をユーザ自身の再生リストに迅速に、容易に関連付けることを可能にする。

【0164】

図26Dは、ユーザが、どのように再生リストを管理することが可能かを示す例示的なスクリーンショットである。この例示的な実施形態において、ユーザのカスタム再生リストが、水平に表示され(再生リストラベル2674として)、再生リストの様々な内容が、適切な見出しの下に表示される(再生リスト内容2676として)。もちろん、代替の構成も可能であることが理解されよう。好ましくは、各再生リストは、その再生リストに関連する固有の説明を有し、再生リストの中の各エントリは、再生リストの中の曲の数、曲の名前、およびアーティストを表示する。ユーザは、好ましくは、曲を再生リストに入れるように「ドラッグ」して、「ドロップ」すること、および/または再生リストから出すように「ドラッグ」して、「ドロップ」することにより、再生リストを編集することができる。ユーザは、検索機能を使用して、ある特定の曲を追加することができる。一部の実施形態では、所与の再生リストから曲のすべてを削除することにより、再生リストが空であることがユーザに通知されることになり、再生リストが削除されるべきかどうかユーザに尋ねられる。

【0165】

ユーザは、新規再生リスト作成ボタン2678を押すことにより、新たな再生リストを作成することができる。好ましくは、ユーザは、新たな再生リストに関する一意の名前を入力するように促される。次に、ユーザは、前述した仕方で再生リストを編集することができる。また、ユーザは、共有ボタン2680を押すことにより、再生リストを共有することもできる。共有ボタン2680は、ユーザが、ユーザの再生リストを公開と指定して、他のユーザが、それらの再生リストの内容を取り出すことができるようにすることを可能にし、好ましくは、ユーザが、例えば、公開、プライベートなどのカテゴリに、選択された再生リストをドラッグアンドドロップすることを可能にする。これに相応して、ユーザは、他のユーザによって作成された、DJなどによって準備されたといったような再生リストを検索し、次に、その再生リストをコピーおよび/またはインポートすることができる。ユーザは、再生リスト削除ボタン2682を押して、再生リストのリスト2684を表示し、次に、再生リストのリスト2684から、削除されるべき再生リストを選択することにより、再生リストを削除することができる。

【0166】

図26Eは、音楽の例示的な検索の結果を示す例示的なスクリーンショットである。この例示的な実施形態において、仮定的なユーザが、「portis*」を検索しており、ただし、アスタリスク(*)は、ワイルドカードを示す。好ましくは、検索結果は、アーティスト(アーティストリスト2686)、トラック/曲(トラック/曲リスト2688)、および/またはアルバム(アルバムリスト2690)によってグループ化される。好ましくは、グループ内の各結果は、その結果に関連付けられたランク2692を有することが可能である。このランクは、例えば、再生リストの中のアーティスト/トラック/アルバムを有するユーザの数、ビルボード(またはその他の)チャート上の現在の位置、過去の売上高などに基づくことが可能である。加えて、好ましい実施形態では、ユーザは、カテゴリ内の結果をアルファベット順に、または格付け順に並べ替えて、特定の結果を探し出すことを容易にすることができてもよい。ユーザの注意力持続時間が有限であるため、およびスペースの制約のため、検索の出力は、好ましくは、いくつかの(好ましくは、少なくとも5つの、かつ、好ましくは、10を超えない)結果に制限される一方で、モアボタン2694が、ユーザが、展開されたリストをブラウズすることを可能にする。

【0167】

図27は、移動デバイスに音楽をダウンロードするための例示的なシステムを示す。ユーザは、ジュークボックス2703にログインした後、ユーザが、ある曲、または複数の曲をダウンロードすることに関心があることを、移動デバイス2703を介して示す。多くのタイプの移動デバイス2702が、多くの異なる通信ネットワークを介してジュークボックス2703と通信することができることが理解されよう。例えば、適切に構成されたセル電話機が、セルネットワークのデータチャネルを介して通信することが可能であり、PDAが、標準のTCP/IPを介して通信することが可能であるといった具合である。ジュークボックス2703は、ネットワーク2704を介して通信して、音楽リポジトリ2706とインターフェースをとる。好ましくは、音楽リポジトリ2706は、広い範囲の潜在的なダウンロードを含む。いくつかの要求に関して、例えば、曲の複数の可能なバージョンが存在する場合(例えば、バンモリソンとジミーバフェットがともに、「Brown Eyed Girl」のバージョンを有するなど)、または確認目的で、ダウンロードを確認することが必要である可能性がある。音楽リポジトリ2706は、要求を受信した後、適切な曲(または複数の曲)を選択し、その曲を、ネットワーク2704を介して移動デバイス2702に送り返す。

【0168】

図27には反映されていないが、代替の実施形態は、適切な曲を別の指定されたデバイスに送信することができる。例えば、バーにおけるユーザが、ある曲を耳にして、その曲が、ユーザのホームコンピュータに送信されることを所望する可能性がある。車内のユーザらが、次回にユーザらがある特定のバーを訪れる際に、曲が利用可能であることを所望し、このため、例えば、曲が、遠隔ジュークボックスにダウンロードされることが可能である。一部の例示的な実施形態では、適切な曲2708が、適宜、異なる通信ネットワークを介して送信されることが可能である。

【0169】

ユーザは、ユーザが入力した料金請求情報に従って料金請求を受けることが可能である。例えば、ジュークボックスインターフェースを介して曲がダウンロードされるといつでも、格納されたクレジットカードに請求が行われることが可能である。別の例として、ユーザは、例えば、Pay-Palアカウントにリンクされた特別なジュークボックス特有のアカウントをセットアップすることができる。関連する態様において、ユーザは、ユーザのセル電話機(または他の適切なポータブルデバイス)を使用して、ジュークボックスとコードを交換することにより(例えば、SMSメッセージを介して)、メディアを購入する、ジュークボックスに登録することなどができる。例えば、ユーザはまず、ジュークボックスに対してユーザの身元を明らかにすることができる(例えば、ジュークボックス上で、または移動デバイスを介して遠隔でログインすることができる)。ユーザのセル電話番号(または、実施形態に応じて電子メールアドレスなど)に応じて、ユーザは、適切な情報を入力しなければならない可能性がある。SMS(または適宜、類似した)許可メッセージおよび/または許可コードが、ユーザに送信されることが可能である。すると、ユーザは、そのコードをジュークボックスに入力して、音楽を購入すること、個人用設定されたコンテンツを作成すること、認識されたユーザに利用可能なフィーチャにアクセスすることなどができる。また、この通信方法は、いくつかの例示的な実施形態において、ユーザが、音楽購入に関して、遠隔で事前検証すること、および/または事前支払いすることができるようにすることも可能である。

【0170】

図28は、音楽を取り出すための例示的なプロセスを示す流れ図である。ユーザは、ステップS2802におけるとおり、例えば、バー、クラブ、ラジオにおいて再生されている曲を聞く可能性がある。ユーザは、通常、曲を聞いた後、ステップS2804で、その曲を購入することを決定する。しかし、ユーザは、例えば、ユーザの友人が、その曲を推奨したため、ユーザが、ある特定のアーティストによる曲のすべてをダウンロードするためなどで、曲を聞かずに、その曲をダウンロードすることを選択する可能性があることが理解されよう。

【0171】

移動デバイスを使用して、ユーザは、ステップS2806で、ジュークボックスシステムにログインし、ステップS2808で、購入する曲を選択する。移動デバイスの使用は、迅速で便利な「衝動」買いを可能にする。一部の実施形態では、どの曲を購入するかを選択は、例えば、所与のロケーションにおいて現在、再生されている局、あるラジオ局で現在、再生されている曲、所与のロケーションにおいて再生されたばかりの局などを選択することにより、自動化されることも可能である。ユーザは、ステップS2810で、例えば、曲の代金をどのように支払うか、曲がダウンロードされるべきデバイスなどの購入パラメータを指定することができる。ステップS2812で、曲は、ジュークボックスインターフェースを介してユーザに配信される。曲は、例えば、ある特定のジュークボックス、ポータブルメディアデバイス、ラップトップ、ホームコンピュータなどの、他のデバイスに配信されてもよいことが理解されよう。

【0172】

図29は、ある特定のロケーション内からジュークボックスにアクセスする遠隔デバイスを有する改良されたダウンロードデジタルジュークボックスシステムのブロック図である。ジュークボックス16が、ある特定のロケーション内、ある特定のバー内などに配置される。ユーザは、ジュークボックス16上で実行されているユーザインターフェース24を介して、ジュークボックス16にアクセスすることができる。しかし、ユーザは、ジュークボックス16に遠隔でアクセスすることもできる。このため、複数の移動デバイス26a～26dが、特定のロケーション内に配置されているのが示されている。移動デバイス26a～26dは、それぞれ、ラップトップ、PDA、セル電話機、およびその他のパーソナルデバイスであるが、他の適切に構成されたデバイスが使用されてもよいことが理解されよう。各移動デバイス26a～26dは、それぞれ、無線送信機27a～27dを備える。移動デバイス26a～26dは、好ましくは、登録されたユーザが、ジュークボックスデバイスに有効にログオンし、例えば、カスタマイズされた再生リスト、個人用設定されたスクリーン、他の認識されたユーザからのメッセージなどの、特定のコンテンツにアクセスすることを可能にする。好ましくは、移動デバイス26a～26dの1つによって(例えば、ユーザ設定、再生リストセットアップなどに)行われた変更は、移動デバイス26a～26dのすべてによって反映される。例えば、ユーザが、ユーザインターフェース24を介して、ジュークボックス上で新たな再生リストを作成した、既存の再生リストを編集した、パスワードを変更したなどの場合、ユーザインターフェース26a～26fが、その変更を反映する。

【0173】

移動デバイス26a～26dは、外部ネットワークを介して、コミュニケーター25を有するジュークボックス16と通信することができる。移動デバイス26a～26dは、LAN、無線インターネット、Bluetooth、または他の任意の適切な通信ネットワークを介して通信することが可能であることが理解されよう。

【0174】

好ましくは、認識されたユーザ情報の中央データベースが、保持され、ジュークボックスデバイス16、16a～f、および遠隔デバイス26a～dのそれぞれによってアクセス可能である。しかし、一部の例示的な実施形態では、認識されたユーザ情報のローカルデータベースが、デバイス上に保持されることが可能である。好ましくは、これらのデバイスは、例えば、インターネットなどの通信ネットワークを介して互いに通信する。しかし、例えば、LAN上の有線通信、無線通信などを介するなどの、他の通信方法も可能であることが理解されよう。

【0175】

図29A～図29Eは、ジュークボックスに遠隔でアクセスするさらなる、限定的ではない例示的な構成を提供する。これらの例示的な実施形態の他の変種、および組合せも可能であり、本明細書で企図されていることが理解されよう。図29Aは、ある特定のロケーション内からジュークボックスにアクセスする遠隔デバイスを有する改良されたダウンロードデジタルジュークボックスシステムのブロック図である。この例示的な実施形態では、ジュー

ークボックス16、および複数の遠隔デバイスが、ある特定のロケーション内に配置される。ジュークボックス16は、ジュークボックスユーザが、例えば、選択を行うこと、プロフィールを作成すること、再生リストを編集することなどができるようにするユーザインターフェース24を含む。また、ジュークボックス16は、無線通信デバイス25も含む。遠隔デバイス26a～26bは、それぞれの無線通信デバイス27a～cを介してジュークボックス16と通信する。この例示的な実施形態では、遠隔デバイス27a～cは、それぞれ、ラップトップ、PDA、およびセル電話機である。他の遠隔デバイスが、ジュークボックス16と通信してもよいことが理解されよう。また、このシステムによる遠隔デバイスは、その特定のロケーションの外部から、ジュークボックス16と動作上、通信することもでき、その範囲は、ジュークボックス16上、および遠隔デバイス27a～c上のハードウェア、ならびにジュークボックス16と遠隔デバイス27a～cが通信するプロトコルによって決まることも理解されよう。このため、例えば、802.11g接続を介した無線通信は、ユーザが、ロケーション内、ロケーションの外部に設置されたテーブル、およびその区域に隣接する周囲の他の地域からジュークボックス16にアクセスすることができるようにすることが可能である。

【0176】

遠隔デバイス26a～cは、ユーザが、ジュークボックス16およびユーザインターフェース24に直接にアクセスする必要なしに、遠隔でジュークボックス16にログインすることを可能にする。このため、ユーザは、ユーザインターフェース24を直接使用することなしに、例えば、曲を再生し、再生リストを編集し、他のジュークボックス関連の活動を実行することができる。遠隔デバイス27a～cは、ユーザインターフェース24と同一であることも、異なることも可能な、デバイス27a～c独自のユーザインターフェースを有してもよいことが理解されよう。ユーザプロフィール情報が、ジュークボックス16上にローカルで、遠隔サーバ(図示せず)上に、または遠隔デバイス27上に格納されることが可能である。好ましくは、遠隔デバイスによって行われた変更(例えば、ユーザプロフィール情報、再生リストコンテンツ、購入されたメディアなどへの)は、ジュークボックス16上、他の任意の関連する遠隔デバイス上、および/または他のジュークボックス上で反映される。

【0177】

図29Bは、デジタルジュークボックスシステムにアクセスするある特定のロケーションの外部の遠隔デバイスを有する改良されたダウンロードデジタルジュークボックスシステムのブロック図である。この例示的な実施形態では、ローカルサーバ22およびユーザインターフェース24を有するジュークボックス16が、ネットワーク14に接続される。図29Aのジュークボックス16に直接に接続する遠隔デバイス26a～26cとは異なり、これらの遠隔デバイス26a～26cは、ネットワーク14にも接続される。このため、ユーザは、潜在的に広い地域にわたってジュークボックス16にログインすることができる。例えば、遠隔デバイス26a～26cは、LAN、WAN、インターネット接続などを介してジュークボックス16に接続することができる。ユーザプロフィール情報が、ローカルサーバ22上に格納されることが可能である。この場合も、好ましくは、遠隔デバイスによって行われた変更(例えば、ユーザプロフィール情報、再生リストコンテンツ、購入されたメディアなどに)は、ジュークボックス16上、他の任意の関連する遠隔デバイス上、および/または他のジュークボックス上で反映される。

【0178】

図29Cは、デジタルジュークボックスシステムにアクセスするある特定のロケーションの外部の遠隔デバイスを有する改良されたダウンロードデジタルジュークボックスシステムの別のブロック図である。図29Cは、複数の遠隔デバイス26a～26cが、ネットワーク14を介してジュークボックス16に接続されているという点で、図29Bと同様である。しかし、図29Cは、ネットワーク14に接続されたデータベース29を含む。データベース29は、例えば、ユーザプロフィール情報、ユーザの再生リスト定義などを格納することができる。この構成は、ネットワーク14に接続された複数のジュークボックスがすべて、常に利用できるわけではない可能性がある個々のジュークボックス、または個々の遠隔デバイスと接触することなしに、中央データベース29にアクセスすることができるため、一部の例示的

な実施形態において有利である可能性がある。

【0179】

図29Dは、第1のデジタルジュークボックスシステムにアクセスするある特定のロケーションの外部の別のデジタルジュークボックスシステムを有する改良されたダウンロードデジタルジュークボックスシステムのブロック図である。この例示的な実施形態では、ローカルサーバ22aおよびユーザインターフェース24aを有するジュークボックス16aが、第1のロケーション内に配置され、ローカルサーバ22bおよびユーザインターフェース24bを有するジュークボックス16bが、第2のロケーション内に配置される。これらのロケーションは、単一のバー内の異なる部屋、2つの別々の施設などであってもよいことが理解されよう。ジュークボックス16aとジュークボックス16bとともに、ネットワーク14を介して接続される。ユーザプロフィール情報が、ローカルサーバ22aと22bのいずれか、または両方において格納されることが可能である。一部の例示的な実施形態では、ユーザプロフィール情報が、1つのローカルサーバ上だけに格納される場合、1つのジュークボックスにログインしているユーザは、好ましくは、他方のローカルサーバ上に格納された情報にアクセスすることができる。他の例示的な実施形態では、ユーザプロフィール情報が、両方のローカルサーバ上に格納される場合、好ましくは、1つのジュークボックス上で行われたあらゆる変更が、その他のジュークボックス上で反映される。さらに他の例示的な実施形態では、ユーザプロフィール情報は、ユーザが、第2のジュークボックスにアクセスした場合に限って、第1のジュークボックスから取り出され、第2のジュークボックスに格納されることが可能である。これらの構成は、ユーザが、ユーザの情報にアクセスしない地域にユーザ情報を配布しないため、有利である。例えば、ワシントンからロサンゼルスにビジネスで旅行するユーザが、ロサンゼルスにおけるジュークボックスにログインすることによって、ワシントンで作成された、そのユーザの特定の情報にアクセスすることができるが、その仮定的なユーザは、モンテリオールにおけるジュークボックスにはログインしていないので、モンテリオールにおけるジュークボックスは、必ずしも、そのユーザの情報をローカルで格納していない。

【0180】

図29Eは、第1のデジタルジュークボックスシステムにアクセスするある特定のロケーションの外部の別のデジタルジュークボックスシステムを有する改良されたダウンロードデジタルジュークボックスシステムの別のブロック図である。図29Eは、ジュークボックス16aが、ネットワーク14を介してジュークボックス16bに接続されているという点で、図29Dと同様である。しかし、図29Eは、ネットワーク14に接続されたデータベース29を含む。データベース29は、例えば、ユーザプロフィール情報、ユーザの再生リスト定義などを格納することができる。この構成は、ネットワーク14に接続された複数のジュークボックスがすべて、常に利用できるわけではない可能性がある個々のジュークボックス、または個々の遠隔デバイスと接触することなしに、中央データベース29にアクセスすることができるため、一部の例示的な実施形態において有利である可能性がある。複数のジュークボックスが、ネットワーク14を介して接続されてもよいことが理解されよう。図30は、ユーザが、「検証されたユーザ」になることによってユーザのステータスを変更することができる1つの仕方を示す例示的な流れ図である。ステップS3002で、潜在的なユーザが、ジュークボックスアカウントを作成して、登録されたユーザとなる。前述したとおり、潜在的なユーザは、あるロケーションにおけるジュークボックス上で、または、例えば、パーソナルコンピュータ、ラップトップ、移動デバイスなどを使用することによってジュークボックスに遠隔でアクセスすることにより、アカウントを作成することができる。検証されたユーザのための集まりに予期せずに、または気の向くままに舞い込んだ潜在的なユーザが、ほとんど、またはまったく事前の計画なしに、即時に検証されたユーザになり、ユーザのアカウントを作成することを所望する可能性もあることが理解されよう。登録されたユーザが、ステップS3004で、集まり、特別イベントなどの通知を受け取る。そのような通知は、電子メールメッセージ、SMSテキストメッセージ、登録されたユーザがジュークボックスにローカルでログオンした際に表示されるメッセージなどを介して送られることが

可能である。いくつかの好ましい実施形態では、登録されたユーザは、それらのユーザが、事前定義された資格を満たす場合に限って、集まりについて知らせを受ける。例えば、ニューヨーク市の行政区に住んでいるすべての登録されたユーザが、マンハッタンにおける集まりの通知を受け取ることが可能であるが、デモインにおける登録されたユーザは、受け取らないことが可能である。別の例として、メタルミュージックを好むことを指定したユーザは、ゴージャスの集まりの通知を受け取らないことが可能である。

【0181】

登録されたユーザは、集まりに関する通知を受け取った後、ステップS3006におけるとおり、そのイベントに参加することができる。このステップには、バー、イベント、社交行事などに出席することがかわることが可能である。ステップS3008で、登録されたユーザは、検証されたユーザになる。このステップは、ユーザが、集まりの最中にジュークボックスにログオンすることを要求することが可能である。代替として、登録されたユーザは、後にジュークボックスに(ローカルで、または遠隔で)入力して、ユーザのステータスを検証済みに変更するパウチャー、コードなどを受け取ることが可能である。いくつかの他の実施形態では、登録されたユーザは、ユーザの名刺を出して、リストに署名して、またはそれ以外でユーザの出席を示して、ユーザのステータスを変更してもらうことができる。このため、ステップS3010において、ユーザは、検証されたステータスに基づき、特権を与えられること、アクセス権を変更させることなどが可能である。例えば、検証されたユーザは、検証されたユーザだけのための、いくつかのグループ/議論フォーラムへのアクセスを得ることが可能である。また、検証されたユーザは、いくつかの例示的な実施形態において、管理されていないポスティングを行うこともできる。他のいくつかの例示的な実施形態では、検証されたユーザは、他のユーザに利用可能でない特別なプロモーションにアクセスすることができることが可能である。1つのそのような特別のプロモーションは、登録されたユーザの携帯電話機にメッセージ(例えば、SMSメッセージ、電子メールメッセージ、ピクチャメッセージなど)を送信する能力を含むことが可能である。いくつかの例示的な実施形態において、1つまたは複数の無料クレジットのためのクーポンが、登録されたユーザに送信されることが可能である。このクーポン自体、多くの形態があることが可能である。例えば、あるロケーションにおける認定された個人(例えば、バーマネージャ、所有経営者)に示されるべき単純なメッセージが、提供されることが可能である。また、このクーポンには、例えば、無料クレジットが、誤って与えられる可能性を低くするのに役立つように、特別な有効性確認コードが含まれることも可能である。有効性確認コードは、認定された個人に示されることが可能であり、あるいはジュークボックスに直接に入力されることが可能である。このコードは、ジュークボックス上に配置された、またはジュークボックスから遠隔のコードのデータベースに照らして、有効性確認されることが可能である。このデータベースは、クーポンが読み取られた後、更新されることが可能である(例えば、同一のクーポンのさらなる使用を防止するように、そのクーポンの有効性が、取り消されてもよい)。代替として、またはさらに、このコードは、データベースが、調べられなくてもよいように、自己検査可能なフォーマットに従って作成されてもよい(例えば、コード自体が、何気ない観察者によって容易に特定可能であっても、そうでなくてもよい事前定義されたフォーマットに従って、クレジット数および/またはさらなる情報で符号化されてもよい)。

【0182】

また、有効性確認コードは、そのクーポンが、ある特定のロケーションまたはロケーションのグループに関してだけ有効であることを示すロケーションコードを含むことも可能である。例として、単一の個人によって所有されるすべてのロケーションにおけるすべてのジュークボックス上における再生のためのクーポンが、提供されることが可能である。同様に、有効性確認コードは、例えば、クーポンの受取人が、そのクーポンを引き換えなければならない期限の日数または絶対の日付および/または時刻などの、時間期限コードを含むことも可能である。例えば、そのコードは、1週間以内、ある曜日に限ってなどの使用に有効であることが可能である。

【0183】

単純なメッセージおよび/または有効性確認コードとは別に、またはそれに加えて、クーポンは、適切なハードウェアを使用してマネージャおよび/またはジュークボックスによってスキャンされることが可能なバーコードの形態で提供されてもよい。クーポンは、例えば、通常の郵便、電子メールなどを介して、テキストメッセージとして届けられてもよい。いくつかの例示的な場合では、クーポンは、バーコードを含むか、別の形態であるかにかかわらず、あるロケーションにおける引き換えのためにクーポンを印刷することが必要である可能性があるのに対して、いくつかの他の例示的な場合では、単に、クーポンを受信した移動デバイスを持ち運ぶことができることが可能である。

【0184】

このように、例示的な実施形態は、デジタルダウンロードジュークボックスのユーザにクーポンを届けるシステムおよび/または方法を提供する。有効性確認コードが、作成される。この有効性確認コードに関連するのが、ユーザのためのある数の無料クレジットである。特定の数が、有効性確認コードの中に含まれる必要はなく、つまり、例えば、クーポンの存在自体が、所定の数のクレジット(例えば、1クレジット、5クレジット、10クレジットなど)に値してもよいことに留意されたい。有効性確認コードは、クーポンに含まれる。有効性確認コードは、ユーザに伝送される。

【0185】

有効性確認コードに含まれるのが、ユーザが、クーポンを引き換えることができる少なくとも1つのロケーション、および/またはユーザが、クーポンを引き換えることができる時間に対する制限であることが可能である。有効性確認コードは、符号化された英数字シーケンスおよび/またはバーコードを含んでもよい。また、クーポンフィーチャは、ジュークボックス施設によって提供される食べ物、飲み物、および/または他の製品/サービスに対する割引を提供するクーポンのために使用されることも可能である。

【0186】

いくつかの例示的な実施形態において、有効性確認コードは、クーポンから得られることが可能である。次に、有効性確認コードが、有効であるかどうか判定されることが可能である。有効性確認コードが、有効である場合、ある数のクレジットが、有効性確認コードに関連付けられた無料クレジットの数に従ってユーザに提供される。有効性確認コードは、バーコードスキャナを介してジュークボックスによって獲得されることが可能であり、かつ/またはユーザが、ジュークボックスに提供されるユーザインターフェースを介して、有効性確認コードをジュークボックスに入力しなければならないことが可能である。代替として、またはさらに、ジュークボックスの認定されたユーザ(例えば、マネージャ、所有経営者など)が、無料クレジットを利用客に提供してもよい。

【0187】

この場合も、他の場合と同様に、プログラミングされた論理回路(例えば、ハードウェア、ソフトウェア、ファームウェア、および/またはこれらの任意の適切な組合せ)の任意の適切な組合せが、そのような機能を可能にすることが可能である。さらに、これを実行するための命令が、任意の適切なコンピュータ可読記憶媒体上に格納されることが可能である。

【0188】

図31は、登録されたユーザが、潜在的なユーザを照会することに関してクレジットをどのように受け取ることができるかを示す例示的な流れ図である。ステップS3102で、ユーザが、ジュークボックスに(ローカルで、または遠隔で)ログオンする。ステップS3104で、ユーザが、潜在的な将来のユーザのために、いくつかの連絡先情報を指定する。ステップS3104は、ユーザが、新たなアカウントを作成した際、登録されたユーザが、あるフォームにアクセスした際などに、アクセスされてもよいことが理解されよう。また、連絡先情報は、適宜、例えば、電子メール、テキストメッセージ、従来の郵便などを送るのに使用されてもよいことも理解されよう。ステップS3106で、招待が、提供された連絡先情報を使用してジュークボックスによって、潜在的な将来のユーザに送信される。最後に、ス

テップS3108で、照会する側のユーザが、クレジットを受け取る。いくつかの例示的な実施形態では、クレジットは、照会されたユーザの数に基づいて与えられることが可能である。いくつかの例示的な実施形態において、クレジットは、ボーナス(例えば、10名、25名、および/または50名の照会されたユーザが、登録されたユーザになると、ボーナスが与えられることが可能である)を含むことが可能な式(例えば、照会され、かつ/または登録された5名のユーザごとに1クレジット)に基づいて与えられることが可能である。

【0189】

図32は、登録されたユーザが、公開再生リストを作成することに関してクレジットをどのように受け取ることができるかを示す例示的な流れ図である。ステップS3202で、ユーザが、ローカルで、または遠隔でジュークボックスにログオンする。ステップS3204で、ユーザが、公開再生リストを作成する。前述したとおり、ユーザは、いくつかのメディアインスタンスを再生リストに関連付け、好ましくは、再生リスト全体に説明的な名前を与える。ユーザは、例えば、ドラッグアンドドロップフィーチャ、検索フィーチャなどを含め、前述した方法のいずれかを使用して、再生リストを作成してもよいことが理解されよう。いくつかの例示的な実施形態では、再生リストを公開と指定することが必要である可能性がある。

【0190】

ステップ3206で、ジュークボックスユーザは、ユーザの通常のジュークボックス活動の一環として、公開再生リストおよび曲にアクセスすることができる。ジュークボックスは、特に、アクセスされた曲および再生リストに関して、ユーザ活動を観察する。いくつかの例示的な実施形態では、ジュークボックスは、アクセスされた曲および再生リストと関係するカウント、パーセンテージなどを保持する。ジュークボックスは、ステップS3210で、他のユーザのアクセス数に基づき、ユーザに報酬を与える。例えば、ユーザは、ユーザの再生リストが、あらゆる再生リストの中で最もよくアクセスされている場合、ある閾値回数を超えてアクセスされている場合、事前定義されたカテゴリの中の他の再生リストより多くアクセスされている場合などに、報酬与えられることが可能である。いくつかの例示的な実施形態では、ユーザは、ユーザの再生リストの内容に基づいて報酬与えられることが可能である。例えば、ユーザは、ユーザの再生リストが、ジュークボックスによる判定により、事前定義された数および/または割合の最も人気のある曲を含むかどうかに基づいて、報酬与えられることが可能である。ユーザは、いくつかの仕方で報酬与えられることが可能であることが理解されよう。例えば、ユーザに報酬が与えられることが可能な1つの明らかな仕方は、ジュークボックス上のさらなる再生のためのクレジットをユーザに与えることによってである。また、ユーザは、いくつかのプロモーション、大安売りなどに参加させられることにより、報酬与えられることも可能である。より一般的には、認識されたユーザは、任意のジュークボックス関連の活動を行うことに関して(例えば、実際のジュークボックス上で、Webインターフェースを介してなど)、報酬ポイントを得ることができる。例えば、認識されたユーザは、ユーザが、ログオンする、メッセージを送信する、再生リストを共有するなどのたびに、ポイントを獲得してもよい。報酬ポイントは、例えば、ジュークボックス上の再生を購入するように引き換えられることが可能である。代替として、ポイントは、認識されたユーザによるジュークボックスのさらなる使用を促進するよう、特別な商品と引き換えられてもよい。

【0191】

オプションとして、ユーザは、ジュークボックス上で、かつ/または他の任意のジュークボックスによって仲介される環境において、ユーザを表すアバターを選択することができる。ユーザのアバターは、例えば、ピクチャとして、動くアニメーションとして、キャラクターとして、またはユーザを識別するのに適した他の任意の仕方で、グラフィックで表されることが可能である。アバターは、登録されたユーザに身体的に似ていても、似ていなくてもよい。

【0192】

一例において、ジュークボックスを操作する個人が、ジュークボックスからローカル再

生リストオブションを選択すると、ジュークボックス上の登録された各ユーザに関する再生リストが、利用可能にされることが可能である。この例は、登録されたユーザがアバターによって表された「Local Playlists」スクリーンの例示的なスクリーンショットである、図33に示される。登録されたユーザの名前、ニックネーム、または他の識別マーク3302a～eの隣にあるのが、その特定のユーザのアバター3304a～eである。これは、例えば、多くの利用客が知っている、または認識する可能性があるバーの常連客を示すことによって、ジュークボックスのさらなるカスタマイズおよび個人用設定をもたらし、偶然のジュークボックス利用客が、カスタム再生リストを作成することにより積極的になるように促すといったようなことが可能である。ユーザが、登録されたユーザに関連付けられ、アバターによって表されるローカル再生リストのグループを選択した後、そのユーザの再生リストが、表示されることが可能である。ジュークボックスを操作するユーザは、次に、アバターによって表されるユーザに関連付けられた個々の再生リストを選択することができる。選択されると、再生リストの内容(例えば、1つまたは複数の曲)が、ユーザのアバター、再生リストの名前、および再生に要求される再生回数と一緒に、表示されることが可能である。

【0193】

再生リスト、または再生リストの一部が、ユーザによって再生のために選択された場合、「now playing」スクリーンの例示的なスクリーンショットである、図34に示された、now playingスクリーンが、表示されることが可能である。このスクリーンは、アバター3304a、登録されたユーザの名前3402、再生リストの名前3404などを表示することが可能である。さらに、登録されたユーザに成り代わって、前置きのオーディオまたはビデオが、曲を紹介するのに、曲の間で遷移するのに、および/または再生リストを終了するのに再生されてもよい。例えば、アバター自体が、アニメ化されてもよい。アバターが、ピクチャである場合、スライドショーが、イメージがアニメ化されて見えるようにすることが可能であり、あるいは様々なイメージが、口の上に重ね合わされて、アバターが話している、または歌っているように見えるようにすることが可能である。アバターが、アニメ化されたキャラクタである場合、同一の、または類似した機能が、提供されることが可能である。例えば、アバターは、例えば、カスタムアニメーション、逆運動学技術などを使用して、音楽に合わせて歌い、かつ/または踊ってもよい(例えば、アバターは、登録されたユーザによってプログラミングされるとおり、またはジュークボックスによって自動的に生成されて、ヘビーメタルミュージックにヘッドバンギングし、カントリーミュージックに、他の現実の、またはシミュレートされたアバターと一緒にラインダンスしてもよい)。これらのフィーチャ、および/またはその他のフィーチャは、任意の適切なアバターアクションプログラミングされた論理回路(例えば、ハードウェア、ソフトウェアなどの任意の適切な組合せを含むジュークボックス上、または遠隔ジュークボックスインターフェース上の、あるいは、そのようなジュークボックスまたはインターフェースによってアクセス可能なプロセッサ)によって可能にされてもよい。一般に、アバターアクションプログラミングされた論理回路は、メディアインスタンスが再生される前、再生されている間、および/または再生された後に、アバターに、そのメディアインスタンスに反応させることができる。

【0194】

これらのフィーチャおよび/または類似したフィーチャは、登録されたユーザが、コンテンツ(例えば、ビデオコンテンツおよび/またはオーディオコンテンツ)をジュークボックスおよび/またはジュークボックス配信ネットワークにアップロードすることができるようにすることにより、提供されることが可能である。例えば、ピクチャ、オーディオファイル、映画などが、単一のジュークボックスによってキャプチャされ、かつ/またはそのジュークボックスに格納され、そのジュークボックス上の登録されたユーザに関連付けられることが可能であり、あるいは複数のジュークボックスが利用できるように、中央ロケーションに格納されることが可能である。

【0195】

これらのフィーチャ、および/または類似したフィーチャは、登録されたユーザが、ユーザのアバターに、どこからでもアクセスすることができるように、ネットワーク化されたジュークボックス環境において提供されてもよいことが理解されよう。これは、例えば、ユーザが、後に訪れる可能性がある故郷、大学、および/または他の場所に、例えば、帰るよう旅行する際に、役立つ可能性がある。このようにして、アバターは、登録されたユーザをかつて知っていた、他の利用客に馴染みがあってもよい。また、これは、ユーザ自身の音楽ライブラリへのアクセスをやはり有するなどしながら、創造的である(例えば、新たな友人をつくる)こと、または自身を紹介する(例えば、紹介のオーディオクリップおよび/またはビデオクリップを介して)ことを所望する、絶えず動いている登録されたユーザにも役立つ可能性がある。

【0196】

アバター、およびアバターによって提示されるメディアは、ジュークボックスに適用されるローカルフィルタに準拠するように強制されることが可能である。例えば、再生リストは、ロケーションが、ラップを許さない場合、選別されてもよく、事前記録されたメッセージが、露骨な言葉に関して編集されてもよく、不適切なヒクチャが、変更されてもよいといった具合である。

【0197】

MyTouchTunesサービスに関連して前述したとおり、適切に構成されたジュークボックスは、趣味のコミュニティ(例えば、特定の主題を中心に組織された)、空想のコミュニティ(例えば、ゲーム、創造的環境などを中心とする)、および/または関係のコミュニティ(例えば、オンライン会議スペース)を含め、様々な仮想コミュニティとのユーザの結び付きを容易にすることが可能である。したがって、ジュークボックスは、直接に、またはオンラインポータルを介して、電子メーリングリスト、オンラインチャットルーム、掲示板、メッセージボード、議論グループ、インターネットフォーラム、または類似物、ブログ、インスタントメッセージングサービス、ポッドキャスト、テキストメッセージングサービス、ファイル共有(例えば、P2Pファイル共有)、MUD(マルチユーザダンジョン)などの1つまたは複数を提供することができる。新たなジュークボックス中心の仮想コミュニティと結び付き、そのようなコミュニティを確立することにより、新たな利用客を引き寄せ、既存の利用客のさらなる関心を発展させることができる可能性がある。実際、メンバについての情報を収集すること(および広告および取引の申し出を適切な対象に向けること)、ペンダをコミュニティに引き付けて、メンバを引き寄せ、消費を促進すること、メンバをコミュニティに引き付け、メンバベースのコンテンツを生成すること、および/またはユーザ間対話を促進して、コミュニティへのユーザの忠誠を構築することにより、仮想コミュニティにおいて、また仮想コミュニティを介して、より高い利益を実現することができることが可能である。

【0198】

前述したアバターは、ジュークボックスによって仲介される仮想コミュニティに関連して使用されることが可能である。例えば、アバターは、ユーザが、コミュニティ内で行うすべてのアクション(例えば、ブログを書くこと、掲示板に投稿すること、メッセージを送信することなど)に関連付けられることが可能である。さらに、アバターは、ユーザが、ユーザ独自のメディアを作成して、共有することを始めるにつれ、より興味深くなる可能性がある。ユーザは、アバターが、そのメディアに仮想的に「付加」されて、または関連付けられて、音楽をアップロードすることができる(例えば、標準の使用許諾契約に準拠して)。ジュークボックスによって仲介されるネットワークは、Webサイト、P2Pファイル共有エンジン、および/またはジュークボックスへの音楽の配信を追跡することもできる。印税が、ユーザのアバターを介して追跡され、ユーザに戻されることが可能である。さらに、ユーザのタイプに基づき、異なる料金/印税構造が、提供されてもよい。例えば、独立のアーティストに関して、多額の料金と小額の印税は、適切でない可能性があるのに対して、大手のレコード会社に関連するアーティストは、そのような直接の費用および間接の費用を負担する能力がより高い可能性がある。管理を容易にするため、曲が、著作

権遵守に関して調べられることが可能であり、この認定プロセスは、自動的に実行されても(例えば、デジタル透かしが存在を調べる、知られている曲の集積との比較)、(例えば、中央分散ネットワークの認定されたユーザ、利用客、運用者、従業員などによって)手作業で実行されてもよい。コンテンツがアップされ、かつ/または契約に同意が行われる際、アバターが、著作権契約および/または使用許諾契約に付加される許容可能な形態のデジタル署名であることが可能である。また、アバターは、タグ付きの曲などの実際の再生中に、アルバムアートとして表示されてもよい。

【0199】

運用者によって使用されることが可能な前述したツールキットに加え、運用者、認定されたユーザ、および/または所有経営者は、構成ダッシュボードを利用して、ジュークボックスを運用すること、管理すること、更新すること、および/または、それ以外で制御することもできる。実際、1つまたは複数のジュークボックスを管理する運用者は、ジュークボックスに提供される診断フィーチャおよび/または管理フィーチャに遠隔でアクセスすることにより、時間および/または金を節約することができる。いくつかの例示的な実施形態において、運用者は、例えば、コンピュータ、ハンドヘルドデバイス、または他の適切なデバイスを使用して、ダッシュボードの管理モジュールおよび/または診断モジュールにアクセスして、独自の(例えば、Webベースの)インターフェースを介して、通知を受け取り、かつ/またはジュークボックスのステータスを調べることが可能である。通信は、双方向であって、運用者が、管理フィーチャおよび/または警報および/または通知に関して、関心対象のジュークボックスに直接に命令すること、および/または問い合わせることができるようにもよい。後段で説明されるフィーチャは、単独で使用されて、または様々な組合せで使用されて、運用者、認定されたユーザ、および/または所有経営者が、配慮を必要とする特定のジュークボックスに注目することを可能にして、移動時間、信頼性、およびジュークボックスの全体的な魅力を高めることもしながら、時間および金を節約することが可能である。そのようなフィーチャに関して、各ジュークボックスは、運用者が関心のある情報を収集し、かつ/または観察して、場合により、この情報を運用者に事前対応的に送信してもよい。通知機構は、所定の閾値を超えると、メッセージを(例えば、電子メール、SMS、または他の適切なフォーマットを介して)運用者に自動的に送信してもよい。これらの所定の閾値は、運用者によって設定されることが可能で、事前構成されることが可能である。

【0200】

診断および/または警報には、例えば、請求書拒否レート(例えば、パーセンテージおよび/または回数の点で)、スタックにおける請求書の数、再起動の回数、温度(例えば、ハードドライブ、CPU、システム、エンクロージャなど)などが含まれることが可能である。さらに、そのような警報/通知は、所定の時刻に送信されるように設定されることが可能である。例えば、CPU温度が、指定された時間にわたってある閾値を超えた場合、再起動の回数が、指定された時間内にある閾値を超えた場合などに、通知が送信されることが可能である。

【0201】

そのような診断技術は、ジュークボックスから遠隔に位置する運用者に提供されることが可能であるとともに、ジュークボックスにおいて直接に提供されることも可能であることが理解されよう。さらに、いくつかの例示的な実施形態では、第1のジュークボックスの診断モジュールを使用して、第1のジュークボックスから遠隔に配置された第2のジュークボックスの診断モジュールが遠隔で調べられることが可能である。

【0202】

図35～図37は、診断警報が、どのように構成されることが可能であるかを示す例示的なスクリーンショットである。特に、図35は、警報選好を設定するための例示的なスクリーンショットである。図35では、警報が送信されることが可能な電子メールアドレスが、ブロック3502で指定される。ドロップダウンボックス3504から曜日を選択することにより、毎週のレポートが、指定されることが可能であり、ドロップダウンボックス3506によって

指定される時刻に、毎日のレポートが、同様に送信されることが可能である。例示的なスクリーンショットの一番下の部分は、様々な試験、および電子メールが送信されるべき頻度をリストアップする。例えば、ドロップダウンボックス3508からの選択により、ハードディスク温度警報が、決して送信されるべきでないか、即時に毎日、送信されるべきか、毎週、送信されるべきかが指定されることが可能である。他の警報が、例えば、ドロップダウンボックス3510、3512(これらのボックス3510、3512は、展開されたドロップダウンボックス3508によって部分的にブロックされる)を設定することにより、同一の間隔で、または異なる間隔で送信されることが可能である。もちろん、他の間隔が、指定されてもよく、他の試験が、観察されてもよく、他のアドレスまたは通知手段(例えば、SMSメッセージ、インスタントメッセージなど)が、使用されてもよいことが理解されよう。

【0203】

図36は、警報閾値を設定するための例示的なスクリーンショットである。図36の左側は、観察されるべき特定の統計(例えば、請求書拒否レート、請求書スタック充填レート、再起動の回数など)を示すのに対して、右側は、閾値が指定されることを可能にする。閾値は、必要に応じて、絶対数および/またはパーセントとして設定されることが可能である。他の試験が、観察されてもよく、任意の適切な閾値が、指定されてもよいことが理解されよう。さらに、まったく閾値を示さないことを使用して、ある特定の試験に関して警報が生成されるのが回避されることも可能である。図37は、様々な試験のステータスを示す例示的なスクリーンショットである。図37において、スクリーンの左側は、特定の試験をリストアップするのに対して、スクリーンの右側は、その試験のステータスを示す。試験は、オプションとして、便宜のために分類されてもよい(例えば、温度、料金収集、ソフトウェア動作など)。図37に示されるとおり、ステータスは、色によって示されてもよい(例えば、緑が、まったく問題がないこと、または許容可能な範囲内の動作を表すことが可能であり、黄が、何らかの懸念材料を表すことが可能であり、赤が、問題の発生または切迫を表すことが可能である)。もちろん、色で識別するスキームの代わりに、他の標識が、使用されてもよい(例えば、数値システムまたはパーセンテージシステムなど)。

【0204】

また、拡張されたスケジューリングモジュールが、ジュークボックスに提供されることが可能である。このモジュールは、運用者および/または他の認定されたジュークボックスユーザが、単一のイベント、および/または繰り返すイベントをスケジュールすることができるようになることが可能である。例えば、バーオーナーが、変形、音楽フィルタ、および/またはバックグラウンドミュージックをスケジュールすることができる。イベントの繰り返しは、毎週、毎月、毎年などであることが可能である。例えば、バックグラウンドミュージックが、パンチ中に再生されることが可能であり、音楽フィルタが、午後および夕方中に適用されることが可能であり、無制限の再生リストが、夜に提供されることが可能である。別の例として、ジュークボックスが、80年代音楽の夜、ヒップホップ音楽の夜、カントリーラインダンスの夜、ディスコの夜などに適応するように、夜ごとに変形されることが可能である。そのような高度なセットアップは、最終的に、時間を節約し、便利さを向上させることが可能である。イベントスケジューラは、毎月のビュー、および/または毎日のビューを含むことが可能であり、毎日のビューは、例えば、時間によって細分されている。新たなイベントが追加された後、適切なビューが、更新されることが可能である。

【0205】

図38は、拡張されたスケジューリングモジュールにリンクされた変形能力の例示的なスクリーンショットである。変形能力に関して、ジュークボックスは、事前定義されたマスタジュークボックスに(例えば、入力領域3802を介して)、特定のジュークボックスに(例えば、入力領域3804を介して、例えば、所有経営者または運用者によって前に定義されたジュークボックス)、または新たなカスタム音楽リストに(例えば、入力領域3806を介して)変形されることが可能である。特に、ジュークボックスの各変形に関するスナップショットが、とられることが可能であり、関連するデータ(例えば、日付、時刻、再生リスト)

が、そのスナップショットとともに格納される。これは、変形プロセスを容易にすることが可能であり、また、所有経営者が、変形を(例えば、それらの変形が、あまり人気がない、それほど成功ではないなどの場合に)「アンドゥする」ことができるようにすることも可能である。次に、ユーザは、「schedule this setting」ボタン3808を選択することにより、他のイベントと同様に、変形をスケジュールすることができる。

【0206】

「schedule this setting」ボタン3808は、他のイベントスケジューリングの場合と同様に、カレンダースケジューリング構成要素を立ち上げることができる。図39は、スケジューリングカレンダーの例示的なスクリーンショットである。スケジューリングカレンダーは、月ごとのビュー3902とともに、毎日のビュー3904も有することが可能である。毎日のビュー3904は、半時間単位、1時間単位などで細分されてもよく、ある特定の日に関する他の任意のイベントスケジュールを示してもよい。

【0207】

また、運用者は、サブユーザアカウントをセットアップしてもよく、そのようなサブユーザは、制限されたロケーションアクセスを有する。

【0208】

他の機能は、拡張されたロケーションセットアップフィーチャを含むことが可能である。

【0209】

本開示は、本発明を、ある特定の実施形態、特定のハードウェア構成要素およびハードウェア構成、特定のソフトウェア構成などに限定するものと解釈されるべきでない、いくつかの用語を使用してきた。例えば、例えば、多くのフィーチャおよび例が、バー内、パブ内、または他の環境内における、それらのフィーチャおよび例の存在に関連して説明されてきた。しかし、本発明の例示的な実施形態に存在するフィーチャは、ジュークボックス(または複数のジュークボックス)が配置されることが可能な任意のロケーションにおいて使用されるように適合可能であることが理解されよう。同様に、いくつかのフィーチャおよび機能が、「ユーザ」、「オーナー」、「運用者」、「利用客」などによる使用に関連して説明されるが、これらの用語は、一般的であり、選択される環境、および使用されるフィーチャに依存して、ほとんどの場合において、互いに区別なく使用されることが可能であることが理解されよう。例えば、初期の曲選択をオーナーおよび/または運用者に制限することが有利である可能性があるが、いくつかの例示的な実施形態では、利用客が、初期の曲選択においてある役割を果たしてもよい。

【0210】

さらに、実施形態が構築されることが可能な1つの仕方を表すに過ぎない特定のハードウェア組合せ、および特定のハードウェア構成が、開示されている。中央サーバは、一部の例示的な実施形態では、機能するジュークボックスを可能にするのに必要なあらゆる種類のサービスを一貫して提供するように、一緒に、または別々に動作する1つまたは複数のサーバから成ることが可能である。例えば、サーバのクラスは、仮想中央サーバを含み、1つのサーバが、メディアを提供し、別のサーバが、メンバシップを追跡し、さらに別のサーバが、使用許可を処理するといった具合であることが可能である。

【0211】

同様に、本明細書で説明されるローカルサーバは、ジュークボックスに組み込まれることが可能である。例えば、ローカルサーバは、ジュークボックス内部の組み込まれた大容量記憶デバイスの一部(例えば、パーティション)として存在するものの、独立して機能するように見えることが可能である。実際、ハードディスクが、より大きく、より安価になるにつれ、ハードディスクは、好ましくは、ローカルサーバの機能を果たすことが可能である。1つの現在、好ましい実施形態は、出荷時ドライブとして25,000の曲を含む120GBハードドライブを含む。しかし、現在、知られている、または今後、開発されるフラッシュメディア、または任意の記憶媒体(または、これらの組合せ)が、ある特定のジュークボックスアプリケーションに関するシステム設計者の所望に依存して、出荷時ドライブ、拡張

されたメディアストレージ、またはローカルサーバとして使用されてもよい。つまり、本発明は、任意の特定のストレージ技術またはストレージ構成に限定されるのではなく、任意の適切なストレージ技術を使用して実施されることが可能である。

【0212】

また、「曲」という用語が、ときとして、以上の説明において使用されてきたが、この用語は、本発明の範囲を限定することを意図しておらず、任意のメディアインスタンス、または任意の複数のメディアインスタンス(例えば、曲、ビデオ、曲/ビデオの組合せ、データ、情報など)が、本明細書の任意の実施形態において使用されることが可能であり、やはり、本発明の意図される範囲内に含まれる。

【0213】

最後に、本明細書で提示されるスクリーンショットおよびソフトウェア構成は、本明細書で開示されるフィーチャを編成し、表示するための1つの好ましい方法に過ぎないことが理解されよう。他の構成も可能であり、したがって、本明細書で企図される。

【0214】

本発明の好ましい態様が、例示され、本明細書で説明されてきたが、様々な変更および/または変形が行われることが可能であることが、当業者には明白であろう。このため、本明細書の特定の説明は、単に例示的であることが意図され、添付の特許請求の範囲より狭く本発明を限定することを意図するものではない。

【図面の簡単な説明】

【0215】

【図1】従来のダウンロードデジタルジュークボックスシステムのブロック図である。

【図2】好ましい実施形態による改良されたダウンロードデジタルジュークボックスシステムのブロック図である。

【図3】ジュークボックスシステムの好ましい実施形態による初期選択スクリーンを示す例示的なスクリーンショットである。

【図4】好ましい実施形態によるローカルサーバ上で曲を検索する際に使用するための例示的な検索スクリーンを示す別のスクリーンショットである。

【図4A】パーソナルミュージックアシスタントを使用して、ユーザ特有のプロファイルに適切である可能性がある曲を検索するための例示的なプロセスを示す図である。

【図4B】パーソナルミュージックアシスタントを使用して、認識されたユーザのプロファイルに適切である可能性がある曲を検索するための例示的なプロセスを示す図である。

【図4C】1つの例示的な実施形態によるユーザに利用可能なジャンルおよびテーマの非限定的なリストを示す例示的なスクリーンショットである。

【図5】好ましい実施形態による、ローカルサーバ上の検索の結果を示し、ある料金でジュークボックスデバイスに所望される曲をダウンロードするオプションをユーザに提供する別の例示的なスクリーンショットである。

【図5A】人気のある曲のリストの中を検索するための例示的なプロセスを示す図である。

【図6】ダウンロードフィーチャへのアクセスを可能にする代替の方法を示す別の例示的なスクリーンショットである。

【図7】ローカルサーバの好ましい実施形態を示すブロック図である。

【図8】商用ジュークボックス、住宅向けジュークボックス、ならびにジュークボックスシステムによって管理される他のダウンロードデバイス、および関連する接続を含む例示的な全体的ネットワークを示すブロック図である。

【図9】マルチゾーンジュークボックスシステムに関する例示的な施設レイアウトを上から見た図である。

【図10】各ゾーン内に選択端末装置を有するマルチゾーンジュークボックスシステムに関する例示的な施設レイアウトを上から見た図である。

【図11】マルチゾーンジュークボックスシステムに関するゾーン選択プロセスの例示的な実施を示す流れ図である。

【図12】マルチゾーンジュークボックスシステムに関するゾーン選択プロセスによる優先

再生の例示的な実施を示す流れ図である。

【図13】各ゾーンに関するキューのサブセットを伴う、優先キューおよび非優先キューのマルチゾーンセットの例示的な実施を示す図である。

【図14】変形能力を有するジュークボックスに関する例示的な配信-初期化スキームを示す流れ図である。

【図15】トリガイベントに基づく自動的ジュークボックス変形開始プロセスの例示的な実施を示す流れ図である。

【図16】ジュークボックス変形プロセスの例示的な実施を示す流れ図である。

【図16A】1つの好ましい実施形態において、ジュークボックスの変形が、どのように開始されることが可能であるかを示す例示的なスクリーンショットである。

【図16B】1つの好ましい実施形態において、ジュークボックスの変形前に、ユーザが、どのように様々な基準を指定することができるかを示す別の例示的なスクリーンショットである。

【図17】拡張されたメディアストレージを有するジュークボックスと、中央サーバとの間の関係を示す図である。

【図18】ある曲が、「標準の」利用可能な再生可能な曲リストの中に入っていない場合の曲選択プロセスに関する例示的なプロセスを示す流れ図である。

【図19】入札に基づく優先順位付け能力を有する優先再生キューに関する例示的なプロセスを示す流れ図である。

【図19A】ジュークボックス投票プロセスの例示的な実施を示す流れ図である。

【図19B(1)】投票キュー表示を示す例示的なスクリーンショットである。

【図19B(2)】投票キュー表示を示す別の例示的なスクリーンショットである。

【図19C】所与のロケーション内の投票機構を示す例示的なブロック図である。

【図19D】所与のロケーション内の投票機構を示す別の例示的なブロック図である。

【図19E】複数のロケーションで使用するための投票機構を示す例示的なブロック図である。

【図19F】複数のロケーションで使用するための投票機構を示す別の例示的なブロック図である。

【図20】1つの例示的な実施形態による音楽、通信などを管理するためのユーザに合わせたシステムの使用に関する例示的なプロセスを示す流れ図である。

【図21A】1つの例示的な実施形態におけるアカウント/プロフィール情報を編集するのに利用できる非限定的な、例示的なオプションを示す図である。

【図21B】1つの例示的な実施形態における音楽を管理するのに利用できる非限定的な、例示的なオプションを示す図である。

【図21C】1つの例示的な実施形態における仲間リストを管理するのに利用できる非限定的な、例示的なオプションを示す図である。

【図21D】1つの例示的な実施形態における他の人々と通信するのに利用できる非限定的な、例示的なオプションを示す図である。

【図21E】1つの例示的な実施形態における付加価値コンテンツにアクセスするのに利用できる非限定的な、例示的なオプションを示す図である。

【図22】1つの例示的な実施形態によるログインページの例示的なスクリーンショットである。

【図23】1つの例示的な実施形態による、認識されたユーザがログインした後に利用可能なフィーチャを示す例示的なスクリーンショットである。

【図24】1つの例示的な実施形態における遠隔でシステムにアクセスするための例示的なプロセスを示す流れ図である。

【図25】1つの例示的な実施形態によるシステムにおける非限定的なリストのサービスおよび/またはフィーチャの1つまたは複数を選択するための例示的なプロセスを示す流れ図である。

【図26A】1つの例示的な実施形態による、システムにユーザがログインする前に利用可能

なフィーチャを示す例示的なスクリーンショットである。

【図26B】1つの例示的な実施形態による、システムにユーザがログインした後に利用可能なフィーチャを示す例示的なスクリーンショットである。

【図26C】ユーザが、どのようにアカウント情報を更新することが可能かを示す例示的なスクリーンショットである。

【図26D】ユーザが、どのように再生リストを管理することが可能かを示す例示的なスクリーンショットである。

【図26E】音楽の例示的な検索の結果を示す例示的なスクリーンショットである。

【図26F】検索の結果を示す例示的なスクリーンショットである。

【図27】移動デバイスに音楽をダウンロードするための例示的なシステムを示す図である。

【図28】音楽を取り出すための例示的なプロセスを示す流れ図である。

【図29】ある特定のロケーション内からジュークボックスにアクセスする遠隔デバイスを有する改良されたダウンロードデジタルジュークボックスシステムのブロック図である。

【図29A】ある特定のロケーション内からジュークボックスにアクセスする遠隔デバイスを有する改良されたダウンロードデジタルジュークボックスシステムのブロック図である。

【図29B】デジタルジュークボックスシステムにアクセスするある特定のロケーションの外部の遠隔デバイスを有する改良されたダウンロードデジタルジュークボックスシステムのブロック図である。

【図29C】デジタルジュークボックスシステムにアクセスするある特定のロケーションの外部の遠隔デバイスを有する改良されたダウンロードデジタルジュークボックスシステムの別のブロック図である。

【図29D】第1のデジタルジュークボックスシステムにアクセスするある特定のロケーションの外部の別のデジタルジュークボックスシステムを有する改良されたダウンロードデジタルジュークボックスシステムのブロック図である。

【図29E】第1のデジタルジュークボックスシステムにアクセスするある特定のロケーションの外部の別のダウンロードデジタルジュークボックスシステムを有する改良されたダウンロードデジタルジュークボックスシステムの別のブロック図である。

【図30】ユーザが、「検証されたユーザ」になることによってユーザのステータスを変更することができる1つの仕方を示す例示的な流れ図である。

【図31】登録されたユーザが、潜在的なユーザを照会することに関してクレジットをどのように受け取ることができるかを示す例示的な流れ図である。

【図32】登録されたユーザが、公開再生リストを作成することに関してクレジットをどのように受け取ることができるかを示す例示的な流れ図である。

【図33】登録されたユーザが、アバターによって表された「Local Playlists」スクリーンの例示的なスクリーンショットである。

【図34】「now playing」スクリーンの例示的なスクリーンショットである。

【図35】警報選好を設定するための例示的なスクリーンショットである。

【図36】警報閾値を設定するための例示的なスクリーンショットである。

【図37】様々な試験のステータスを示す例示的なスクリーンショットである。

【図38】拡張されたスケジューリングモジュールにリンクされた変形能力の例示的なスクリーンショットである。

【図39】スケジューリングカレンダーの例示的なスクリーンショットである。

【符号の説明】

【0216】

10' ジュークボックスシステム

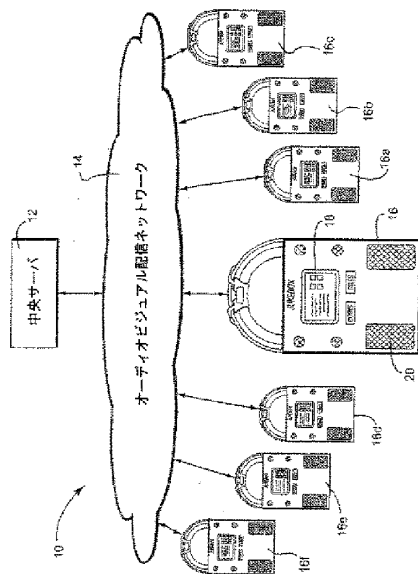
12 中央サーバ

14 オーディオビジュアル配信ネットワーク

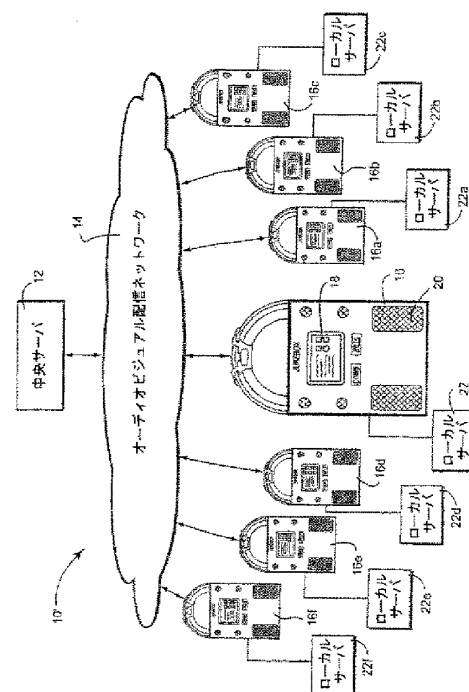
16、16a、16b、16c、16d、16e、16f ジュークボックスデバイス

- 18 スクリーン
 20 オーディオシステム
 22、22a、22b、22c、22d、22e、22f ローカルサーバ

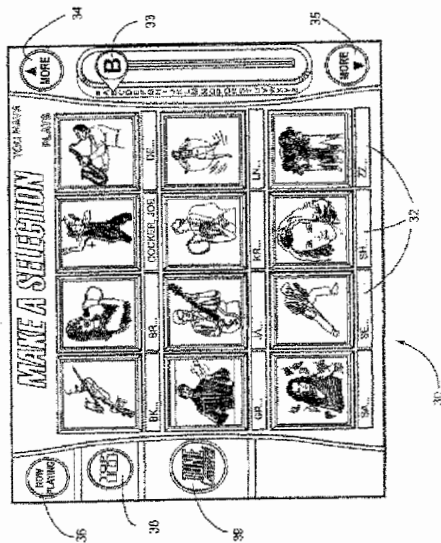
【図1】



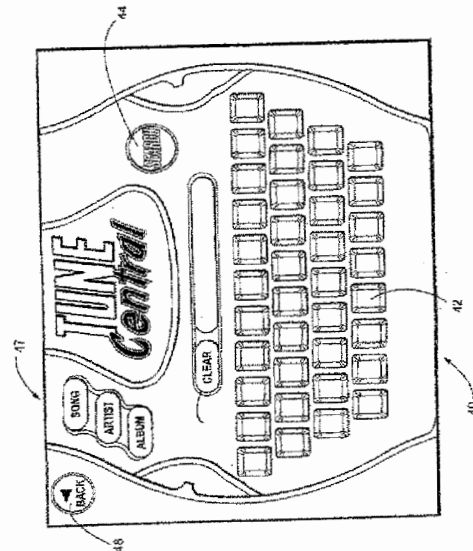
【図2】



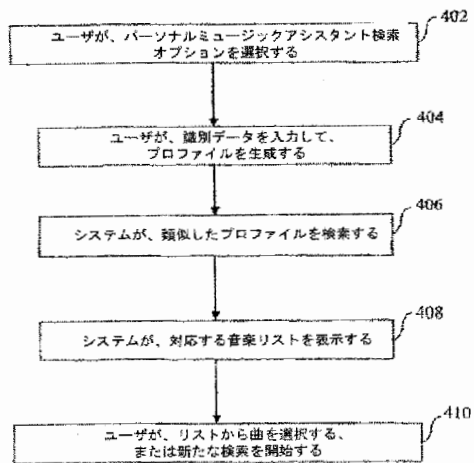
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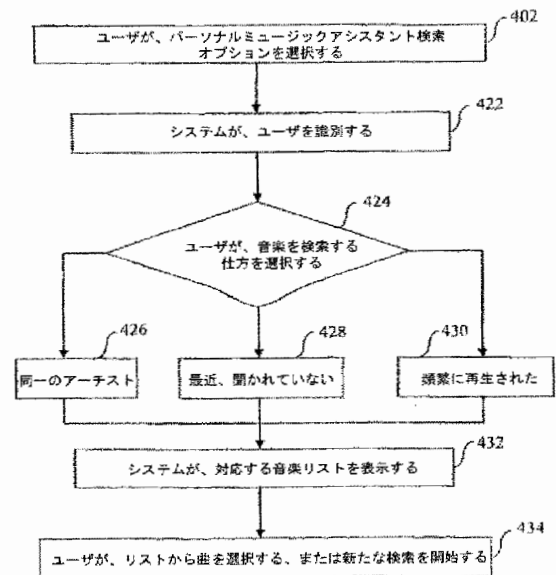
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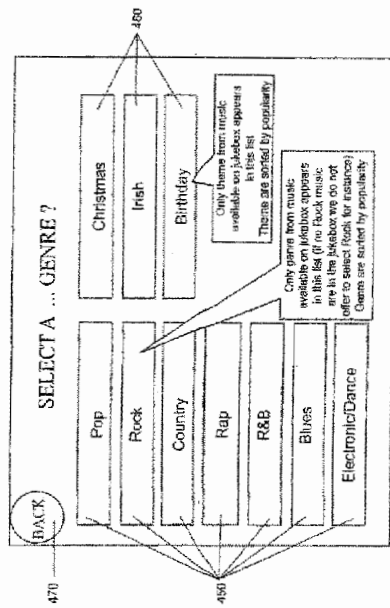
【図4A】



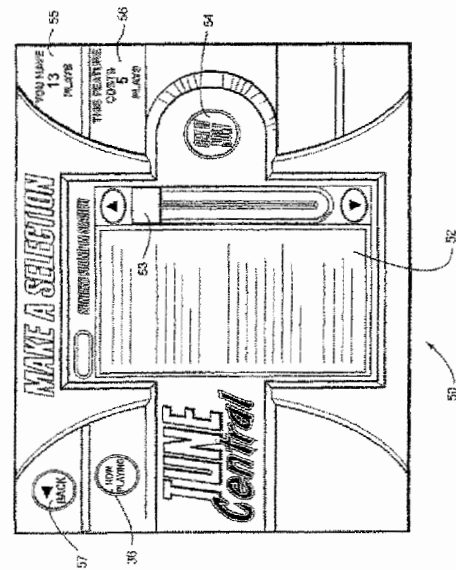
【図4B】



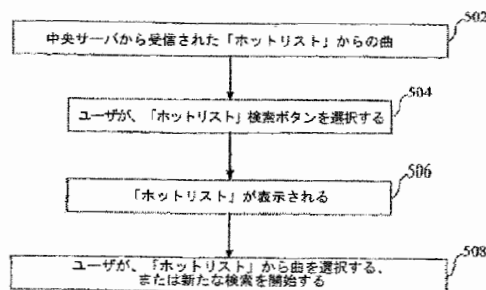
【図4C】



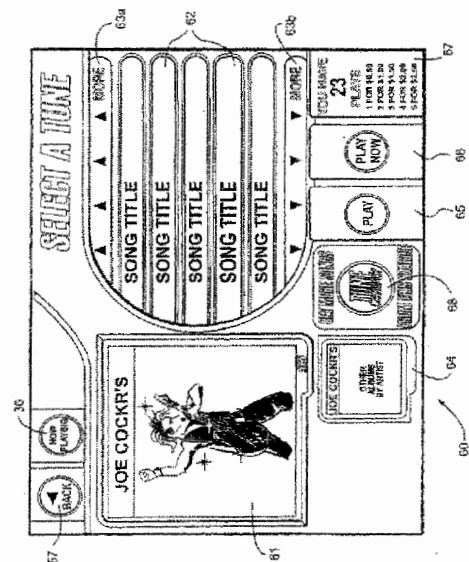
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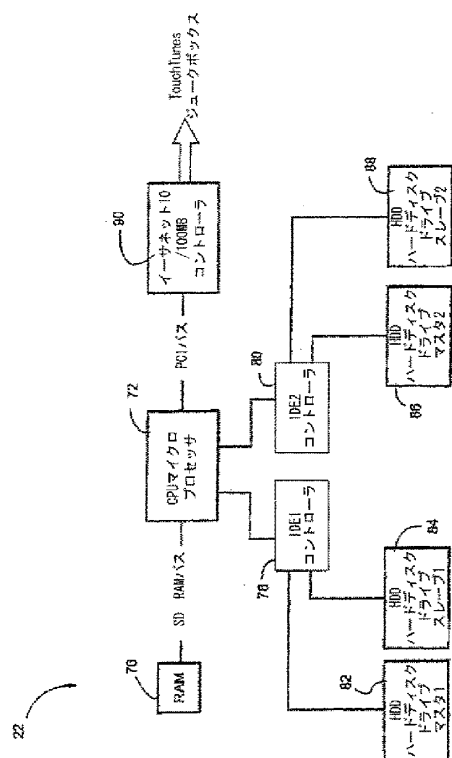
【図5A】



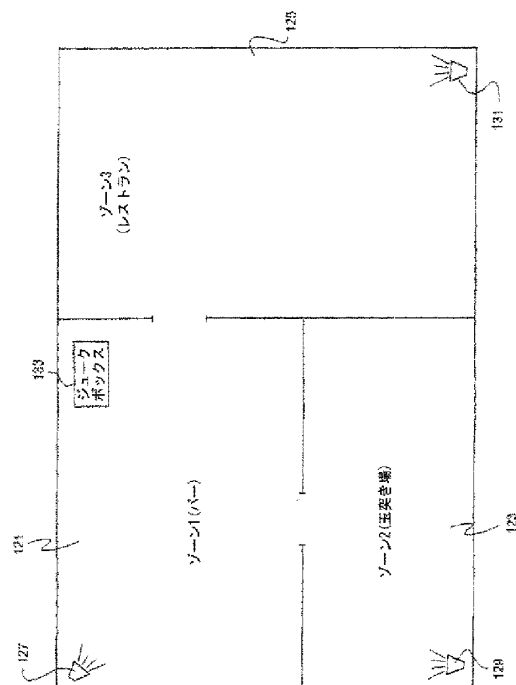
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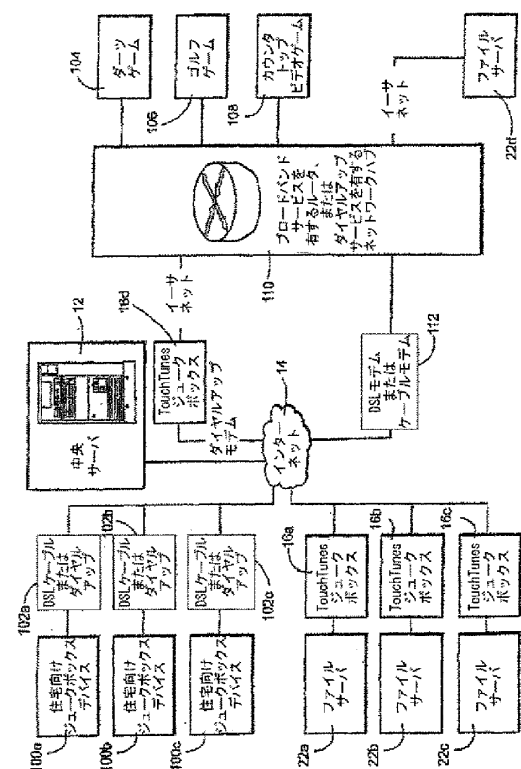
【例7】



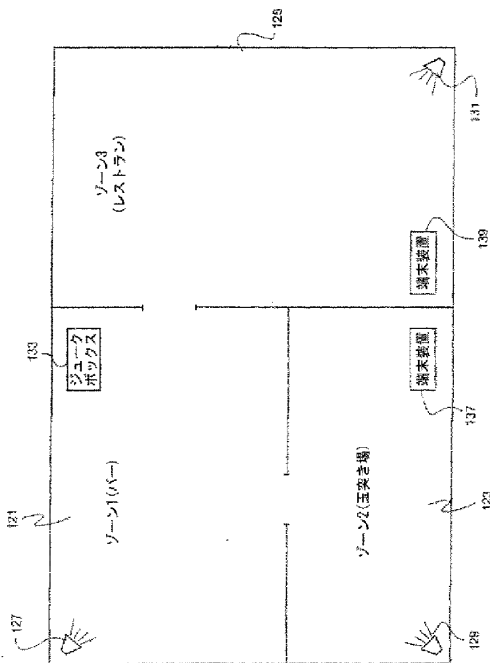
【例9】



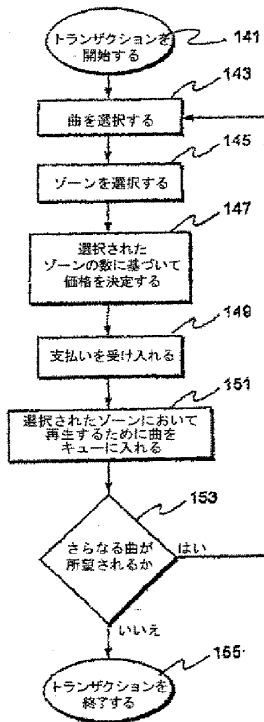
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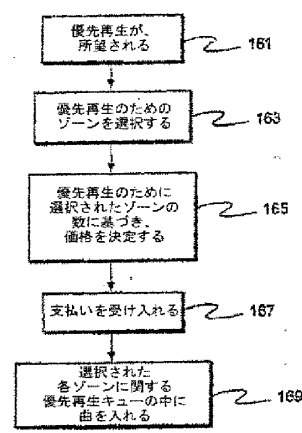
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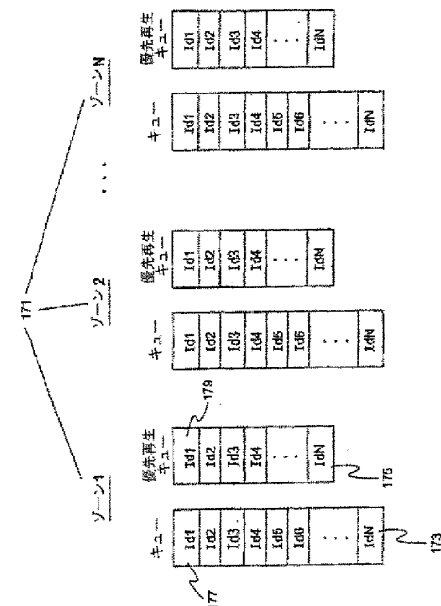
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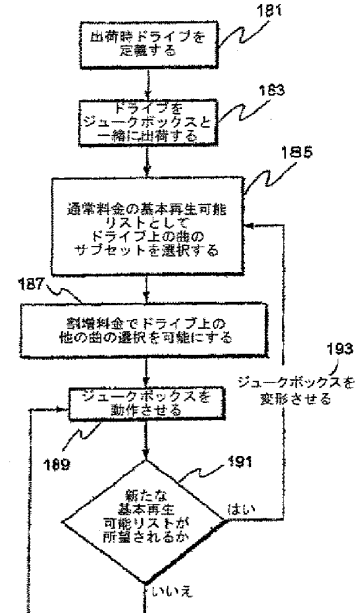
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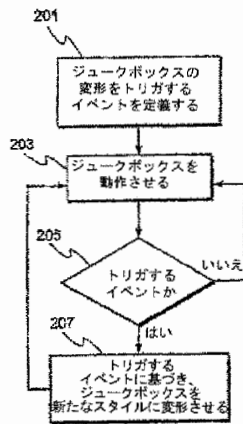
【図13】



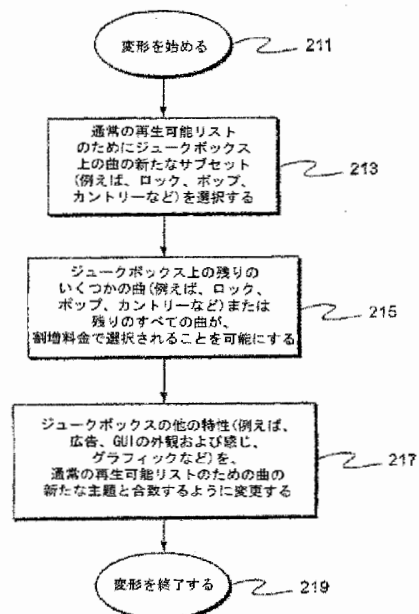
【図14】



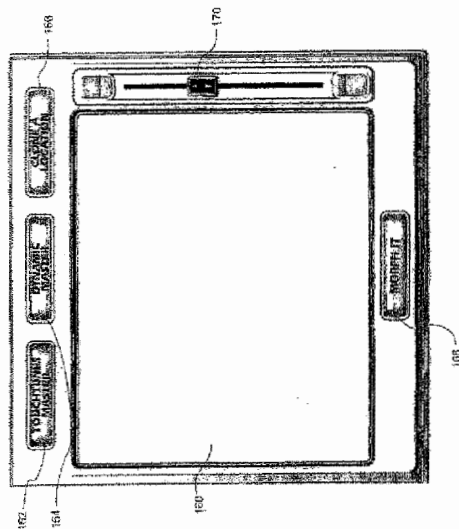
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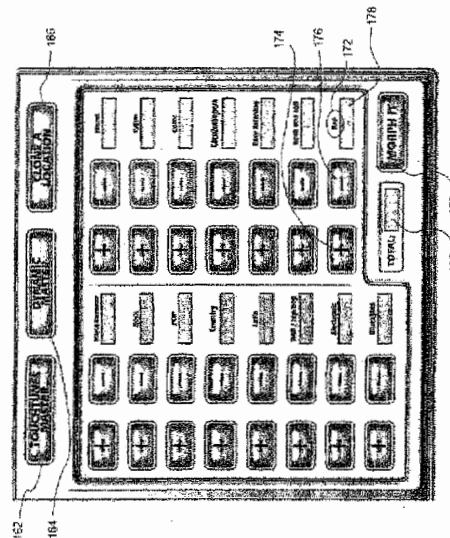
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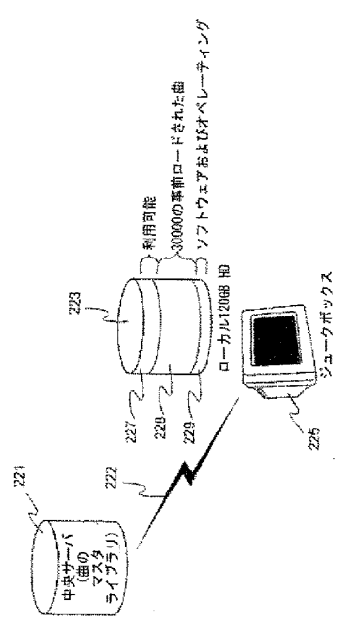
【図16A】



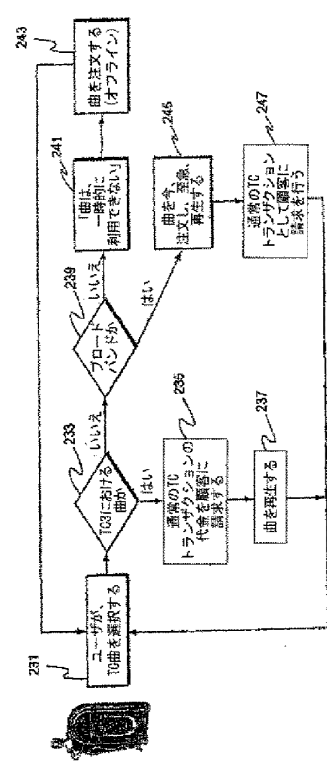
【図16B】



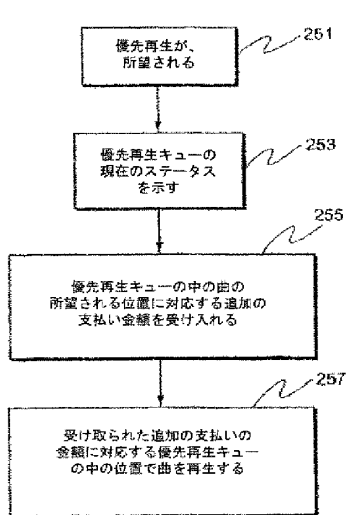
【図17】



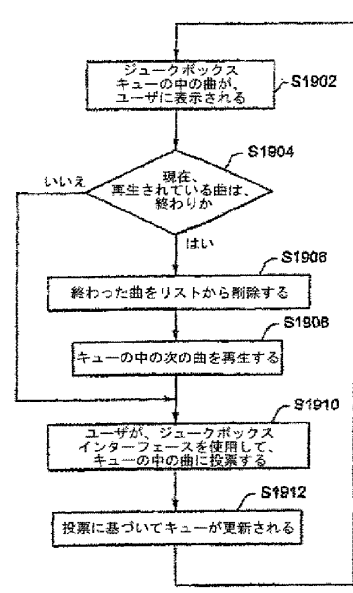
【図18】



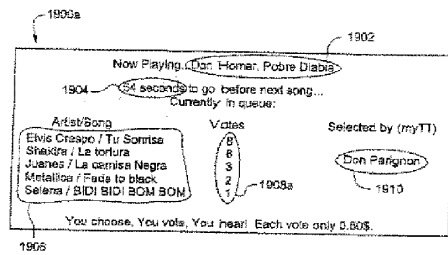
【図19】



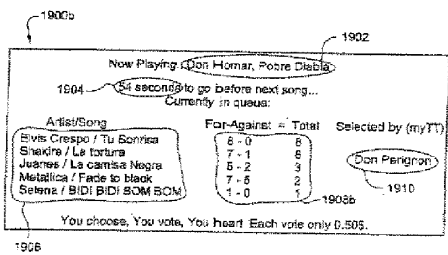
【図19A】



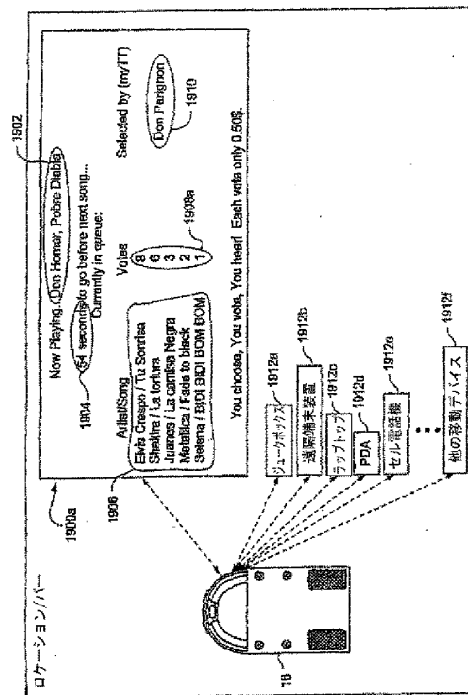
【図19B(1)】



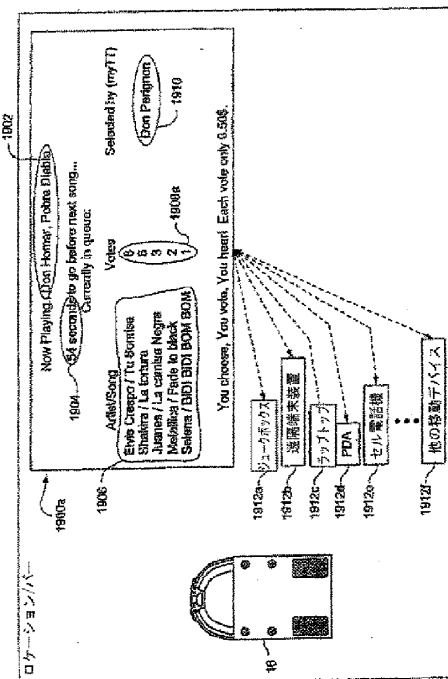
【図19B(2)】



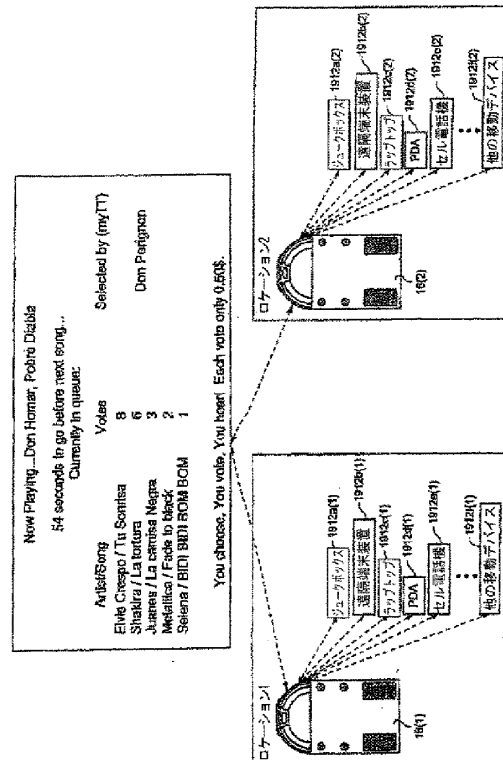
【図19C】



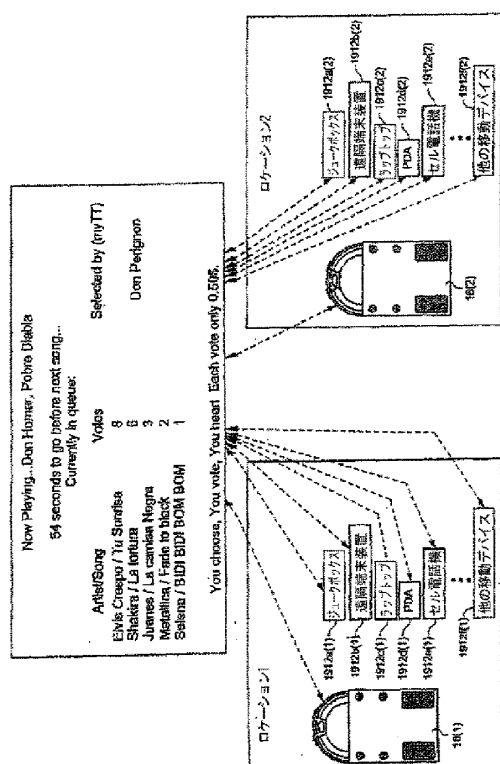
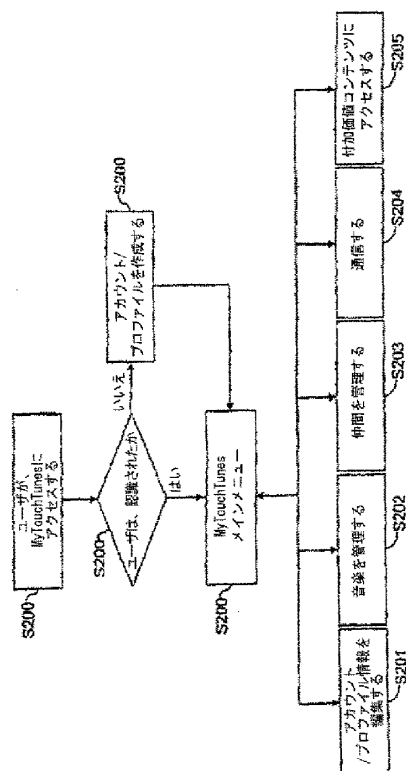
【図19D】



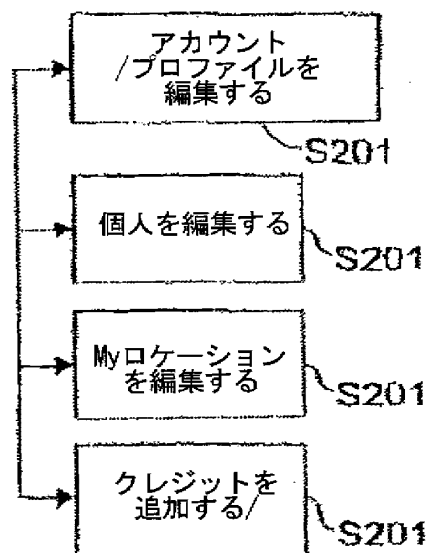
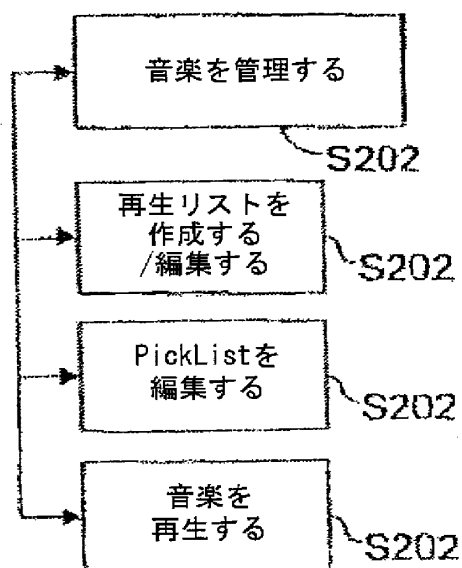
【図19E】



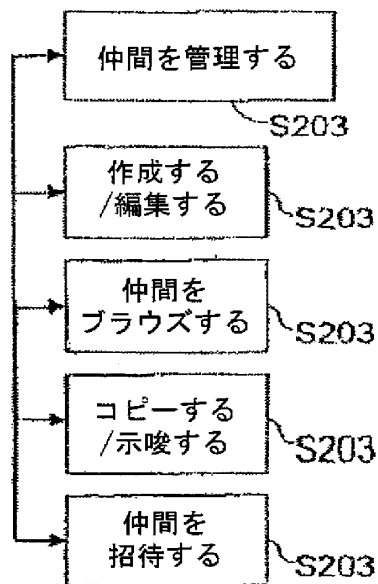
【图20】



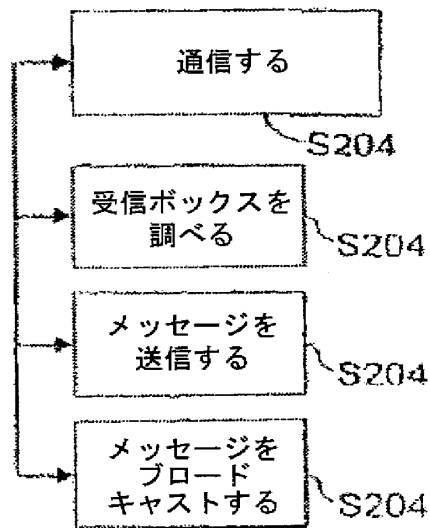
【☒21B】



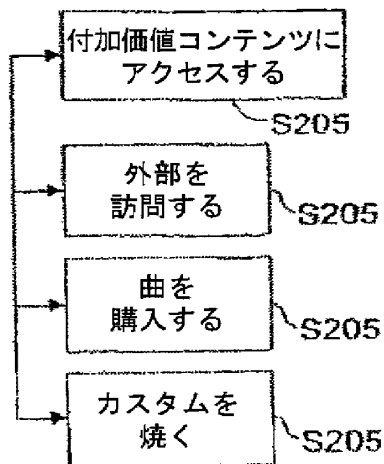
【図21C】



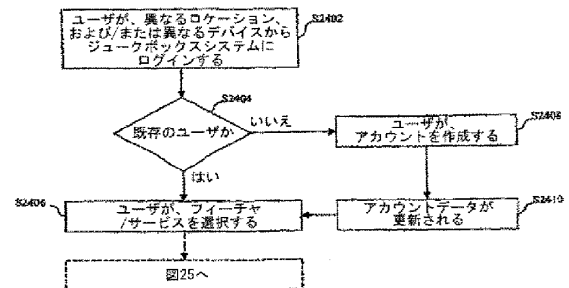
【図21D】



【図21E】



【図24】



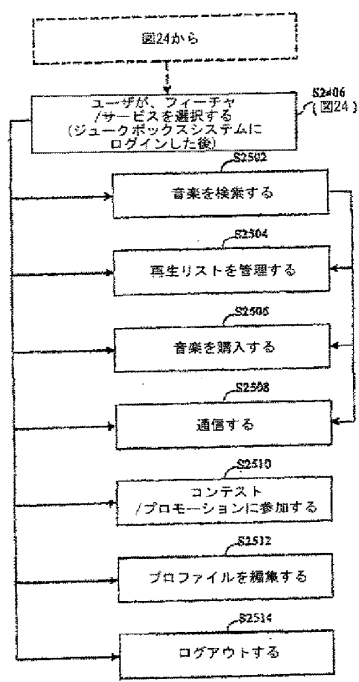
【図22】

この図は公序良俗違反のため不掲載とする

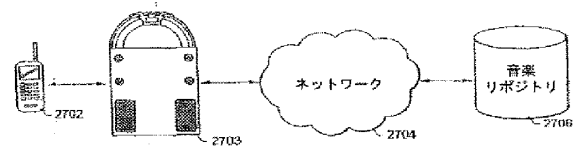
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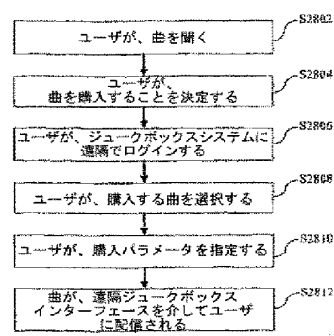
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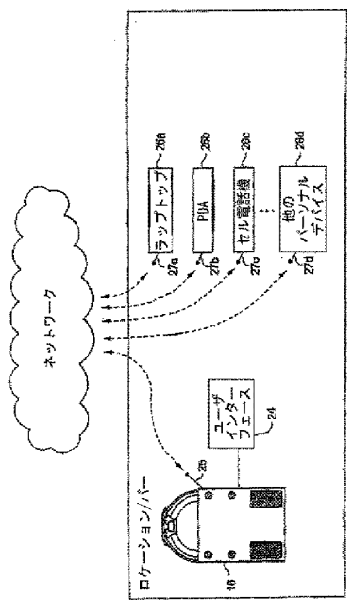
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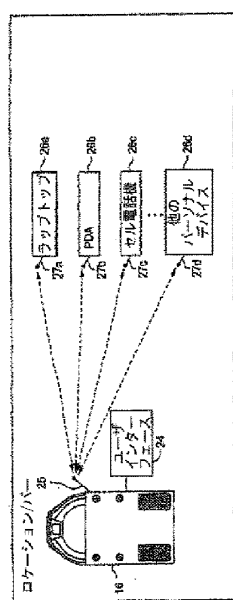
【図28】



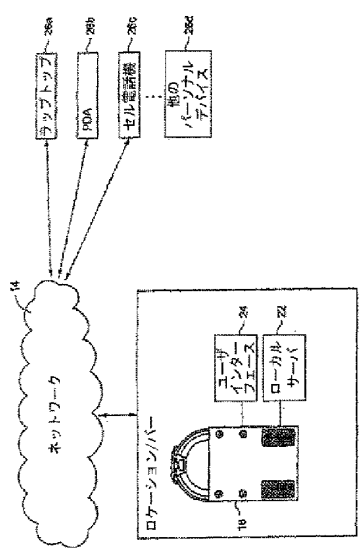
【図29】



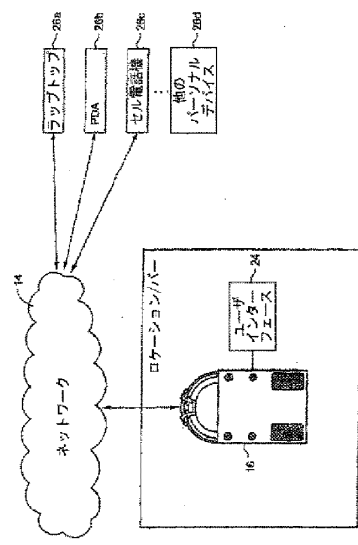
【図29A】



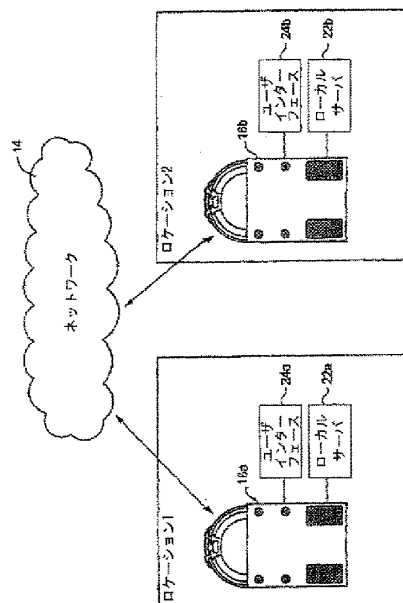
【図29B】



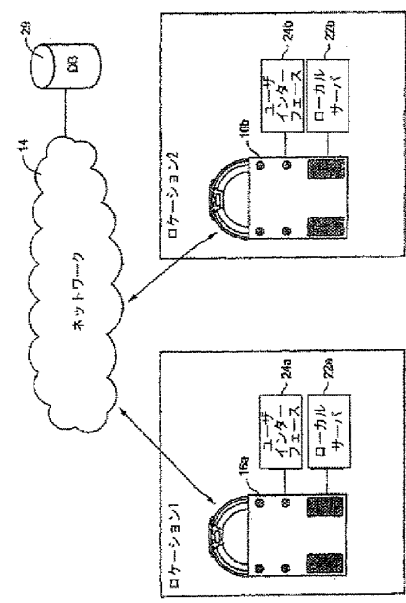
【図29C】



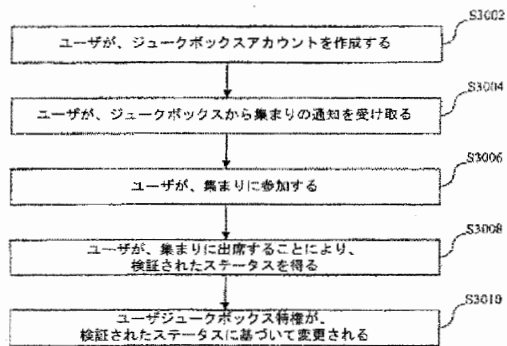
【図29D】



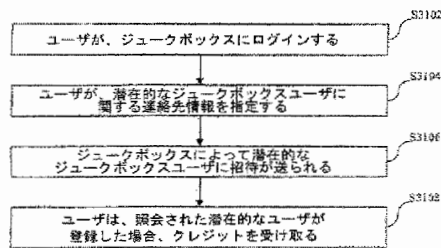
【図29E】



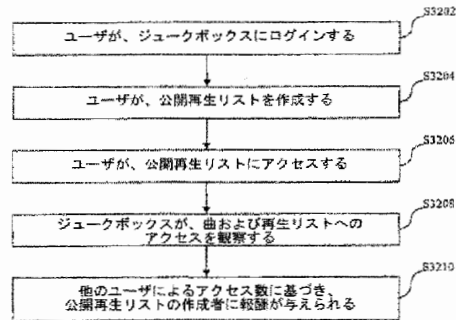
【図30】



【図31】



【図32】



【図26A】

この図は公序良俗違反のため不掲載とする

【図26B】

この図は公序良俗違反のため不掲載とする

【図26C】

この図は公序良俗違反のため不掲載とする

【図26D】

この図は公序良俗違反のため不掲載とする

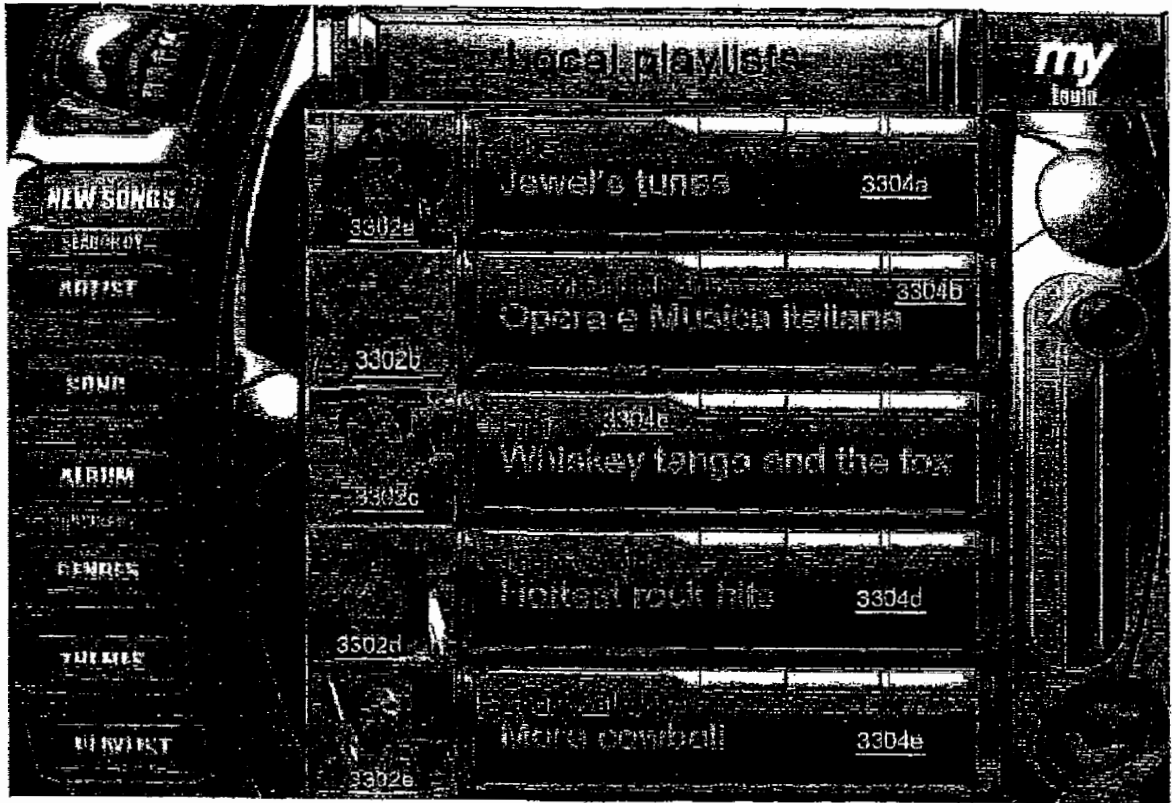
【図26E】

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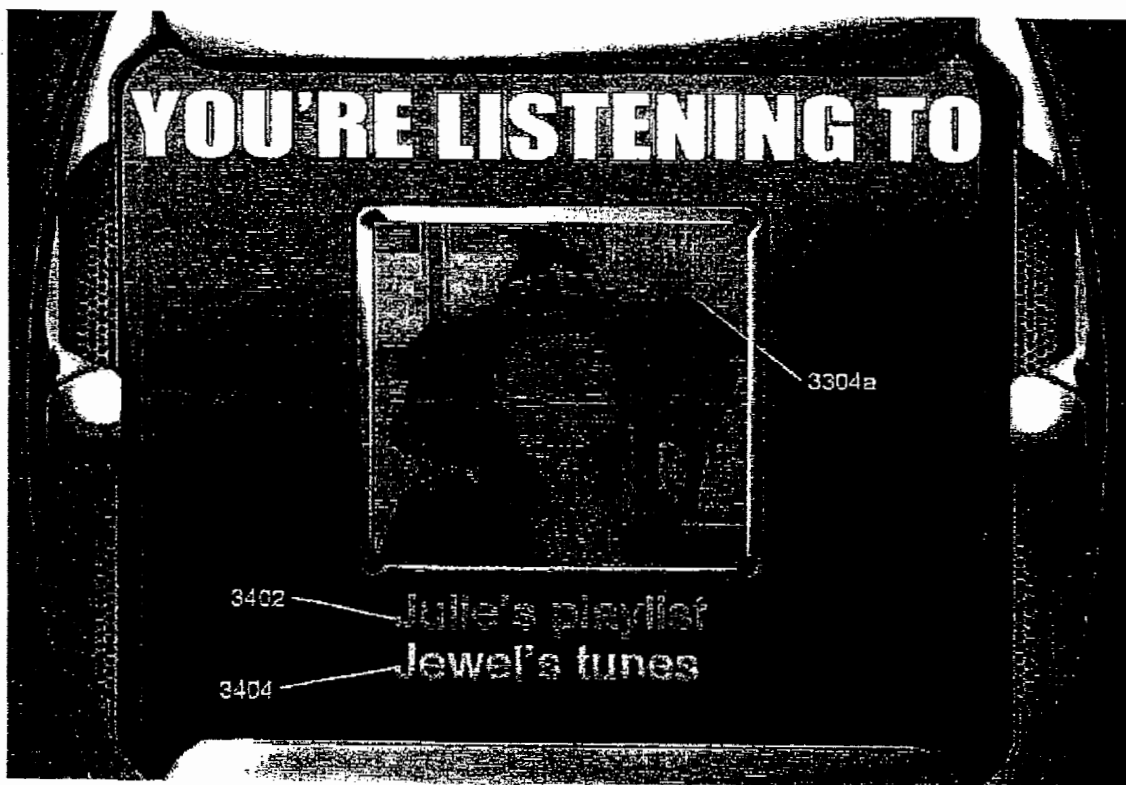
【図26F】

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【図33】



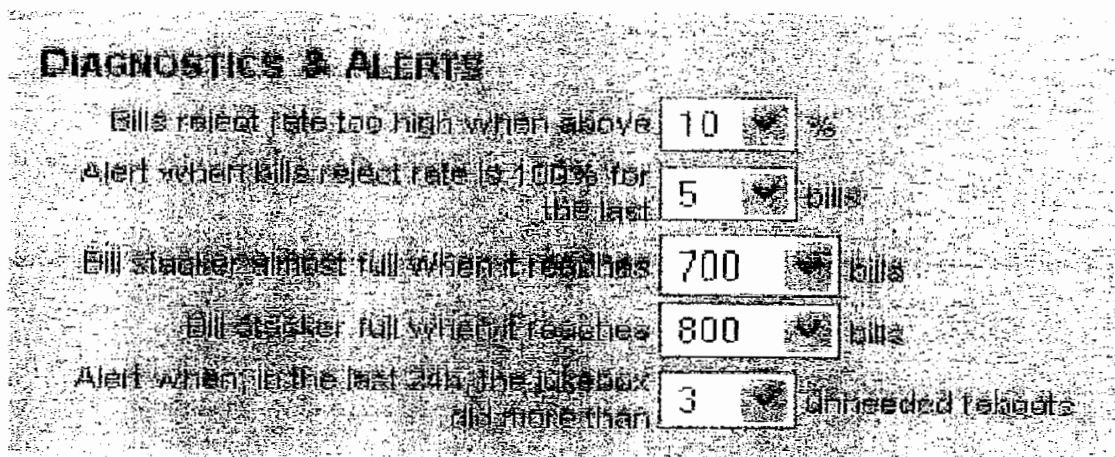
【図34】



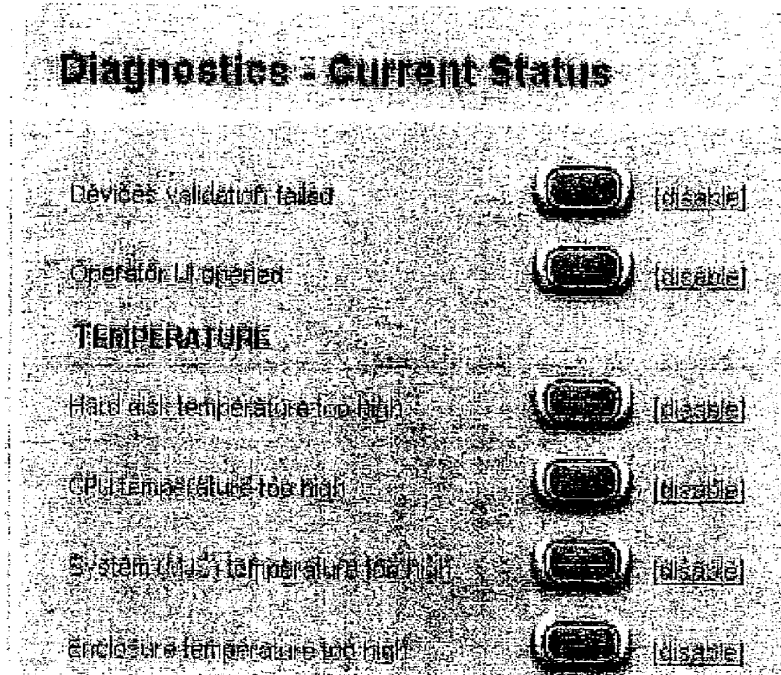
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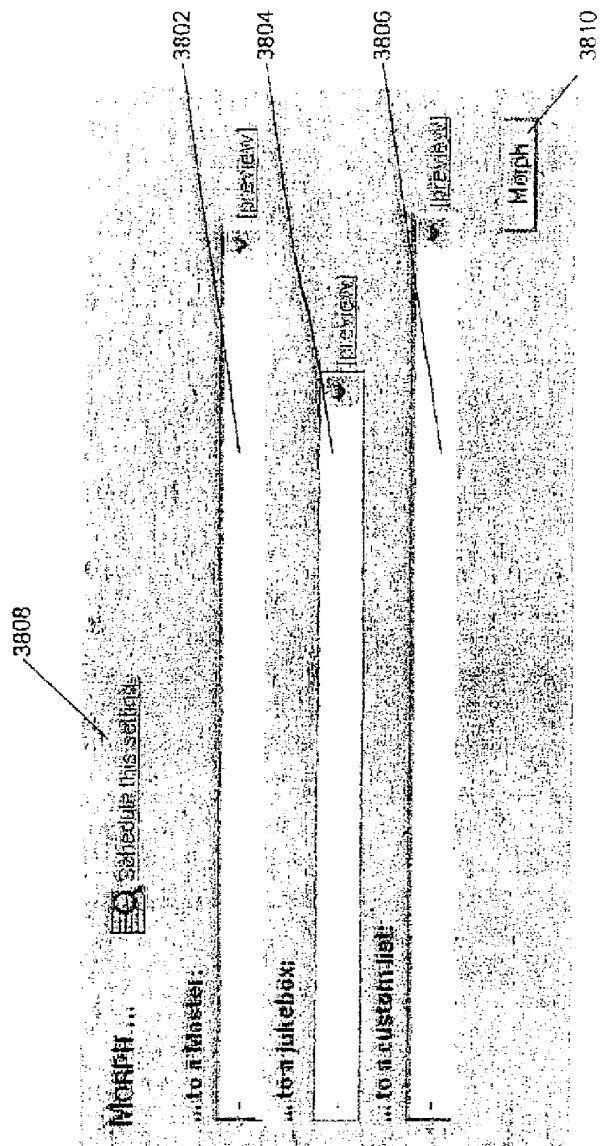
【図36】



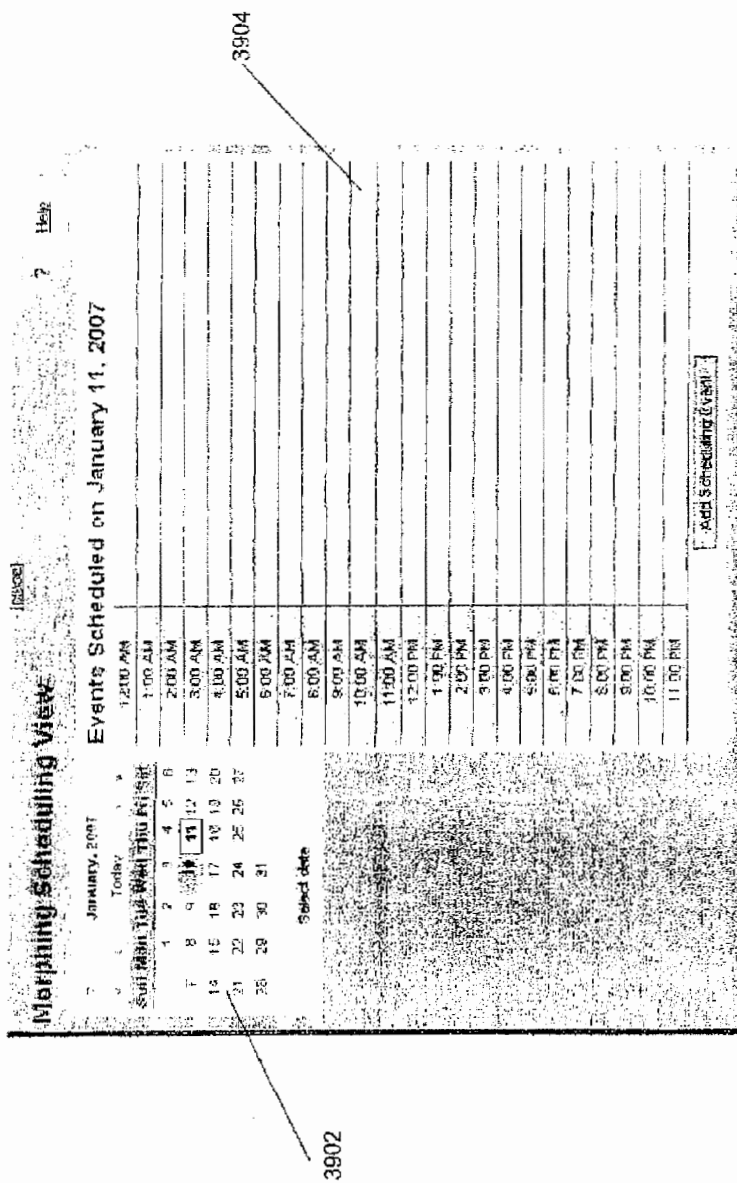
【図37】



【図38】



【図39】



- (72)発明者 ドミニク・ディオ
カナダ・ケベック・H3E・1P1・パードゥン・ナンズ・アイランド・ウィリアム・ポール・1
75・#702
- (72)発明者 ガイ・ネイサン
カナダ・ケベック・H3E・1T4・パードゥン・ナンズ・アイランド・シュマン・デュ・クラブ
・マリン・201・#1903
- (72)発明者 フランソワ・ブリオン
カナダ・ケベック・H7X・4C3・ラヴァル・ギラルドー・302
- (72)発明者 ロイク・グラットン
カナダ・ケベック・H3Z・1W9・ウェストマウント・サン・タントワーヌ・ケス・3139
- Fターム(参考) 5D108 CA04 CA07 CA15 CA29

【外国語明細書】

DIGITAL DOWNLOADING JUKEBOX WITH ENHANCED COMMUNICATION FEATURES

CROSS-REFERENCES TO RELATED APPLICATIONS

This application claims the benefit of application Serial No. U.S. 11/902,658, filed on September 24, entitled "Digital Downloading Jukebox with Enhanced Communication Features".

FIELD

The instant invention relates to, for example, jukebox systems and, more particularly, to digital downloading jukebox systems of the type which typically include a central server and remote jukebox devices that communicate with the central server for royalty accounting and/or content updates and, still more particularly, to jukebox systems that provide user-tailored music management, communications, and other tools for recognized users using the jukebox system. Exemplary embodiments of the instant invention improve such systems by providing a local server for each jukebox device in the jukebox system network. The local server provides a second and more expansive source of content (e.g., audio and/or visual data) that can be selected by a user of the jukebox device for reproduction on the jukebox device. The local servers may provide a mirror of or subset of the central server, thereby enabling the entire library of audio and/or visual data, or at least the most requested content, to be conveniently available to each jukebox device without the need to download requested content, that is not available on the mass storage device of the jukebox device itself, from the central server. The collective group of local servers may also act as a network of distributed content servers that can be controlled by the central server through each jukebox device to provide services to other devices, such as, for example, other jukebox devices. In addition, the jukebox device and local server can, under control of the central server, operate as a "central hub" or management device for various downloadable fee-based devices present in a location with the jukebox device.

Exemplary embodiments additionally can provide other mechanisms to enhance individual recognized users' experiences, such as, for example, by allowing users to create and manage play lists that can be retrieved on one or more jukeboxes, create and manage buddy lists that can send messages to individuals or broadcast to large groups of individuals, etc. In other words, in one embodiment, the jukebox includes features which enable the jukebox to recognize specific jukebox users, through prior registration or the like, and to provide user-specific services to the user. For example, the jukebox may allow the recognized user to create, edit and/or store custom playlists on the jukebox that can be selected by the user or other registered users designated by the creator of the playlist for play on the jukebox. The jukebox may also allow registered users to communicate with other registered users through the jukebox. Registered users may also be provided with credit on the jukebox so that cash is not necessary for using the jukebox services. The jukebox may also be used to provide background music services in addition to conventional jukebox services in a unique and advantageous manner as described herein. The jukebox may additionally increase a sense of ownership by interacting with an interactive application accessible at a location remote from the jukebox, such as, for example, via the Internet. The remote application may connect individuals to an online community through which users can, for example, communicate with other users, build playlists, share playlists, suggest music to others, participate in promotions, link to targeted advertisements, etc. Additionally, the jukebox and/or remote application may allow users to download information to a portable device, such as, for example, a cell phone, a personal digital assistant, an MP3 player, etc. In certain exemplary embodiments, users' actions on jukeboxes and/or via remote interfaces may be represented by avatars specific to and/or customizable by the user.

BACKGROUND AND SUMMARY

Jukeboxes have been around for decades and provide users with the ability to select desired music for reproduction in a convenient and advantageous manner. Jukeboxes have conventionally been provided in commercial establishments, such as restaurants and bars, in order to provide desired music on demand for patrons thereof

for a fee. Over the last several years, a new generation of jukebox devices have become available that provide significant improvements in the operation thereof for all parties involved. More specifically, the conventional standalone phonorecord and CD jukeboxes are being replaced by digital downloading jukeboxes that are controlled by and communicate with a central server. An example of this new generation jukebox system is shown in U.S. Patent No. 6,308,204, the entire disclosure of which is incorporated herein by reference. A leading provider of this new generation of jukebox systems is TouchTunes Music Corporation.

Figure 1 shows an overview of an exemplary embodiment of a digital downloading jukebox system 10 (hereinafter referred to simply as a "jukebox system"). As shown in Figure 1, the jukebox system 10 includes a central server 12 that contains a master library of audio content (typically music), as well as or alternatively audiovisual content (typically music and associated video or graphics), that can be downloaded therefrom. The jukebox system also includes a series of remote jukebox devices 16, 16a-16f. Each of these jukebox devices are generally located in a bar, restaurant, club or other desired location, and are operable to play music in response to receiving a payment from a user, such as coins, bills, credit/debit card, etc., and having one or more songs selected by the user for play. In an alternative embodiment, a music service is paid for on a subscription basis by the location, and the selected music is free for the end-user. The jukebox device 16 typically includes a screen 18 that presents information to the user and allows the user to select songs therefrom, as well as an audio system 20 that plays the selected songs. The screen 18 may also be used for displaying song-related video or graphics. The screen 18 may also be used to display advertisements for the jukebox itself in order to attract customers thereto, to display other types of advertisements, and/or to display any other desired information.

The jukebox devices 16 (sometimes referred to as simply "jukeboxes" herein) are operable to communicate with the central server 12 through a communications network 14, such as, for example, the Internet. The jukeboxes 16 periodically communicate with the server 12 in order to provide information to the server 12 regarding the specific songs that have been played on the jukebox. The central

server then uses this information in order to determine the appropriate royalties and/or other payments that are owed for songs played on each jukebox.

Thus, one significant advantage of this new generation of jukeboxes is that the sound reproduction and/or other applicable music rights can be adhered to in a more accurate and reliable manner, thereby assuring the proper royalties are paid to the artists or music owners. The central server 12 can also provide new songs to the jukebox 16 in order to assure that the appropriate or most popular songs are maintained on the jukebox based on the specific customers at that location. Thus, the songs available on each jukebox can be customized through communication with the central server in order to provide the songs and/or types of music that customers generally request at each jukebox location. As described in the above-referenced U.S. Patent No. 6,308,204, the central server can also advantageously be used to update the operating software on the jukeboxes in order to, for example, change the operation of the jukebox, such as to provide new or improved features. Thus, another significant advantage of this new generation of jukeboxes is that the songs (or other audio and/or visual content), and the operation of the jukebox itself can be remotely changed as desired without the need to have someone (such as a routeman) personally service the jukebox. Instead, such updates can be done using the central server 12.

As indicated above, the jukebox devices 16 each include a mass storage device, such as a hard drive, which stores the songs and associated video/graphics data (if any), as well as any other desired graphical information for reproduction on the jukebox. The mass storage device of the jukebox typically has limited storage capacity relative to the storage device of the central server 12. As a result, only a fraction of the songs stored on the central server are typically stored on the mass storage device of the jukebox at any one time. There may be other reasons as well, such as for security of the data or limited room in the jukebox itself, for having limited storage capacity on the jukebox and/or limiting the number of songs stored thereon. For example, physical space may be limited on wall-mount jukeboxes or the like, which are designed to be small in size as compared to free-standing models. As explained above, the songs on the jukebox can be changed through communication

with the central server, but typically any one jukebox only stores a relatively small subset of the complete library of songs maintained by the central server at any one time.

In order to maximize the revenue that a jukebox generates, it is important to make the most desired songs available on the jukebox over time. If customers cannot find songs they like on the jukebox, usage of the jukebox (and the revenue generated thereby) will dramatically decrease. On the other hand, it is impossible to predict in advance exactly what a customer at any particular location will desire to play on the jukebox. In fact, there are likely many instances where a customer would have selected a song that exists on the central server but is not currently present on the jukebox. As a result, the jukebox may not be enjoyed, and used to its fullest extent. In order to address this problem and increase revenue, jukebox systems have in the past provided a feature which enables the user to search for songs on the central server from the jukebox and request an immediate download of a desired song from the central server to the jukebox for an additional fee. This feature enables the user to play any song in the master library of songs maintained by the central server using the jukebox, regardless of whether or not the specific song is presently stored in the mass storage of the jukebox itself. Thus, the user can first look for desired, songs on the local storage of the jukebox and then, if desired, search further on the central server for desired songs. The jukebox device typically charges an additional fee (such as five credits instead on one credit) for an immediate download and play of a song from the central server as opposed to a standard play directly from the jukebox's local storage.

One problem, however, with the immediate downloading feature is that it is desirable to have an immediate and high speed connection with the central server to implement. In addition, the central server and network must be prepared for and capable of handling such requests in a reliable and efficient manner for the feature to properly operate. These requirements cannot always be met and, as a result, implementation of this feature has been limited. For example, many locations that have jukeboxes do not have high speed connections (such as DSL) and instead use dial-up modem connections. Jukeboxes which rely on dial-up connections generally

are only designed to communicate with the server periodically and do not allow the user to immediately download a song. They have, however, enabled a user to vote for a song to be downloaded at a later time when the dial-up connection is made. This, of course, is not as satisfying to the user as being able to immediately download a song. Other problems can arise in connection with this download feature if the network or server is not currently available for the download, due to traffic, malfunctions or the like.

For most users, a relationship with a jukebox extends only as far as a choice of songs to play. Casual users may find it difficult to make selections rapidly and/or to locate specific songs. Regular users, however, frequently choose the same songs, and they may become frustrated with always having to make the same selections. Both problems may result in a loss of revenue from, intimidation, frustration, etc.

Thus, it is a feature of the exemplary embodiments to create a relationship between the jukebox and the patrons, in view of the further feature of creating a relationship with a whole community of jukebox users. Such a system can establish a trust relationship between the jukebox and the patrons, while also creating a sense of ownership for the patrons by allowing them to customize their services and communicate with their friends. Such a relationship and feeling of ownership can greatly increase the convenience of using a jukebox and its related services while also decreasing intimidation, frustration, etc.

In certain exemplary embodiments described herein, the jukebox can be made the centerpiece for music services as well as other services that enhance the experience of users as well as the revenues of location managers. For example, a jukebox can become a contact point at a given location through which users can reach out to their friends. For instance, users may call a get together using the jukebox, share music through the jukebox system, or meet up at a jukebox if they become separated at a crowded club. Similarly, a jukebox can become, for example, a contact point at a given location through which the manager of the location can reach out to customers. For instance, a manager can advertise specials, notify regulars of special events, etc. Such communications can be initiated by users, by location managers, or by the jukebox itself.

When jukebox users leave a location, their connection to the music, and, more particularly, the jukebox itself, typically ends. Users provide massive amounts of data to the jukeboxes, such as, for example, which users like certain types of music or certain songs, when they listen to music, how frequently they listen to it, etc. With conventional jukeboxes, this information is not leveraged to provide additional features and/or services to interested users. Thus, collected information typically is wasted, and users experience only a fleeting sense of ownership. Users cannot, for example, recreate the experience of a particular night out.

Proprietors also lose out, because their connection to jukebox users disappears when users leave their establishments. Thus, proprietors lose customers until they return and are forced to rely on traditional, often ignored, methods of bulk advertisement. This form of advertising typically lacks specificity and does not reflect a personalized sense of belonging for users. Additionally, proprietors also cannot recreate specific events at which, for example, they had a particularly profitable nights.

Online communities typically, for example, on the Internet, have sprung-up to try to connect disparate users in a virtual space. These communities provide potentially continuous access to a broad range of features and users alike. However, conventional online communities are virtual only - when a user steps away from a computer, the connection, quite literally, is broken. Thus, while users may access this broad, host of features and customize their experiences and even build an online persona complete with, for example, buying and viewing habits, the experience is fleeting.

By connecting jukebox users through an online community, however, the information provided to the jukebox can be used to provide a variety of features and services to the users. And, because jukeboxes are present at thousands of meeting points, connections exist at thousands of physical locations apart from the online virtual community. Thus, by leveraging the data collected by jukeboxes and tying users to an online community, a mixed real-and-virtual community may be established, for example, creating a lasting sense of ownership, personalizing services for potentially all users, drawing users together, establishing a vibrant socially active

community of users connected by music, etc. Proprietors may similarly benefit by, for example, achieving a pipeline to additional business by connecting with customers while they are away from their locations, thus drawing them to their locations, etc.

In essence, a jukebox with an associated remote application can, for example, change users' perceptions of jukeboxes. Jukeboxes need not be stand-alone devices accessible only at bars. Rather, the instant invention allows users to consider jukeboxes as, for example, media centers, meeting points, portals to online communities, etc.

Moreover, with the advent of web-enabled portable devices (such as, for example, cell phones, personal digital assistants, etc), users even can take the virtual community with them wherever they go. Users potentially may use portable devices to directly download music at any time, such as, for example, when they hear a song at home, in the car, at a bar, etc.

In certain exemplary embodiments, users' actions on jukeboxes and/or via remote interfaces may be represented by avatars specific to and/or customizable by the user. Jukeboxes may include avatar action programmed logic circuitry (e.g., any appropriate combination of hardware, software, or the like) to take actions on behalf of and/or represent the actions of the user. Such actions may include introducing songs with audio and/or video, singing and/or dancing along with music, marking online transactions (e.g., messages, postings, file transfers, etc.), etc. As such, user avatars may yet further increase senses of ownership and/or belonging, drawing users to jukeboxes. Moreover, avatars may further increase the enjoyment of the individual user, regular patrons, and/or newcomers to an area

There is a need for an improved jukebox system that addresses various needs of users, location owners, and operators. The instant invention is designed to address these and other issues and to provide enhanced functionality for such jukebox systems.

BRIEF DESCRIPTION OF THE DRAWINGS

These and other features, objects and advantages of the instant invention will be further understood by review of the following detailed description of the exemplary embodiments when read in conjunction with the appended drawings, in which:

- Figure 1 is a block diagram of a conventional downloading digital jukebox system;
- Figure 2 is a block diagram of the improved downloading digital jukebox system in accordance with a preferred embodiment;
- Figure 3 is an exemplary screen shot showing an initial selection screen in accordance with a preferred embodiment of the jukebox system;
- Figure 4 is another screen shot showing an exemplary search, screen for use in searching for songs on the local server in accordance with a preferred embodiment;
- Figure 4A shows an exemplary process for using a Personal Music Assistant to search for songs that might be appropriate for a user-specified profile;
- Figure 4B shows an exemplary process for using a Personal Music Assistant to search for songs that might be appropriate for a recognized user's profile;
- Figure 4C is an exemplary screen shot showing a non-limiting list of genres and themes available to a user in accordance with one illustrative embodiment;
- Figure 5 is another exemplary screen shot showing the results of a search on the local server and providing the user an option of downloading a desired song to the jukebox device for a fee, in accordance with a preferred embodiment;
- Figure 5A shows an exemplary process for searching through a list of popular songs;
- Figure 6 is another exemplary screen shot showing an alternative method of allowing access to the downloading feature;

- Figure 7 shows a block diagram of a preferred embodiment of the local server;
- Figure 8 shows a block diagram of an exemplary overall network including commercial jukeboxes and residential jukeboxes, as well as other downloading devices and associated connections that are managed by the jukebox system;
- Figure 9 shows an overhead view of an exemplary establishment layout for a multi-zone jukebox system;
- Figure 10 shows an overhead view of an exemplary establishment layout for a multi-zone jukebox system with selection terminals in each zone;
- Figure 11 is a flowchart showing an exemplary implementation of a zone selection process for a multi-zone jukebox system;
- Figure 12 is a flowchart showing an exemplary implementation of a priority play by zone selection process for a multi-zone jukebox system;
- Figure 13 shows an exemplary- implementation of a multi-zone set of priority and non-priority queues, with a subset of queues for each zone;
- Figure 14 is a flowchart showing an exemplary distribution and initialization scheme for a jukebox with morph capability;
- Figure 15 is a flowchart showing an exemplary implementation of an automatic jukebox morph initiation process based on a triggering event;
- Figure 16 is a flowchart showing an exemplary implementation of a jukebox morphing process;
- Figure 16A is an exemplary screen shot showing how a morph of a jukebox can be initiated in one preferred embodiment;
- Figure 16B is another exemplary screen shot showing how a user can specify various criteria before a morph of a jukebox in one preferred embodiment;
- Figure 17 shows the relationship between a jukebox with expanded media storage and a central server;

- Figure 18 is a flowchart showing an exemplary process for a song selection process when a song is not in the "standard" available playable song list;
- Figure 19 is a flowchart showing an exemplary process for a priority play queue with prioritization-based-on-bidding capability;
- Figure 19A is a flowchart showing an exemplary implementation of a jukebox voting process;
- Figure 19B(1) is an exemplary screenshot showing a voting queue display;
- Figure 19B(2) is another exemplary screenshot showing a voting queue display;
- Figure 19C is an exemplary block diagram showing a voting mechanism within a given location;
- Figure 19D is another exemplary block diagram showing a voting mechanism within a given location;
- Figure 19E is an exemplary block diagram showing a voting mechanism for use with multiple locations;
- Figure 19F is another exemplary block diagram showing a voting mechanism for use with multiple locations;
- Figure 20 is a flowchart showing an illustrative process for using a user-tailored system for managing music, communications, etc. in accordance with one exemplary embodiment;
- Figure 21A shows non-limiting, illustrative options available for editing account/profile information in one exemplary embodiment;
- Figure 21B shows non-limiting, illustrative options available for managing music in one exemplary embodiment;
- Figure 21C shows non-limiting, illustrative options available for managing buddy lists in one exemplary embodiment;
- Figure 21D shows non-limiting, illustrative options available for communicating with others in one exemplary embodiment;
- Figure 21E shows non-limiting, illustrative options available for accessing value-added content in one exemplary embodiment;

- Figure 22 is an exemplary screen shot of a login page in accordance with one exemplary embodiment;
- Figure 23 is an exemplary screen shot showing the features available after a recognized user has logged in, in accordance with one exemplary embodiment;
- Figure 24 is a flowchart showing an exemplary process for accessing a system remotely in one exemplary embodiment;
- Figure 25 is a flowchart showing an exemplary process for selecting one or more of a non-limiting list of services and/or features in a system according to one exemplary embodiment;
- Figure 26A is an exemplary screen shot showing the features available before a user logs into a system according to one exemplary embodiment;
- Figure 26B is an exemplary screen shot showing the features available after a user logs into a system according to one exemplary embodiment;
- Figure 26C is an exemplary screen shot showing how a user might update account information;
- Figure 26D is an exemplary screen shot showing how a user might manage playlists;
- Figure 26E is an exemplary screen shot showing the results of an illustrative search for music;
- Figure 27 shows an exemplary system for downloading music to a mobile device;
- Figure 28 is a flowchart showing an exemplary process for retrieving music;
- Figure 29 is a block diagram of an improved downloading digital jukebox system with remote devices accessing a jukebox from within a particular location;

- Figure 29A is a block diagram of an improved downloading digital jukebox system with remote devices accessing a jukebox from within a particular location;
- Figure 29B is a block diagram of an improved downloading digital jukebox system with remote devices outside of a particular location accessing the digital jukebox system;
- Figure 29C is another block diagram of an improved downloading digital jukebox system with remote devices outside of a particular location accessing the digital jukebox system;
- Figure 29D is a block diagram of an improved downloading digital jukebox system with another downloading digital jukebox system outside of a particular location accessing the first digital jukebox system;
- Figure 29E is another block diagram of an improved downloading digital jukebox system with another downloading digital jukebox system outside of a particular location accessing the first digital jukebox system;
- Figure 30 is an exemplary flowchart showing one way in which users may change their status by becoming "verified users";
- Figure 31 is an exemplary flowchart showing how registered users may receive credits for referring potential users;
- Figure 32 is an exemplary flowchart showing how registered uses may receive credits for creating public playlists;
- Figure 33 is an exemplary screenshot of a "Local Playlists" screen in which registered users are represented by avatars;
- Figure 34 is an exemplary screenshot of a "now playing" screen;
- Figure 35 is an exemplary screenshot for setting alert preferences;
- Figure 36 is an exemplary screenshot for setting alert thresholds;
- Figure 37 is an exemplary screenshot showing the status of various tests;
- Figure 38 is an exemplary screenshot of the morphing capabilities being linked to the enhanced scheduling module; and,
- Figure 39 is an exemplary screenshot of a scheduling calendar.

DETAILED DESCRIPTION OF EXEMPLARY EMBODIMENTS

Referring now to the drawings, Figure 2 shows a block diagram of an exemplary preferred embodiment of an improved jukebox system 10'. The jukebox system 10' includes similar elements as shown in Figure 1 and described above, including a central server 12, communications network 14, and remote jukebox devices 16, 16a-16f. However, the jukebox system 10' further includes local servers 22, 22a-22f respectively connected to each of the jukebox devices 16, 16a-16f. The central server 12 includes a master library of songs (and/or other content). Each of the jukebox devices includes a subset of the master library on a local storage device of the jukebox. The central server may be used to individually manage the contents of the jukebox device, by monitoring usage of and updating the subset of songs on each of the jukebox devices with the intent of maximizing the usage thereof. The central server 12 periodically receives data from each of the jukeboxes for the purpose of royalty accounting and payment for songs played. The jukebox devices may connect to the network in any suitable manner, such as dial-up modem or broadband modem (e.g., DSL, cable, wireless broadband, or satellite). The communications network 14 may be any suitable network capable of distributing data (e.g., audiovisual data) from the central server 12 to the jukeboxes 16 and enabling data to be uploaded from the jukeboxes 16 to the central server 12.

The songs (and/or other data) are preferably digitized, compressed and encrypted by the central server 12 prior to sending songs to the jukeboxes for security and bandwidth purposes using known techniques. The songs are then decompressed and decrypted by the jukeboxes for storage and reproduction thereon. Thus, each of the jukeboxes maintains in a database a library of digitized songs for play on the jukebox, wherein the library can be changed or updated through communication by the central server. The jukeboxes preferably also receive and store data constituting images (e.g., still and/or moving video and/or graphical images) that can be displayed on the display 18 of the jukebox device 16. In one exemplary embodiment of the invention, the jukebox devices have similar structure and operation described in U.S. Patent No. 6,308,204 referenced above. Thus, the

jukebox devices 16 each preferably include one or more microprocessors, such as a main CPU and an audio DSP, a memory, such as a hard drive, for storing songs and/or other content, a display of displaying visual items, an audio arrangement 20 for providing audio, a communication system for enabling the jukebox to communicate with the central server 12 through the communications network 14, and operating software, preferably including a multitasking operating system, that controls the operation of the jukebox. The operating software is also preferably updateable through communication with the central server 12 as described, for example, in U.S. Patent No. 6,308,204 referenced above. The jukeboxes 16 further include one or more payment devices, such as coin, bill and/or credit card input devices, for enabling a customer to pay for usage of the jukebox device in a convenient manner. The screen 18 is preferably a touch screen that enables the user to input selections by touching the screen,

Each jukebox device has, in one embodiment, a local server 22 that can be accessed by the jukebox device. The local servers are respectively connected to the jukebox devices using Ethernet or other type of local connection. In another embodiment, the local server may simply be a logical extension (e.g. partition, directory, or area) of the jukebox's hard drive, rather than a separate hardware device. The local servers 22 each preferably include a mirror copy of the master library of musical recordings maintained by the central server 12. The local server 22 can be loaded with the master library by the entity that owns and/or controls the jukebox network prior to shipping the local server and jukebox device to the jukebox distributor or operator. Of course, over time, the local sever will no longer correspond identically to the central server, due to the fact that the central server is preferably continually updated with additional or new songs. Thus, the local servers 22 are also preferably updated periodically to maintain a correspondence with the library on the central server 12. This updating can be done, for example, by the central server 12 through communication with the jukebox devices connected with the local servers 22 using, for example, either dial-up or broadband modems. Alternatively, the updating can be done personally with an update tool that can be connected by a routeman or other person directly to the jukebox or local server for the

purpose of updating the contents of the local server. The portable tool could include a removable storage medium, such as a hard drive, that could be returned to and reused by the owner of the jukebox system for future updates. The tool itself could be kept by the operator or other person in charge of maintaining specific jukeboxes for use upon receipt of the updated removable storage medium from the owner of the jukebox system.

For security reasons, the local server 22 preferably does not include all of the digital data that constitutes any one song that is stored on the local server 22. In addition, the part of the song that is on the local server is encrypted. The jukebox device 16 contains the missing part of each of the songs on the local server, thereby enabling the jukebox to assemble the complete song based on the contents of the local server and the memory on the jukebox device. The missing data located on the jukebox is needed in order to decrypt the songs. For example, a single block (or other small fraction) of data for each song may be missing on the local server but present on the jukebox device, and the encryption may be based on the missing block and may proceed on a block by block basis. Thus, none of the blocks can be decrypted without obtaining and/or decrypting a preceding block. This feature provides significant security and prevents or deters theft or other type of unauthorized use or copying of the songs on the local server. Thus, in this embodiment, each local server must be specifically assigned to a specific jukebox device so that the decryption described above can be properly performed.

In accordance with a preferred exemplary embodiment, the local servers may also each be individually registered with and identified to the central server 12, so that the central server can individually manage and monitor each local server. The same is true for the jukebox device itself, i.e., it may also preferably be registered with the central server so that it too can be individually monitored and managed by the central server. As will be understood from the foregoing description, the local servers become an important and advantageous part of the jukebox system by allowing the contents thereof to be accessed by the jukebox device to provide additional services (such as providing additional songs) not available on the jukebox device itself. As will be explained below, the song library of the central server and/or

the storage capacity itself can be advantageously used to provide services to other jukeboxes, such as fee-based residential and commercial jukeboxes and/or other fee-based equipment. One preferred use of the local servers may be to provide an immediate song downloading feature for the jukebox device will now be described below in detail with reference to the exemplary screen shots of Figures 3-6.

Figure 3 shows an exemplary screen shot for a music selection screen 30 as displayed on the touch display of the jukebox device. As can be seen in Figure 3, this selection screen, which is preferably the initial selection screen displayed to a customer, includes graphical representations 32 of the various album covers for songs that are stored in the memory of the jukebox device. The albums covers are shown in alphabetical order and the virtual slide bar 33 can be used to scroll through the available albums. Up and down arrows (34 and 35) are also provided for stepping through the available albums. A "Now Playing" button 36 is also provided for showing information on the song currently playing on the jukebox (if any). A "Top Ten" button 38 is also provided for showing a list of the ten most popular songs on the jukebox. A "Tune Central" (TM of TouchTunes Music Corporation) button 39 is also provided, the function of which will now be described in detail with reference to Figure 4.

If the user does not see an album of interest in the display of album covers or desires for any reason to search for available songs that are not present on the jukebox device, the user may select the "Tune Central" button 39. When the "Tune Central" button is pressed, the display on the jukebox is changed from that of Figure 3 to that of Figure 4. The exemplary screen shot of Figure 4 shows a search screen 40 which enables a search to be performed on the local server 22 connected to the jukebox device. This screen 40 provides a virtual keyboard 42 for use in entering a search request. The search can be done by album, artist, song, or genres or themes (i.e. categorized lists of songs, preferably based on popularity, that help a user find a particular song), based on the associated buttons 47. Once a search is typed, the user touches the "Search" button 44 and a search of the contents of the local server is initiated. Input from the virtual keyboard can be cleared using the "Clear" button 46.

Genres may comprise, for example, "pop," "rock," "country," "rap," "R&B," "Blues," etc. Themes may comprise, for example, "Christmas," "Irish," "Birthday," etc. In some exemplary embodiments, these lists can be retrieved from the server, while in other exemplary embodiments the lists may be built dynamically, based on, for example, certain parameters. In still other embodiments, the server may select which lists to enable or send to the jukebox based on criteria, such as, for example, location, music type, etc. A system might only display genres and/or themes for which music is downloaded. However, a system might display all genres and/or themes and enable download of other songs not available to the local jukebox. It also will be appreciated that although FIG. 4C, discussed in detail below, shows the genres and themes on the same screen, other embodiments may present them separately.

FIG. 4C is an exemplary screen shot showing a non-limiting list of genres and themes available to a user in accordance with one illustrative embodiment. Buttons 450 arranged on the left side of the screen correspond to themes available to a user, while buttons 460 arranged on the right side of the screen correspond to genres available to a user. Button 470 exits this selection mechanism. It will be appreciated that other arrangements of the buttons are available, and that the list of themes and genres provided is for non-limiting, illustrative purposes only.

Similar to the genres and themes search, a user may, for example, search for a song using a Personal Music Assistant, an exemplary process for which is shown in Figure 4A. Preferably, after pressing a Personal Assistant button (step 402), the jukebox would ask certain information to identify the user if the user is not already identified (step 404). Such information could include, for example, age (or date of birth), preferred style, background, place of birth, or other information that could be used to generate a profile of the user. The jukebox then preferably could compare the profile information to selections made by other users with similar profiles (step 406) from, for example, the specific jukebox, the particular establishment, or a national database and recommend songs (step 408). For example, the jukebox might suggest a song by "The Doors" to a male user from California who was born in 1960. The user could then choose a song from that list or initiate a new search (step 410).

Furthermore, instead of entering identifiers, as shown in Figure 4B, the Persona! Music Assistant could recognize a user in other ways (step 422), such as, for example, after a credit card or a pre-programmed site-specific identification card is swiped by the jukebox. Preferably, the Personal Music Assistant would maintain a list of selections made by the user. The user's list of selections could be maintained, for example, on a local jukebox terminal, on a site's central jukebox server, on a remote server, or on an identification card, etc. After the Personal Music Assistant recognizes the user, it could then recommend songs based on, for example, songs by the artists the particular user enjoys (step 426), songs played frequently by the user (428), songs not heard recently by the user (430), etc,

Additionally, a Personal Music Assistant recognizing, for example, a preferred customer or a customer with a large number of credits might morph the jukebox into a jukebox more enjoyable to that specific user. In a related feature, certain patrons may "rent" a jukebox for, for example, a special occasion. Accordingly, a patron could pay a certain amount of money to morph the jukebox into a particular type of jukebox. For example, a patron having a party (e.g. a birthday party) at a particular location could rent a jukebox and morph it into a jukebox that allows only that patron's favorite songs. In another related feature, a patron could exclusively rent a jukebox and prevent others from selecting songs by, for example, programming a playlist through another device (e.g. the web or other communications network operably connected to the jukebox system) or by restricting access to the jukebox through a password or the like. Credits could be, for example, purchased by the user; or given to the user as a reward, for example, for purchasing drinks or souvenirs at an establishment, or for being a regular repeat customer. Thus, a Personal Music Assistant could be implemented in a manner that makes selecting songs a more enjoyable, dynamic, and responsive process while removing the immediate pressure placed on the user to know which song to choose.

When a search is initiated from screen 40, the screen changes to that shown in Figure 5 to display the results of the search. As shown in Figure 5, the results of the search are listed. More particularly, in this example, a list of songs that satisfy the search request are listed. The list could also be by album if the search was album

based. The user can scroll through the search results using slide bar 53. The user is also shown a display 55 of the number of current credits and a display 56 of the number of credits that are required to download a song from the local server to the jukebox device. The user can go back to the previous screen by touching the "Back" button 57. If the user selects a song from the search list and then touches the "Get It Now" button 54, the jukebox is operable to immediately download the selected song from the local server to the jukebox for play on the jukebox. The downloaded song can be queued up with any other selected but unplayed songs (if any) for play on the Jukebox. In this example, the download costs five credits instead of one credit like a normal selection from the storage of the jukebox itself. Once the downloaded song is played, it may be deleted from the jukebox device (together with any graphical data, such as the album cover graphic) that was also downloaded from the local server in connection with the song download). Preferably, however, the song will be kept in a "smart cache" to determine if and/or when the song should be deleted. For example, the song may be cached for a predefined period (e.g. a day, a week, two weeks, etc.) while the jukebox monitors plays of the song. If the number of plays are above a certain threshold for the predefined period, for example, the song may be kept: if the number of plays is below the threshold, the song may be deleted. The song may or may not be added to the master list of readily available songs, depending on the specific embodiment of the invention - e.g. the song in "smart cache" may appear as if it is not readily available to the user while in fact it is stored on the jukebox in, for example, a protected area. In this way, the user has the option, through use of the "Tune Central" button, to temporarily obtain on the jukebox any song from the master library of recordings without the need to contact the central server 12. As a result, the jukebox provides a more enjoyable experience for the user, while also increasing revenue generated thereby.

Also providing an enjoyable experience for the user is the central servers' capability to identify "hot hits," preferably in real-time. Preferably, new songs could be made available in a master catalog - that is, they need not reside on local servers or expanded media storage. Then, songs played frequently in a given area (ranging from, for example, a single site or group of sites, to a state or country, to a global

connection) could be identified as popular. These songs, or "hot hits," preferably could be downloaded by, or sent to, individual jukeboxes. Individual jukeboxes preferably would maintain lists of "hot hits" in real-time, allowing users to search through the most popular songs at any given time. Alternatively, a jukebox might maintain a list of "hot hits" without downloading the popular songs, thereby potentially saving download time and resources. As a result, the jukebox could provide an enjoyable experience for the user by providing easy access to the most popular songs. Thus, a jukebox may have access to a "hot list" of "hot hits" generated by a jukebox, jukeboxes, designated by an operator, created automatically and/or especially for enabled jukeboxes, etc. This "hot list" may work as an alternative to themes and/or genres, and/or it may function with themes and/or genres to provide the most popular ("hot") songs within a given theme and/or genre. Similarly, a user also may select a song from a list of recently played songs. Such a list may be maintained (e.g., stored and updated) on an individual jukebox, or it may be maintained at a central location so as to be accessible and/or updatable by any suitably configured jukebox device.

Figure 5A shows an exemplary process for maintaining a "hot list" on a jukebox with a broadband connection. It should be noted that the same process could apply for a system with a different type of connection, though more time and resources may be used to download a song over a slower connection. In step 502, songs from a master catalog are received by a site's central server. Of course, it should be noted that songs could be stored to a local jukebox's storage media. In step 504, a user using a jukebox terminal would select a "Hot List" button. After the "Hot List" is displayed (step 506), the user could select a particular song or initiate a new search (508).

Figure 6 shows another exemplary screen shot of a song selection screen 60 that is displayed when a user touches an album cover graphic from the screen 30 of Figure 3. Thus, this screen shows an alternative (or typical) method of selecting a song, wherein the song is selected directly from the subset of songs that are directly available from the storage device of the jukebox itself (rather than the local server). In this example, Joe Cocker's Greatest Hits was selected from the screen of Figure 3.

As shown in Figure 6, the resulting screen display 60 shows the selected album graphic 61 and a list of the songs 62 that are available on the jukebox for that album. The jukebox may or may not include all of the songs for a particular album. The available songs can be scrolled through if necessary using scroll bars 63a and 63b. The user has the option, through the "Play" button 65, to select a song from the list for play on the jukebox. A "Play Now" button 66 is also provided for enabling the user to select a priority play of the song, thereby giving the song a higher priority than songs selected using the "Play" button 65. This priority feature preferably requires more play credits than the normal play. A display 67 shows the number of credits available for the user. Button 64 shows other albums for the same artist being shown at 61, thereby enabling a user to easily search through the albums for a particular artist for a desired song.

As also shown in Figure 6, a "Tune Central" button 68 is displayed that enables the user to search for songs by this same artist on the local server, as explained in connection with Figure 4. In other words, button 68 takes the user to the search screen 40 of Figure 4 for searching the local server. The user can then proceed to search the local server and select songs therefrom, if desired, as described above in connection with Figures 4 and 5. Thus, as explained above, the user can access the local server at various screens in a convenience and efficient manner, depending on the desires of the user when interacting with the jukebox screen.

As can be seen from Figures 3-6, the user is provided with the option of playing songs that are resident on the jukebox device itself or, alternatively, selecting songs from the local server for download and play in an efficient and reliable manner, thereby significantly improving the operation of jukebox systems, particularly those that cannot quickly, easily or reliably receive downloads of music on demand from a central server. It is noted that the screen shots of Figures 3-6 are only exemplary and any suitable screen configurations can be used to provide the functionality described herein. In addition, the jukebox operator is provided with the ability through operator screens (not shown) to set filters per genre or style of music in order to limit access to the end user and avoid undesirable music being played at a specific location. For example, these features preferably are available through schedules defined by a

jukebox operator and/or location manager, enabling, for example, a location manager to prevent hard rock or heavy metal music to be played before 1 am. The same features can be used, for example, by operators or location managers to configure playlists and/or set the available music selections for particular dates and/or times. For example, operators or location managers may wish to restrict musical selections to Irish music on Saint Patrick's Day, love songs on Valentine's Day, Frank Sinatra music during dinner hours, etc.

In still other exemplary embodiments, a location might provide a "Virtual DJ." According to one such system, a prepared playlist would be provided to the jukeboxes of a location. Preferably, the playlist can be tailored to the details of the location and/or the event to take place, such as, for example, locale, theme, target audience, etc. Furthermore, some illustrative playlists could be created by famous DJs to attract patrons. DJs could attach their names to playlists and associate themselves with various locations, even though they are not present. A playlist might extend for several songs or for several hours, depending on the needs of the location. It is to be appreciated that coinage need not be collected during such a Virtual DJ event, as a special fee can be associated with the service of using a special playlist. It also is to be appreciated that DJs may prepare special playlists for one-time use, or they may make available repeatable special playlists. These repeatable playlists may, for example, be downloaded by jukeboxes or provided to the jukebox in any suitable manner. Thus, making third part playlists available for use on the jukebox for a fee provides a whole new revenue source for the jukebox operators, location owners and playlist generators, such as well-known DJs or the like. When a playlist is made active, the normal functions of the jukebox are preferably stopped during play of the songs on the playlist.

Figure 7 shows a block diagram of the electronic elements that define the local server 22 in accordance with an exemplary embodiment. As shown in Figure 7, the local server 22 includes a CPU 72 (e.g., AMD Elan 800 MHz), a flash memory (e.g., 8MB) containing the BIOS and OS, a pair of master/slave hard drives (82, 84 and 86, 88, respectively), a pair of IDE controllers 78 and 80 for the hard drive pairs respectively, a RAM 16 (e.g., 32MB), an Ethernet controller for controlling

communication with the jukebox device 16, and the appropriate buses interconnecting the various elements. Of course, other configurations or arrangements for the local server 22 may be used. A unique identifier may be provided in the local server for enabling the local server to be uniquely identified and registered by the jukebox and/or central server. The identifier may, for example, be located in the flash memory 74.

As will be appreciated from the description of the invention above, the addition of the local server significantly enhances the operation of the jukebox devices that are part of a jukebox system. However, the local servers also provide other benefits and features that will now be described.

A collection of local servers 22 may be used as a network of distributed servers that can be controlled by the central server 12 through its associated jukebox device 16 to provide music services to other devices. For example, the local servers and associated jukebox can be used to deliver requested songs to a dedicated residential or commercial jukebox device (or other suitable jukebox device) in addition to providing song services to the specific jukebox to which it is connected and assigned. Thus, the network of distributed servers can provide a support network for implementing residential and commercial jukeboxes of the type which allow a user to download songs for reproduction and/or storage at a residential or commercial location for an appropriate fee. As a result, the jukebox system operator can provide and control commercial jukeboxes and well as residential jukeboxes through the jukebox system. In this embodiment, the jukebox device and/or local server are connected to the Internet (or other suitable network) using a broadband modem and is provided with software that can selectively deliver song files to any dedicated residential jukebox device (also connectable to the Internet) under control of the central server. The central server receives requests from a residential jukebox and, by analyzing traffic on the network, provides instructions to a selected jukebox device to download the requested song file (either from its memory or from the local server) to the residential jukebox for a fee or under a subscription plan for the residential jukebox. In certain exemplary embodiments, requested songs may be streamed to a jukebox. It will be appreciated that streamed media may originate at a

dedicated server, a network of streaming servers, from a jukebox or jukeboxes (such as peer-to-peer or multipeer downloading), etc.

In accordance with another exemplary aspect of the invention, the local server and jukebox device are used, under control of the central server, to provide management services for other types of coin operated or payment triggered equipment, such as gaming devices, installed in the same location as (or in close proximity to) the jukebox. In other words, the jukebox system is preferably used to update the functionality of and/or manage other downloading devices present in the same location. As a result, the jukebox becomes a "central hub" for all downloading equipment in a location. This feature is achieved, in one embodiment, by networking all of the downloading devices in a single location together with the jukebox and local file server. The central server can then download information to the local server together with instructions to the jukebox as to which devices should updated with what data and/or software. The jukebox device and central server can also be used to collect information from the other downloading devices to which it is managing and upload that information to the central server for reporting/accounting purposes. Thus, the owner/operator of the jukebox system can act as a third party service provider to other coin-op companies for the purpose of managing and/or updating their equipment,

The large amounts of memory provided by the local servers and the fact that they are provided and accessible at thousands of locations over a well controlled network, turns the jukebox system into a powerful tool that can be used to perform a variety of functions in the coin-op industry. More and more coin-op manufacturers are going towards games that are software upgradeable through their internal hard drives. These updates are done periodically, but as these devices increase there will be an ever increasing need for a system that can reliably and efficiently perform the updates from a remote location. The jukebox system described herein satisfies this need by enabling all suitable electronic coin-op devices at a jukebox location to be managed by the central server using the jukebox and local server at the location. The central server can download software or data updates, store them on the local server and then dispatch the updates to the intended units of equipment in the establishment.

Thus, the jukebox system can act as a third party service provider to other companies in the coin-op business, thereby significantly enhancing the functionality of the jukebox system.

As an example, there are currently about 140,000 Merit coin-operated countertop devices in the USA, each of which enables users to play games and the like for a fee. Many of these devices operate with a hard drive that can be upgraded with new software. Merit does this by shipping CD-ROMs to operators who then need to drive to each location and manually update each machine. In accordance with the instant invention, however, all suitable coin-op equipment at a location are connected (directly or indirectly) with the local jukebox and local server assigned thereto. This enables the central server to receive the intended software update for any device, together with information that identifies what devices are to be upgraded with what software. The upgrade services are preferably fee based and provide an additional revenue stream for the jukebox system. The central server then downloads the software to the local servers with the upgrade instructions to further download the upgrades to the appropriate device(s).

As explained above, the local server enables songs to be downloaded to a commercial jukebox to which it is assigned or to residential jukeboxes under control of the central server. In addition, the local servers can be used for an on-premise networked application which manages other coin-op devices. These various features of the instant invention are illustrated in Figure 8.

Figure 8 shows a block diagram of a complete jukebox system network as contemplated by an exemplary embodiment. As explained above, the system includes a central server 12 connected to a communications network 14, a series of commercial jukeboxes 16a, 16b and 1.6c with associated local music file servers 22a, 22b and 22c, a series of residential jukeboxes 100a, 100b and 100c connected to the network via broadband devices 102a, 102b and 102c, and an on-premise network shown on the right hand side of Figure 8. This on-premise network includes a jukebox device 16d connected via a router or network, hub 110 to a local file server 22d, a number of additional coin-op equipment, such as a dart game 104, a golf game 106 and a countertop videogame 108, and a broadband modem 112 connecting this

local network to the communications network 14. With this exemplary configuration as shown in Figure 8 all of the functionality described herein can be implemented through the jukebox system of the instant invention.

Figure 9 shows an overhead view of an exemplary establishment layout for a multi-zone jukebox system, in accordance with one embodiment of the invention. In accordance with an exemplary embodiment, the establishment has three zones 121, 123, 125. Each zone is equipped with its own set of speakers 127, 129, 131, which are operably connected to the jukebox 133. Different music may be played simultaneously in all three zones 121, 123, 125 and all the music may be played from a single jukebox 133. The jukebox 133 may be provided with additional hardware, as needed, to allow this implementation.

Alternatively, the user may elect to have a song played in more than one of the zones 121, 123, 125 simultaneously, or in more than one of the zones at different times. The user may have to pay additional credits to implement either of these features. A preferred embodiment of a multi-zone system could play music at a high quality in the different zones using the system described in application Serial No. 11/023,390, filed December 29, 2004, entitled "Wireless Digital Transmission System for Loudspeakers," which is a continuation in part of Serial No. 09/161,584, filed on September 28, 1998. The entire contents of both applications is incorporated herein by reference. Using this system, for example, a jukebox could compress and transmit audio data through AC power lines to an addressable location, where it could be received, decompressed, converted, and played. In fact, any of the jukebox components herein could be implemented in a manner that uses AC power lines as a communication network for operation.

It is to be appreciated that Wireless Digital Transmission System can be used for other purposes in other embodiments where data needs to be sent between two or more devices. For example, this system could be used to configure dummy terminals. In such an embodiment, the Wireless Digital Transmission System could be used to send information such as, for example, whether to morph, what songs are appropriate given a particular morphing of the jukebox, the zones in which selected music should be played, maximum volume levels, etc, in addition to sending music to the speaker

systems. The operator may also restrict what kind of music is available in a given zone, based on the type of activity in the zone, the time of day, or any other suitable selection criteria. For example, in Figure 9, zone three 125 is a restaurant. Restaurant patrons may not wish to listen to the same type of music as someone in zone one 121, which is a bar room in Figure 9, or in zone two 123, which is a pool room. The operator may recognize this and restrict the type of music that can be played in zone three 125. Alternatively, the operator may restrict the volume of the music in any given zone. For example, patrons of a pool room 123 or a restaurant section 125 may not want the music as loud as it is desired to be in the bar room 121. And maybe the restaurant section 121 is to be kept quieter even than the pool room 123. The owner can adjust and control all suitable settings to provide the most versatile, patron friendly environment in each of the zones, based on any suitable criteria.

Figure 10 shows an overhead view of an exemplary establishment layout for a multi-zone jukebox system with selection terminals in each zone. In accordance with an exemplary embodiment, the bar has three zones 121, 123, 125. Each zone is equipped with its own set of speakers 127, 129, 131, which are operably connected to the jukebox 133. Different music may be played simultaneously in all three zones 121, 123, 125 and all the music may be played from a single jukebox 133. The jukebox 133 may be provided with additional hardware to allow this implementation.

In the embodiment of Figure 10 there are also one or more "dummy" terminals 137, 139 located throughout the establishment. An exemplary illustrative dummy terminal could use X-server technology. These terminals 137, 139, which may be stand alone devices or may be provided as part of the interface on a gaming machine or other suitable device with a digital display, allow selection of songs from the jukebox 133 for the zone in which they are located (or possibly other zones). These terminals 137, 139 duplicate the zone restrictions imposed on the main jukebox interface and selection criteria. The terminals 137, 139 may be restricted to only allowing selection of music for play in the zone where each respective terminal is located, or they may allow selection for play in one or more different zones.

Additionally, the graphical interface of the terminals 137, 139 may change in accordance with available selections, themes of the bar, themes of the room in which each terminal is located, or any other suitable criteria.

Figure 11 is a flowchart showing an exemplary implementation of a zone selection process for a multi-zone jukebox system. In accordance with an exemplary embodiment, the jukebox first begins the transaction 141 with the user. The user is instructed to select a song 143, and select one or more zones 145 in which the song is to be played. The jukebox then determines the price based on the number of zones selected 147. The jukebox accepts payment from the user 149 and queues the song for play in the selected zone or zones 151. Next, the jukebox checks to see if the user would like to select another song 153. If the user wants another song, the process returns to the select song step 143 and repeats from there. If the user is finished making selections, the process ends 155.

Figure 12 is a flowchart showing an exemplary implementation of a priority play by zone selection process for a multi-zone jukebox system. In accordance with an exemplary embodiment, certain jukebox systems may be provided with one or more priority queues corresponding to one or more zones. If priority play is provided for a zone or zones, the jukebox first checks to see if the user would like to select priority play for the selected song 161. If priority play is selected, the jukebox then provides an option for the user to choose a zone or zones in which priority play should occur 163. Based on the number of zones selected for priority play, the jukebox determines a price 165, and accepts payment of that price 167 from the user. The jukebox then places the song in a priority play queue for each selected zone 169.

Figure 13 shows an exemplary implementation of a multi-zone set of priority and non-priority queues, with a subset of queues for each zone. In accordance with an exemplary embodiment, each of N zones 171 may be provided with its own set of queues, comprising a priority queue 175 and a non-priority queue 173. A list of songs selected for play is maintained within each queue 173, 175. Each song in each queue may be provided with an identifier 177, 179, which identifies the song, and/or the position of the song in the queue, and/or any other suitable factors.

In certain exemplary embodiments, the songs in a queue also may be reordered for other reasons. For example, the songs may be reordered to smooth transitions between songs, thus providing better (e.g., more aurally pleasing, less abrupt, more smooth, etc.) end-to-beginning playback of the songs. Thus, in certain exemplary embodiments, a method for altering the order of playback songs is provided. In such a process, the "color" of the song may be determined for two or more songs in the play list. The color of a song relates to characteristics of the song, such as, for example, the volume, rhythm, beat, tempo, etc. of the song. After the color determination, the colors may be compared, and the songs may be reordered to group like (or dislike) songs. Optionally, the color of the song(s) may be determined and also may be used to control lighting, displays, etc. to provide a more interactive experience.

Figure 14 is a flowchart showing an exemplary distribution and initialization scheme for a jukebox with morph capability. In accordance with an exemplary embodiment, the contents of a factory drive are defined at the point of manufacture 181. This same drive (or expanded media storage) may be shipped out with all jukeboxes 183, and may, for example, only contain a subset of the total number of songs available on the central server. This feature simplifies the manufacturing process by enabling the identical drive to be shipped with all jukeboxes. Once the jukebox containing the drive has reached its destination, the operator may select a subset of songs on the drive as the basic playable list 185. This selection can be made based on the type of establishment, the type of music the establishment's patrons typically prefer, or any-other suitable criteria. The operator may also allow the central server to recommend a basic playable list. The drive may also allow selection of songs not on the basic list, but present on the factory drive, for an additional fee 187. This list of "alternate" songs might not include all songs however, as the operator might desire to restrict access to songs that don't meet the theme of the establishment. For example, a country bar owner might not ever want to allow selection of rap or hip-hop songs on the jukebox. In certain exemplary embodiments, a jukebox will ship with a large number of songs (e.g. 25,000 songs) on a very large storage medium, such as, for example, a hard drive, flash media, etc. The jukebox

according to this embodiment would be without a "personality" until an operator and/or owner selected a subset of all songs to make available to patrons. In this way, operators and/or owners can define "personalities" for their jukeboxes through definition of the basic playable list.

Once the songs on the drive have been appropriately categorized (i.e. the basic playable list has been defined), the jukebox begins operation 189. As long as a new basic playable list is not desired 191, the jukebox continues to operate 189 with the currently selected basic playable list. If a new basic playable list is desired 191, the jukebox morphs 193 into a "new" jukebox, selecting a different playable subset of songs for basic selection 185, and preferably changing additional characteristics (e.g., graphics, ads, look and feel, etc.) of the jukebox as dictated by the morph. The ability to morph jukeboxes thus enables owners and/or operators to maintain a single jukebox with a plurality of triggerable personalities which can be switched between automatically, at pre-defined events/times, manually, etc.

Figure 15 is a flowchart showing an exemplary implementation of an automatic jukebox morph initiation process based on a triggering event. In accordance with an exemplary embodiment, the user may define an event 201, for example a themed night or a time of day, as a triggering event which triggers a jukebox morph. The jukebox then operates as normal 203, checking periodically to see if the triggering event occurs 205. If the triggering event has not occurred, the jukebox simply continues to operate 203, but if the triggering event occurs, the jukebox is morphed into a "new" jukebox. The triggering events may be one time events, or they may be scheduled to occur weekly, daily, monthly or scheduled based on any other suitable criteria. It should be noted that in a multi-zone configuration, different zones may be morphed while others do not change. This feature of the illustrative embodiments allows, for example, a given zone or zones to be dedicated to a certain kind of music while the other(s) may vary based on any variety of factors, such as the time of day, an owner's desire to change the music, or a user's request.

Figure 16 is a flowchart showing an exemplary implementation of a jukebox morphing process. In accordance with an exemplary embodiment, when the jukebox begins morphing 211, it selects a new subset of songs to be the basic playable list

213. The jukebox then allows some or all of the remaining songs on the jukebox to be selected for an enhanced fee 215. Some of the remaining songs may be restricted based on what triggered the morph. Other characteristics of the jukebox may also change 217, for example, the user interface may be changed, and different advertising may be displayed which corresponds with the predicted tastes of the crowd for which the jukebox has been morphed. Other suitable changes may also be made. In one example of a preferred embodiment, a club owner has a hip-hop night on Wednesdays, beginning at 9:00 pm and ending at 4:00 am. At 9:00 pm on Wednesdays, the jukebox morphs into a hip-hop jukebox, with a basic selection of appropriate music. In accordance with the morph, the jukebox blocks all access to genres of music such as country music, classic rock, jazz, blues and oldies, and the jukebox limits the available selection of hard rock additional songs to "hip-hop-esque" hard rock songs. The graphics on the jukebox convert to edgy, urban graphics, and the advertising changes accordingly, displaying products such as apparel, drinks, and goods which should appeal to the hip-hop crowd. At 4:00 am, the jukebox morphs back into the "standard" jukebox for that club, or into any other suitable jukebox. Alternatively, the jukebox may remain set in hip-hop mode until the next triggering event occurs. Again, it should be noted that in a multi-zone configuration, different zones may be morphed while others do not change. In the above exemplary non-limiting embodiments, the system might morph into hip-hop in one zone for the night, while the "standard" music for the club remains playing in another area.

Figure 16A is an exemplary screen shot showing how a morph of a jukebox can be initiated in one preferred embodiment. Area 160 is blank to indicate that no selections have been made. A user will select a morphing option by choosing, in this exemplary embodiment, for example, either TouchTunes Master button 162, Dynamic Master button 164, or Clone a Location button 166. It is to be appreciated that other embodiments may provide further morphing options in place of or in addition to the aforementioned buttons. In this embodiment, pressing the TouchTunes Master button 162 will populate area 160 with a list of master selections pre-selected by the TouchTunes music department. These master selections may include, for example, collections of specific songs (e.g., all songs by British bands, all songs by a particular artist, etc.), ratios of particular themes or genres (e.g., a mix

of 75 percent Hip-Hop and 10 percent Pop, 10 percent Rock, and 5 percent techno for a particular dance club feel, etc.), etc. A user may peruse the list using slider bar 170. Once the user is satisfied, pressing Morph It button 168 will morph the jukebox. It is to be appreciated that pressing the Morph It button 168 may trigger an immediate morph, or it may save the morph and instead initiate it at a predefined trigger time. It also is to be appreciated that morphing data may be stored for morphing the jukebox on which the selection was made or any number of connected jukebox terminals.

Pressing Dynamic Master button 164 is one exemplary, non-limiting way of providing increased user control over the morph. Figure 16B is another exemplary screen shot showing how a user can specify various criteria before a morph of a jukebox in one preferred embodiment. A discussion of the buttons with identical numbers in FIG. 16A and FIG. 16B is omitted. Designations for the genres and/or themes of music are provided, such as, for example, Rap label 172. To specify a given percentage of music to be played, a user may press increase button 174 or decrease button 176. The amount of music selected, for that particular genre or theme will appear in a theme music amount area 178, while details about the total amount of music selected will appear in total music amount area 180. In a preferred embodiment, a user may specify percentages of a genre and/or theme to play. For example, a user may select only 100 percent Country music for a country night. Alternatively, as another non-limiting example, a user may select a mix of World and Latin music for an international feel. It is to be appreciated that although a preferred embodiment uses percentages, other embodiments may use different measures for specifying how many of which type of music should play, such as, for example, the number of songs in a given category, the amount of playtime in a given theme and/or genre, etc. It also is to be appreciated that in other embodiments, a user need not use the buttons to increase or decrease the amount of music; that is, in other embodiments, a user may directly input the amount of music via an input mechanism. Thus, it is to be appreciated that morphing may be defined and/or triggered by a remote device operably connected to a jukebox,

When a user presses Clone a Location button 166 (e.g., from FIG. 16A or FIG. 16B), the jukebox displays a list of the jukebox devices, preferably including a

unique identifier for each jukebox device. In other embodiments, however, the list may include, for example, other locations owned by the same owner of the present location, etc. After selecting the jukebox terminal, the present jukebox could morph into that jukebox. Alternatively, the jukebox selected could morph into the present jukebox, thus providing, for example, remote morphing. Still alternatively, the list might include information about non-digital jukeboxes that exist in addition to the digital jukeboxes preferred by the exemplary embodiments. This morphing feature can allow users to replace existing jukeboxes quickly and easily by allowing the new jukebox to morph into an older jukebox that is being replaced, such as a conventional record, CD or digital jukebox. Additionally, operators who maintain multiple jukeboxes may, for example, apply a route-based cloning and/or morphing process to provide, for example, a custom or particularized, set of songs.

Still further, a user can morph a location based on a location's data, specific to, for example, a given date or the latest data known. This allows users to morph a jukebox based on, for example, the popularity of a given morph - e.g., users may choose to revert to a successful morph to try to recreate the entertainment of that evening. This feature also enables users to recover from hard drive crashes and/or other jukebox failures. For example, if a hard drive has been replaced after an old one crashes, a user may morph (or clone) an existing location at a given date and/or time before the crash to restore the songs that were available before the crash because, preferably, the server maintains information sufficient for this functionality. Thus, using the morphing function gives users increased control while also providing backup and restoration features.

Figure 17 shows the relationship between a jukebox with expanded media storage and a central server. In accordance with an exemplary embodiment, the central server 221 contains a master library of songs, such library comprising all songs that are currently available to be downloaded and all songs currently installed on jukebox hard drives. The central server may communicate 222 with the remote jukebox 225 containing a local hard drive 223. The hard drive 223 on the jukebox may have several sections, including available space for downloads 227, space occupied by preloaded songs 228, and space for software and an operating system 229.

Additional suitable sections may be added, for example, a section containing different pictures for altering the GUI. The jukebox 225 may communicate with the central server 221 to download songs, upload usage information, update software, and perform any other suitable functions.

Figure 18 is a flowchart showing an exemplary process for a song selection process when a song is not in the "standard" available playable song list. In accordance with an exemplary embodiment, the user first selects a song 231. The jukebox checks to see if the song is available on the local hard drive as a "non-standard" selection 233. If the song is available on the local hard drive, the jukebox charges the customer the price set for obtaining and playing a non-standard song 235 and plays the song 237 (or adds it to a playlist, when appropriate).

If the song is not available on the local hard drive, the jukebox checks to see if a high-speed connection to the central server is available 239. If there is no high-speed connection, the jukebox informs the user that the song is temporarily unavailable 241 and orders the song for download 243. The jukebox may or may not charge an additional amount for ordering the song. If, however, there is an available high-speed connection to the central server, the jukebox orders the song immediately and uses the high-speed connection to download the song right away, queuing it up for playing 245. The jukebox then charges the customer the price of a non-standard selection 247. In certain exemplary embodiments, a jukebox may retrieve songs offline, either after a location closes or before it opens. In certain exemplary embodiments, a jukebox may immediately download a song over a dedicated line with a dial-up connection. In certain other exemplary embodiments, a song may be downloaded from another jukebox (or other jukeboxes) rather than from a central or limited database to reduce network strain. In certain exemplary embodiments, the jukebox may download songs via a distributed media service in which portions of a given song may be downloaded from a plurality of sources and reassembled for the target jukebox. It is to be appreciated that such a peer-to-peer (or jukebox-to-jukebox) or multipeer (several jukeboxes to jukebox) digital downloading network will need to track song licensing information. In certain exemplary embodiments, if a song is not available on a jukebox but other versions or covers are available, the

jukebox may recommend to these other songs to the searching user. For example, a user searching for an unavailable Trisha Yearwood version of "How Do I Live" may be recommended available versions by Dolly Parton and/or LeAnn Rimes. Preferably, recommendations will be smart enough to ignore similarly named songs that are completely different, such as, for example, the Everly Brothers' "Oh, Pretty Woman" and the Motley Crue's "Pretty Woman."

The factory drive explained above, combined with the morphing capabilities, eliminates the need for the local server explained above, as the factory drive can prove the same services as the local server, without the need for a separate hardware device. In other words, the factory drive embodiment herein enables a jukebox to be shipped with a single mass storage device of any technology (or multiple technologies and/or multiple devices acting as a single mass storage device), while still enabling a basic playable list to be defined, an expanded playable list to be defined, morphing capabilities, local server services to be provided, as well as all other features described herein. The content of the factory drive, as shipped, is preferably defined using historical, statistical information on customer preferences.

Figure 19 is a flowchart showing an exemplary process for a priority play queue with prioritization-based-on-bidding capability. According to an exemplary embodiment, the user first indicates that he would like priority play 251. The jukebox then displays the current status of the priority play queue 253. This display may include information such as how many songs are in the queue, what the top bid is, how much has been bid on each song, which songs are "locked in," and any other suitable information about the priority queue. The jukebox then allows the user to select how much additional money the user would like to pay to place his song in a particular spot on the priority list and accepts payment in the selected amount 255. After accepting the payment 255, the jukebox places the song in a position on the priority list corresponding to the additional amount received from the user 257.

Alternatively, in another exemplary aspect of the illustrative embodiments, a user can bid on the right to have a song played before other songs previously selected for priority play are played. In a preferred embodiment, the user is shown the top

price paid for a priority play, and can pay more than that price to obtain the highest priority available.

Another exemplary aspect of the illustrative embodiments does not allow a user to be shown how much anyone else has paid for priority. The user can pay however much the user desires to spend to obtain a priority ranking, and then receive a ranking of priority based on the amount paid.

In accordance with a further exemplary aspect of the illustrative embodiments, a user can pay however much the user desires to spend to obtain a priority ranking in accordance with the previous exemplary aspect, and then be shown the priority spot which has been obtained based on the paid amount. If this spot is not satisfactory to the user, the user can pay additional money to move the song up in priority ranking, and be shown the new priority ranking obtained based on the additional money paid. The user can repeat this process until the desired priority ranking has been obtained. The user can also pay additional money to make it harder for other users to pre-empt the selected priority spot on the list in a bidding-type situation. Any other suitable method of increased-pay-for-increased-priority may also be implemented,

In accordance with an additional exemplary aspect of the illustrative embodiments which may provide a "lock in" feature, a user can "lock in" a priority ranking with a payment of a pre-selected amount. For example, if a user pays 15 credits to obtain a ranking of 3rd in priority, and wishes to guarantee the third ranking, the user may pay, for example, 4 more credits to "lock in" the ranking. Since locking in the ranking may require the "lock in" of all the rankings above the user as well, the user may be required to pay a certain amount to "lock in" all songs above the user's selection. In one such situation, the user can either choose to pay the price quoted for the "lock in" or pay the same or a varying amount of credits in an attempt to prevent future over-bidding or to move the user's song up further in the priority list.

In accordance with another exemplary aspect of the illustrative embodiments, any of the aforementioned bidding strategies may be implemented, and the user may be shown how much everyone has paid for their particular rankings. This allows the user to know exactly how much he will have to pay to obtain a certain priority

position. If the "lock in" feature is implemented, this will also let a user know if it is cheaper to pay the price to "lock in" the song or to pay to move up on the priority list. All of these options result in increased revenue for the operator.

It should be noted that although the embodiments above describe a system for distributing media to non-movable jukeboxes, alternative embodiments using similar systems could distribute media to portable jukebox devices and are contemplated by, and within the scope and spirit of, this invention. A portable jukebox may be, for example, a PDA, a cell phone, or any other movable device capable of receiving and playing music. Furthermore, media may be distributed to portable jukeboxes using the above described methods (e.g. through a broadband connection, wireless connection, etc.), or any other appropriate method, more suited to the particular portable device, such as, for example, using Bluetooth technology. Additionally, the jukeboxes described above typically are for commercial purposes. However, jukeboxes for other purposes such as, for example, playing residential media, also are contemplated by, and within the scope and spirit of, this invention.

In other exemplary embodiments, an employee of a location with a jukebox can be rewarded based on the number of plays customers pay for on the jukebox. In one such exemplary embodiment, the system keeps count of how many plays have been paid for by customers. Every time a certain number of plays have been paid for, the bar employee receives one free play. In some embodiments, a separate counter will be maintained, while in other embodiments only one counter may be necessary to calculate the number of free plays available. Thus, certain exemplary embodiments encourage employees to promote the jukeboxes by compensating the employee with free plays. This also can help reduce the similarity of song-selection throughout the day, preserve free plays for employees of different shifts, etc.

Similarly, jukebox users may vote for particular instances of media to alter their priority in playlists. In this way, jukebox users can, for example, "battle" for control over the music to be played in a particular zone or particular zones within or among locations. It will be appreciated that this voting/battle mode may be implemented by using, for example, a dynamic queue, a priority queue, multiple queues, etc. It also will be appreciated that a jukebox could put into a voting mode

automatically (e.g. at a particular time of day and/or on a particular day of the week), or it could be triggered manually. FIG. 19A is a flowchart showing an exemplary implementation of a jukebox voting process. Unlike conventional jukebox operations, or even jukebox bidding modes, a list of songs in a jukebox queue is displayed to users in step SI 902. This list may be generated automatically, by operators, bar managers, patrons at a bar, etc. The list could be displayed, for example, on a jukebox, or, more preferably, on one or more stand-alone monitors. Additionally, in certain example embodiments, the list could be viewed by a plurality of mobile devices and/or terminals. Preferably, the information displayed contains at least the artists and names of coming songs, and a number associated with the priority of those songs. The list should be sorted by this number, so that the song with the most "votes" is displayed as the "next" song to be played, followed by the next highest song, etc. Step SI904 determines whether the currently playing song is over. If it is over, step SI906 removes it from the queue, and the next song is played in step SI 908. In another embodiment, users could vote to stop/skip the currently playing song (or instance of media) by, for example, exceeding the number of votes the song had before the jukebox started playing.

As users see the coming songs, they will be tempted to push up the songs they like so the songs and/or push down the songs they do not like. In general, the more users who vote, the greater the ambiance of good songs. Thus, after step SI 908, or in the case that the song is not over, the jukebox receives users' votes for particular songs in step S1910. Voting can be based on credits (as users buy credits), or tied to a user's account. In certain example embodiments, users must login to place a vote, and in certain example embodiments, users can vote a limited number of times. Preferably, users may vote from at least two distinct places within a location. The queue is updated based on this voting in step S1912, and the process returns to step SI902, where the displayed list is refreshed.

In certain example embodiments, the queue may be based on the total number of votes for particular songs. In certain other example embodiments, users may vote for and/or against certain songs and the "net" information may be displayed, indicating the number for and against, or merely the net result. If there are more

votes against a song than for, the system can perform one or more of the following functions. For example, the jukebox simply may keep the song in the queue with a negative number of votes. Alternatively, the jukebox may keep the song in the queue with a zero or negative number of votes, but, for example, always wait until the net vote reaches at least 1 before playing the song. Still alternatively, the jukebox may drop any song that reaches zero or a negative number of votes.

FIG. 19B(1) is an exemplary screenshot showing a voting queue display. Display 1900a shows the currently playing song 1902 and its progress 1904. It will be appreciated that progress 1904 of currently play song 1902 may be displayed in any number of ways, such as, for example, as a number indicating the time remaining, a sliding bar, etc. Artist and song list 1906 preferably is displayed based on the corresponding number of votes in votes area 1908a. Optionally, user area 1910 may indicate information about the users voting for the songs, such as, for example, the user placing the song on the list, the last user to vote for the song, the name of the group "battling" to have the song played, etc.

FIG. 19B(2) is another exemplary screenshot showing a voting queue display. FIG. 19B(2) is similar to FIG. 19B(1), but votes area 1908b reflects the "net" voting of users. That is, for example, in FIG. 19B(1), the display indicates the number of votes for and against a particular song, as well as the net voting result.

FIG. 19C is an exemplary block diagram showing a voting mechanism within a given location. Display 1900a corresponds to FIG. 19B(1), though it will be appreciated that a display corresponding to FIG. 19B(2) could be substituted, as well as any other display indicating the same or similar information. The list preferably is updated by jukebox 16. Users may vote to update the list (e.g. vote for/against a song) via jukebox 16, or by any device optionally connected to jukebox 16, such as, for example, another jukebox 1912a, a remote terminal 1912b, a laptop 1912c, a PDA 1912d, a cell phone 1912e, another mobile device 1912f, etc.

FIG. 19D is another exemplary block diagram showing a voting mechanism within a given location. FIG. 19D differs from FIG. 19C in that users voting from

optionally connected devices 1912a-f may affect the voting list directly, without interfacing with jukebox 16.

An alternate application includes a voting-mechanism where multiple locations (e.g. bars, restaurants, venues, etc.) are networked together through, for example, the Internet. Those linked locations allow more patrons to "battle" against each other. In this embodiment, the networked jukeboxes would be playing the same song and collectively all users from all locations would decide on the playing order through the voting mechanism or the like. Thus, FIG. 19E is an exemplary block diagram showing a voting mechanism for use with multiple locations. A first jukebox 16(1) with a first plurality of mobile devices 1912a(1)-f(1) within a first location communicating therewith, as well as a second jukebox 16(2) with a second plurality of mobile devices 1912a(2)-f(2) within a second location communicating therewith with a may affect a playlist accessible by jukeboxes 16(1) and 16(2).

FIG. 19F is another exemplary block diagram showing a voting mechanism for use with multiple locations. It is like FIG. 19E, except first plurality of mobile devices 1912a(1)-f(1) and second plurality of mobile devices 1912a(2)-f(2) may directly affect the voting list.

An alternate application of this voting feature relates to enhancing the "pause" mode of a jukebox. A jukebox, while in pause mode, typically prevents users from selecting music from the jukebox. For example, bartenders may use a pause mode when a special event (e.g. a football game) is presented at the location and when the audio system of the location is playing the match or event. The voting feature can be used to select which songs will be played after the special event is over. It has been widely observed that patrons watching a game very often leave the location once a match is over. This alternate method of selecting songs to be played is likely to motivate patrons to stay in a location and/or continue using the jukebox, etc.

Figure 20 is a flowchart showing an illustrative process for using a user-tailored system for managing music, communications, etc. in accordance with one exemplary embodiment. It is to be appreciated that the Personal Music Assistant may

work as a complement to, component of, etc. to systems according to this exemplary embodiment. In step S2000, a user accesses the user-tailored system, dubbed MyTouchTunes in this case. It is to be appreciated that a user may access the system at a number of locations, such as, for example, via an Internet front-end, at a jukebox terminal, etc. It further is to be appreciated that the user may access the system in any number of ways, such as, for example, by inputting a username and password combination into the Internet front-end or jukebox terminal, by swiping a credit card at a jukebox terminal, by scanning a site-specific "smart-card" or flash card, etc. In certain example embodiments, a user may be recognized through biometric data, such as, for example, finger prints, retinal scans, etc. It will be appreciated that biometric authentication may work independently, or together with, more conventional forms of identifying users, such as, for example, username/password combinations, smart-cards, and the like. The system determines whether it recognizes the user in step S2002. If the user is not recognized, the user can be prompted to create account/profile information, as in step S2004. The account/profile information will be described in more detail below with reference to FIG. 21 A.

If the user is recognized in step S2002, the MyTouchTunes main menu is displayed for the user in step S2006. From the MyTouchTunes main menu, users using a system according to this exemplary embodiment may, for example, edit account/profile information in step S2010, manage music in step S2020, manage buddies in step S2030, communicate with others in step S2040, and/or access various value-added contents that may be provided by the system in step S2050. These and other features will be described in more detail below with reference to FIGS. 21A through 21B.

FIG. 21A shows illustrative options available for editing account/profile information in one exemplary embodiment. As noted above, it is to be appreciated that the exemplary steps illustrated in FIG. 21A may correspond to the steps required for initial account/profile setup. In such an initial setup case, however, an exemplary system might require the user to complete all of the required information at one time before allowing the user to access the main menu (step S2006 in FIG. 20). It also is to be appreciated that such account/profile information could be stored in various

locations as necessary to a given implementation, such as, for example, in a central location or database, on a particular establishment's server, on specific jukebox terminals, and/or on a removable card that would identify the user and the user's information when inserted into a jukebox.

A user would first input personal data in step 2012. Such personal data may include, for example, an e-mail address for the user, a telephone (preferably mobile phone) number, and/or a name. The system also may prompt the user to create a unique username and password so that the system can identify the user at later times and from other locations. Alternatively, the system may use the user's e-mail address as a username and only require a corresponding password. Still alternatively, in the case of residential jukeboxes, for example, an exemplary system may not require a password at all. Based on the personal data, the exemplary system may gather enough information to recognize the user at logins (e.g., online access, preferred location, other locations within a given area, etc.) as well as send e-mail and text messages to the user. These exemplary features will be discussed in greater detail below.

In step S2014, users may specify various locations they frequent. This feature may enable user-tracking by various establishments, advertisers, etc. Establishments may, for example, provide special offers to regulars or others in a particular area to entice them to come visit. Advertisers may monitor the personal information, location information, and playlists to provide, for example, concert tickets, special offers on CDs, t-shirts, and/or other merchandise that may seem to be appealing based on the user's preferences. Still further, if such account/profile information is stored locally on various terminals, the system may communicate with those other terminals to provide the playlists in preparation for a user's visit. In step S2016, a user may add credits and/or specify payment information. Users may, for example, use their credit cards to increase the number of credits they have before entering an establishment, or while at an establishment. Alternatively, users may become "certified" by, for example, providing credit card information and allowing jukeboxes to automatically bill their credit cards for future credits purchased, etc. It is to be appreciated that step S2016 can be used to purchase credits for oneself or for someone else, such as, for example, one's buddies, an employee of the establishment,

etc. It also is to be appreciated that a user may transfer money from an account, such as, for example, a Pay-Pal account, to a jukebox-specific account, or a user might use such an account to purchase music and/or services directly.

In certain exemplary embodiments, a user may register by creating an account at a jukebox and providing a verifying contact mechanism, such as, for example, a mobile phone number and/or email address. After a user creates an account and inputs verifying contact mechanism, a message may be sent to that location. For example, a user may receive a text message. Once that message is received, the user may be prompted to input a code to the jukebox, send a return email or text message, etc., to become fully registered and/or to take advantage of certain other advanced features, such as, for example, one or more of the features disclosed herein.

FIG. 21B shows illustrative options available for managing music in one exemplary embodiment. In step S2022, a user may create or edit already existing playlists. Users preferably can search through songs using the above-described methods and/or use system-generated suggestions to build their playlists. Preferably, users also may give their playlists descriptive titles, such as, for example, "Favorite Jams," "80's Night", "Smooth Love Songs", etc. Users may further designate their playlists as "public" or "private" to allow or restrict other users from browsing their playlists. Users may, in step S2024, rate their songs and/or playlists through the PickList feature. The PickList also may generate a custom playlist comprising the songs the user has rated the highest at a given time. Alternatively, a PickList may comprise a list based on the HotHits list described above. In step S2026, a user may play music on a jukebox using, for example, any of the above described methods with or without using these playlists. Normally, selecting a playlist for play on a jukebox will add those songs to the regular queue. However, as described above, users may pay an additional fee and/or use the bidding mechanism described above to play their music immediately. Furthermore, selecting a playlist (or playlists) for play on a jukebox normally will play all songs in that particular playlist (or in those particular playlists). However, users may unselect certain songs to prevent them from being played. This enables users to further customize playlists dynamically when they are playing their music.

In addition or in the alternative, further playlists may be imported from external sources. For example, software and/or hardware devices often store user-defined and/or custom-generated (e.g., most frequently played, recently added, etc.) playlists. By way of example and without limitation, playlists may be extracted from iPods, portable MP3 players, and/or other hardware devices. This may be done at the jukebox using an appropriate hardware connector provided to the jukebox and/or at a device remote from the jukebox (e.g., on a home computer) where it is more likely that an appropriate hardware connector will already be located (e.g., a proprietary docking port, a USB connection, etc.). In either or both cases, the playlists may be uploaded to a central audiovisual distribution server after the play list is imported to the local device. Similarly, by way of example and without limitation, playlists may be extracted from iTunes, Windows Media Player, and/or other software applications. In many cases, such devices and/or software applications store playlists in an unencrypted and/or unencoded manner such that the playlists may be read in a manner similar to a flat text file. In this and/or other ways, the imported playlists may be made available to one or more jukeboxes connected to the audiovisual distribution network. Of course, in certain other cases, it may be necessary to decrypt, unencode, and/or perform other processing on the playlists of other hardware devices and/or software applications, prior to processing to make the now-readable playlists suitable for use, for example, on a jukebox or in connection with a jukebox-related service.

This feature may be advantageous to users in that the need to create and subsequently recreate existing playlists may be reduced, at least insofar as a single playlist may be defined on a device and subsequently used on a jukebox at a later time without a significant amount of reentering, reselection, and/or regrouping of the same information. Such techniques also provide yet a further means of interacting with and customizing the jukebox to the user's preferences, e.g., by enabling the customization of playlists through devices and/or software applications with which the user may be more comfortable and/or have more experience, have different interfaces, etc.

Thus, systems and/or methods of distributing a playlist to one or more digital jukeboxes connected to an audiovisual distribution network are provided in certain example embodiments. A peripheral device having a peripheral playlist stored in a

computer-readable storage medium thereon is connected to a digital jukebox via a connector. The peripheral playlist is retrieved for the digital jukebox. This may be accomplished by programmed logic circuitry (e.g., any suitable combination of hardware, software, firmware and/or the like) provided to the jukebox. The peripheral playlist is converted to a jukebox playlist. The jukebox playlist is published at least on the jukebox. Similarly, a software application running on a computer device (e.g., a PC, a laptop, PDA, or other similar portable computing device) may be read by a computer and published to a server on the audiovisual distribution network. The peripheral device may be, for example, at least one of a portable music player, a computer, and/or a mobile phone.

The peripheral playlist may be at least temporarily stored to a computer-readable storage medium of the digital jukebox. It may be determined whether the jukebox playlist is to be designated a public jukebox playlist or a private jukebox playlist. Based on the determination, the jukebox playlist may be made publicly available or kept private to the particular registered user importing the playlist.

The jukebox playlist may be sent to a server of the audiovisual distribution network, and the jukebox playlist may be made available to jukeboxes connected to the audiovisual distribution network and/or remote devices connected to the audiovisual distribution network. The person importing the playlist may have to log into the jukebox or a portal to the jukebox before being allowed to import the playlist.

A jukebox device to accomplish the same and/or similar functionality may also be provided, wherein at least some of the functionality is enabled through the use of a processor provided to the jukebox.

In certain exemplary embodiments, it may be advantageous to prevent users from creating playlists directly at the jukeboxes using the jukebox interfaces themselves. For example, this may reduce the amount of crowding around a jukebox, monopolization of the jukebox by a single or a few users, etc. In addition or in the alternative, users may have a reduced ability to create playlists directly at a jukebox. For example, a user may be able to create a playlist from a predetermined set of songs. That predetermined set of songs may include only those songs played by the

user on that particular night, at the jukebox(es) at that particular location, at any jukebox(es) connected to the audiovisual distribution network, etc. In this way, it may be possible to enable the user to benefit from the customization of playlists techniques described herein directly at the jukebox while also reducing the time required to do so. In addition, the users may be prompted to create playlists in which songs that have been played, and thus may be likely to be played again, are added.

Thus, in certain exemplary embodiments, systems and/or methods for playing instances of media on a digital downloading jukebox are provided. A user is required to log into the jukebox. A plurality of selectable instances of media are provided for play on the jukebox. User selection of an instance of media from the plurality of selectable instances of media is enabled via a user interface to the jukebox. For each user selection, the instance of media is added to a playlist stored in a computer-readable storage medium of the jukebox. The instances of media in the playlist are played in order. The plurality of selectable instances of media is selected from a master group of instances of media in dependence on a previous play of an instance of media by the user. In certain exemplary embodiments, the plurality of selectable instances of media is formed from instances of media previously played by the user within a predetermined time period and/or from instances of media previously played by the user at a predetermined location. In certain exemplary embodiments, the plurality of selectable instances of media is formed from instances of media stored in a list of recently played instances of media associated with the user.

Once songs are added to a playlist, a playlist may be played in whole or in part. For example, if a user has enough credits, the entire playlist may be played. However, if the user does not have enough credits, the user may either enter additional payment to purchase a sufficient number of credits to play the entire playlist or may play only a portion of the song. In the latter case, the user may be prompted to select a subset of songs for play from the playlist. Alternatively, the songs from the playlist may be played in order until the credits are exhausted. Optionally, a prompt for the user to enter additional payment to play the whole playlist may be presented when appropriate, and it may be removed from a display once sufficient payment has been inserted.

Also with respect to the amount users may be charged for plays, it will be appreciated that in certain exemplary embodiments, registered and/or verified users may be given "price breaks" on plays. For example, users normally may be charged a premium for selecting a song that is not currently available on the jukebox (e.g., requires a download, e.g., from the audiovisual distribution network or other source). In certain exemplary embodiments, certain preferred users may only be charged a reduced or the minimal amount of credits (e.g., only 1 credit), no matter what the source of the song. In certain other exemplary embodiments, preferred users may only be charged the minimal amount of credits if the song was recently played by the particular user. In still other exemplary embodiments, preferred users may only be charged a reduced or the minimal amount of credits if the song exists in a playlist associated with the user (e.g., a playlist the user created, a playlist created by another that the user played in whole or in part, etc.). Thus, it may be possible in certain exemplary embodiments to reward verified users by reducing the number of credits charged for certain instances of media associated with the playlists of those users.

As such, systems and/or methods of playing an instance of media on a digital downloading jukebox are provided in certain exemplary embodiments. A plurality of instances of media on the jukebox is provided. An interface is provided to a user of the jukebox to allow the user to search for a particular instance of media for play on the jukebox. It is determined whether the instance of media is available for immediate play on the jukebox free from additional processing by the jukebox (e.g., without the need for further downloading or retrieval of the instance of media beyond that which is standard for plays, without the need for non-standard searches, etc.). When the instance of media is available for immediate play, the user is charged a predetermined first number of credits and the instance of media is queued up for play. When the instance of media is not available for immediate play, it is determined whether the user is a verified user. When the user is a verified user, a second predetermined number of credits is charged if the instance of media exists in a playlist associated with the user, but otherwise a third predetermined number of credits is charged. The instance of media is queued up for play. The second predetermined number of credits is less than the third predetermined number of

credits, and the third predetermined number of credits is greater than the first number of predetermined credits.

With respect to the charges, then, in certain exemplary embodiments, the price for a normal play may be the same as the price for a premium song requiring download when the user is a registered user. In certain exemplary embodiments, the playlist associated with the user is a playlist previously created by the user, a playlist including songs recently played by the user, and/or a playlist created by another user but recently played at least in part by the user.

The determination of whether the instance of media is available for immediate play on the jukebox free from additional processing by the jukebox may include determining whether the instance of media resides on a storage area separate from a dedicated storage area of the jukebox comprising immediately playable instances of media. The separate storage area may be a remote server, a partition of the dedicated storage area of the jukebox, a separate storage area within the jukebox, etc. The determination of whether the instance of media is available for immediate play on the jukebox free from additional processing by the jukebox may include, in addition or in the alternative, determining the user's search methodology provided by the interface. For example, a premium may be charged for a "super-search" as compared to a title or artist search. Of course, these examples are provided by way of example and without limitation, and other search methodologies described herein may be provided, and the pricing may be distinguished on the basis thereof, alone and/or in combination with the type of user and/or the existence of the instance of media in a playlist associated with the user,

Here, as elsewhere, any suitable combination of programmed logic circuitry (e.g., hardware, software, firmware, and/or any suitable combination thereof) may enable such functionality. Additionally, instructions for carrying out the same may be stored on any suitable computer-readable storage medium.

FIG. 21C shows illustrative options available for managing buddy lists in one exemplary embodiment. Users may keep track of buddies (e.g., friends, regulars at a bar, etc.). The system may use any instant messaging standard, such as, for example,

AOL Instant Messenger, MSN Messenger, Yahoo! Messenger, ICQ, a custom jukebox-specific protocol, etc. to send and receive messages. Preferably, the system would alert the user of the buddies' statuses (e.g., online, inactive, busy, etc.). In step S2032, a user may create or edit an already existing buddy list. Users may search for buddies by, for example, inputting their names, e-mail addresses, etc. Additionally, users may invite friends to join by issuing invitations, preferably by e-mail or SMS (Short Message Service) text message to a mobile phone, in step S2038.

Users also may browse their buddies' music collections, playlists, PickLists, etc., as shown in step S2034. Users then could develop their own playlists and/or PickLists based on what they perceive as popular amongst their friends. Users also may copy music from their buddies or send music to their buddies, preferably for a fee, as shown in step S2036. In a step not shown, users also may designate particular songs to play when their friends log in to the jukebox, again, preferably for a fee. Other features pertaining to communication between buddies will be described below with reference to FIG. 21D.

FIG. 21D shows illustrative options available for communicating with others in one exemplary embodiment. Preferably, the system can send e-mails and SMS messages to mobile phones. It is to be appreciated that the system should be able to communicate using any variety of communications protocols, such as, for example, via e-mail and/or e-mail-like messages accessible only by the MyTouchTunes system. It also is to be appreciated that bar managers may also send messages (e.g., via e-mail, SMS, etc.) to alert others of events, specials, etc. In step S2042, users may check their inbox. Users may receive messages from, for example, their friends (or buddies), proprietors, and/or advertisers, etc. Buddies may suggest, for example, songs or places to gather. In some exemplary embodiment, buddies may receive instant messages on their mobile phones alerting them of a get-together, a particular user's location in a bar, etc. Furthermore, users may leave messages for others to appear on someone else's next log-in. Proprietors may send messages to their regulars or others they wish to target to alert them, for example, of drink specials, upcoming special events, etc. Advertisers may, based on demographic information provided by users, extend special offers. In step S2044, users may send messages to

others¹ e-mail address, MyTouchTunes inbox, mobile phones, etc. Alternatively or in addition to step S2044, in step S2046, users may broadcast messages to buddies or groups of others, to, for example, alert a particular group of upcoming parties, etc. Thus, exemplary systems can learn about networks of friends through the messages that users send. In addition, the jukebox may directly provide an interface for the features described above, or it may provide for wireless access to the features, for those users that may bring a wireless device with them to the location (or within range) of the jukebox. Some or all of the features can also be provided to the users remotely by, for example, enabling users to access the jukebox services through the internet or other suitable communications network, using for example, the user's home computer. In certain exemplary embodiments, communications may be made to individual users, predefined groups of users (e.g., family, coworkers, teammates, friends, etc.). Also, custom messages may be composed, or more standard messages may be selected (e.g., "Meet us at X Bar," "What are you doing tonight?", etc.). Groups and/or canned messages may be implemented and used regardless of whether the message is sent via email, SMS, IM, etc.

Similarly, certain exemplary embodiments provide systems and/or methods for establishing and managing a registered user's connections. A registered user may seek to establish a connection with another user or prospective contact at a jukebox or remote from a jukebox. In either case, a user may be prompted to log in. After logging in, the user is prompted to enter information about the prospective connection. This may include, for example, an email address, a mobile phone number, a physical address, a jukebox-specific messaging service, and/or the like.

Once the information is entered, a confirmation message may be sent (e.g., via the audiovisual distribution network) to the prospective contact. The confirmation message may identify the registered user (e.g., by name, login name, with a picture or avatar, etc.) and/or provide contact information for the registered user (e.g., a phone number, email address, jukebox login name, etc.). The prospective contact may determine whether a connection between the two should be established, e.g., by verifying that the prospective contact really knows who the registered user is.

In a case where the prospective contact is not already a registered user of the jukebox services, the prospective contact may be requested, and in some cases required, to become a registered user of the jukebox services itself.

After receiving a confirmation message, the prospective contact may choose whether to accept the registered user as a connection. This may be done, for example, by sending a message including an acceptance code or logging into a particular website or jukebox. For example, an email message, text message, or the like may be sent from the prospective contact as a confirmation. The acceptance code may be a particular alphanumeric code uniquely identifying the registered user and the prospective contact, or it simply may be the email address or mobile phone number of the prospective contact (e.g., sending an SMS, email, telephone call, and/or other message with a blank message, a "YES" message, an alphanumeric acceptance code, etc.). Alternatively or in addition, the acceptance code may be embedded in a hyperlink accessible by the prospective contact and/or may be accessible via a computer-based interface (e.g., a website) or jukebox interface. In still other exemplary embodiments, the prospective contact may have to log into a website or jukebox without the use of a preformed hyperlink having an acceptance code embedded therein.

Once the acceptance code is sent from the prospective contact, it may be validated, e.g., by a server in the audiovisual distribution network. The validation process may include determining whether the registered user is valid, whether the prospective contact is valid, etc. If this is successful, a connection between the registered user and the prospective contact may be established.

Once a connection is established between two people, they may communicate with each other via the jukebox and/or jukebox-related services. For example, they may send messages as described above, share credits, etc.

Each user may determine how and/when communications may be received. For example, a user may use an interface provided on a jukebox or remote to the jukebox to limit messages to a single communications medium. For example, a user may wish to only receive messages on a particular email address (even though

another user has provided an alternate email address), in which case messages may be forwarded to the specified email address. In another example, a user may wish to never or only be contacted by SMS messages. Still further, a user may limit the times at which messages may be received, e.g., to prevent messages from being sent too late in the evening or too early in the day (for example, which might otherwise disturb the user). In such cases, the messages may be queued at a jukebox or at a server for later delivery. Users additionally may specify that they will not accept any incoming connections unless specified in an access control list. Thus, it will be appreciated that communications among connections may be managed and/or filtered, for example, to specify when, to where, and from whom they may be received, etc. A registered user can then use the connections list to communicate with the parties in the list using the jukebox or remote device that can access the jukebox network, such as by logging on to the network using a personal computer or the like.

FIG. 23 E shows illustrative options available for accessing value-added content in one exemplary embodiment. Users could access external websites (preferably of partners) as shown in step S2052 for any number of reasons. For example, advertisers may place banner ads pertaining to the particular song or album selected to allow users to place orders for related merchandise. Advertisers may use profile information to target ads, e-mails, text, etc., potentially anywhere (e.g., online, on a jukebox, etc.). For example, jukebox use can be used as a "qualifier" for targeting certain products (either conspicuously or non-conspicuously) to individuals who spend more than a certain amount on jukeboxes yearly. Users also may access external websites to link to a particular band's website, to order sheet music or guitar or bass "tabs," to view lyrics, etc. Also, in step S2054, users may follow a link to download/purchase songs and/or albums from a third-party provider such as iTunes or Amazon.com. In some exemplary embodiments, this step may also be linked to from the "Manage Music" screen (step S2020), etc. Additionally, in step 82056, users may order custom CDs and have them shipped, preferably for a fee. Users could base these orders on their playlists, their PickLists, their buddies' playlists, the HotHits list described above, etc.

It is to be appreciated that although the above-described exemplary MyTouchTunes system has been described in terms of a broad concept for making accounts available on any compatible system, other exemplary embodiments are contemplated where accounts may be made available to smaller subsets of jukeboxes, including, for example, a single jukebox terminal acting in isolation. Furthermore, a large group may act as if it was one system in isolation, or, alternatively, it could have complete knowledge of all user accounts/profiles, locations, etc.

FIG. 22 is an exemplary screen shot of a login page in accordance with one exemplary embodiment. A user may enter an e-mail address in username area 2200 and a password in password area 2202. As a shortcut, a user may select a common e-mail domain from domain list 2204. It is to be appreciated that although this embodiment requires an e-mail address and a password, other embodiments may require a username that need not be an e-mail address. Similarly, it is to be appreciated that a username and password may not be required, for example, on a home jukebox system, and that in yet further embodiments, only a username may be required. A user can use keyboard 2206 to enter the required information, though it is to be appreciated that other input devices, such as a keyboard attached to a home computer, may be used. It also is to be appreciated that the login screen shown in FIG. 22 may appear on an actual jukebox device, on a computer (e.g., through a web-interface, custom software, etc.), on a mobile device, etc.

Preferably, a successful log-on displays a main menu for users. Figure 23 is an exemplary screen shot showing the features available after a recognized user has logged in, in accordance with one exemplary embodiment. A customized greeting (e.g., the user's name) is displayed in area 2300. Playlists 2302, 2304, and 2306 are displayed with descriptive names along with the number of credits required to play a full playlist. For example, to play playlist 2302 costs 7 plays. Preferably, playlists are presumed to be public, which allows other users to view them. Playlists may be designated private, however, as is, for example, playlist 2306. Users can create additional playlists by pressing playlist creator button 2308, and they can edit already existing playlists by pressing playlist manager button 2310. The playlist

manager may allow users to, for example, change existing playlists by adding and/or removing songs, deleting playlists completely, etc.

Users can manage a buddy list by adding a buddy by pressing add buddy button 2314. Alternatively, they can remove a buddy by pressing remove buddy button 2316. Users also can communicate by sending messages to their buddies by pressing send message button 2318. This feature can, in certain embodiments, for example, send SMS messages to mobile devices, send e-mails, send messages to appear when the recipient user next logs-in, etc. In still other embodiments, users can, for example, use this feature to send songs, donate credits, etc. Indicator area 2320 shows that the current user has received one message, and it identifies the sender. In other embodiments, however, other information may be displayed, a more complete inbox (e.g., with folders, forwarding, etc.) may be provided, etc. Credit indicator area 2322 provides account status. In this embodiment, it indicates that the user has previously purchased 20 plays online, and has entered enough money in a local jukebox terminal for 5 additional plays, for a grand-total of 25 potential plays.

Another aspect of the instant invention relating to improvements in background music will now be described. The local server and Tune Central feature described herein, open up a whole new world for background music on a jukebox. Currently, when no paid for music is queued, jukeboxes may play background (ambiance) music with the help of an auxiliary system connected to the jukebox, such as DMX or satellite radio. Alternatively, if they play background music with their own music, they are limited to the songs available for play on the jukebox, typically about 2000 songs. These songs are also available for selection by jukebox patrons. Conventional digital jukeboxes are typically connected to a remote network through the internet for accessing extended music. These applications are therefore not well suited for providing background music services other than the music locally available as this would require constant download or streaming from the remote servers for the background music.

However, jukeboxes equipped with an external server or large music partition (such as Tune Central described herein) can select background music to play from that large pool of songs, even if the song is not available for play through the

standard interface. This permits a vast musical variety for background music using jukeboxes. The background music selection can be made automatically by the jukebox randomly or with an advanced algorithm that selects music based on genres, themes, era or other criteria. Also, background music lists can be preprogrammed by a disk jockey or the like (background playlists) and sent from a central server to jukeboxes. Since music rights can differ for songs available in pay-per-play mode and background music, there is, in an embodiment of the invention, an indicator for each song or song aggregations (e.g., all songs from ABC record label) allowing the jukebox to distinguish between songs that can be played as background music (as opposed to normal jukebox play) based on the availability of the proper music rights, thereby enabling proper royalty accounting as well. In an embodiment of the invention, the jukebox offers advanced options for allowing an operator to configure how background music will be performed, based on genre, themes, indicators (flags), such as billboard hit, Christmas music, playlists (e.g., DJ John - summer 2005), etc. In another embodiment of the invention, the operator configures the jukebox to allow the bartender to further configure the background music mode through, for example, a remote control; much like the bartender would select a radio station in prior background music systems. This mode offers genres or theme based music or playlists for enhancing background music functionality. The operator can, for example, configure which genre, theme or playlists is available to the bartender. In addition, a remote control with programmable keys combined with suitable software can be provided to allow the bartender (or other person) to select (e.g., for a particular zone) a certain background music play list or TV audio input for special event like, for instance, a football game in a bar zone, thereby providing convenient and effective background music services and other services through the jukebox device.

Thus, similar to the background music services, "channels" of music may be provided. A proprietor of a location may select a channel to play. A list of channels may be defined by the location and/or defined by the content provider (e.g., the jukebox provider). In certain exemplary embodiments, channels may correspond to the themes or genres of music already provided to the jukebox (e.g., rap, rock, country, etc.). In addition or in the alternative, the proprietor of the location may

group songs together to form custom channels for or at a particular location or group of locations, and/or shareable between many potentially unrelated or unassociated jukeboxes (e.g., as a sort of public channel). The thousands of songs provided to the jukebox may be played in a coherent grouping as background music, for example, through this channel feature. For example, a predetermined amount of time after no money has been inserted into the jukebox and/or after all songs in the queue have been played, a channel of music (e.g., selected by a proprietor, manager, etc.) may play. The channel may be activated in this illustrative case substantially free from direct user input, for example, so that the switching from normal jukebox operation to background mode is substantially transparent to the normal patrons and/or authorized users (e.g., managers, proprietors, etc.) of the jukebox. Of course, it will be appreciated that authorized users of the jukebox may also cause the jukebox to change from normal operation mode to a channel mode and vice versa, control the switching of channels, choose between channels, etc.

Numerous advantages are provided by this feature. For example, no auxiliary source of media is required to play instances of media provided to the jukebox, thereby reducing the need for, for example, satellite radio, broadcast radio, CD players, and/or other equipment (e.g., in addition to or in place of the jukebox), some of which may require additional costs in terms of hardware and in terms of monthly subscription fees. Rather, a jukebox channel may provide hundreds or even thousands of instances of media playable directly from the jukebox. In this way, numerous different channels of background, music can be provided on the jukebox using the music stored, in the jukebox or available on the network. The available channels can then be selected by an authorized person using, for example, the remote control provided with the jukebox. The various channels could also be displayed on the jukebox on request, so that a channel can be selected from the list.

Furthermore, unlike traditional broadcast and enhanced satellite radio channels, a jukebox patron may exercise control over the content of a particular channel by more than simply tuning to a new channel. For example, an authorized user (e.g., a proprietor, manager, etc.) may use a jukebox remote control to skip songs, replay songs, remove songs from the channel, add songs to the channel, etc.

Currently, this functionality is not possible in connection with broadcast or satellite radio channels. The jukebox channel optionally may learn from this control behavior and, similar to a ratings system, may adapt the channel to more frequently play popular songs, less frequently play skipped songs, etc.

Still further, jukebox channels are not subject to the same kinds of physical problems associated with traditional broadcast and satellite radios. For example, whereas a satellite radio may be inoperable during bad weather conditions, a jukebox channel may function substantially independent of the outside conditions. Similarly, a jukebox channel is not subject to the same kinds of geographical restrictions associated with the antennas of traditional broadcast channels, in that, for example, a jukebox channel may be played wherever there is a jukebox. Indeed, a jukebox channel does not even require the presence of a persistent or establishable internet connection, although it will be appreciated the same may be advantageously used in connection with certain exemplary embodiments where it is desirable to having streaming media, updatable channels, etc.

Thus, certain exemplary embodiments provide systems and/or methods of operating a digital downloading jukebox. A plurality of channels is defined, with each said channel having a plurality of instances of media associated therewith. A channel is selected from the plurality of channels on the part of an authorized user. One or more patron uses of the jukebox is responded to in a first mode, with at least some of said uses including adding instances of media to a queue for play on the jukebox and playing said queued instances of media. When the jukebox is not responding to a patron request and/or playing a queued instance of media, the instances of media in the channel are played in a second mode. The first mode is different from the second mode, and the second mode is not directly controllable by a patron of the jukebox. In certain exemplary embodiment, the second mode may be a background music mode (e.g., such that the music is played at a lower volume, etc.).

In certain exemplary embodiments, the instances of media may be associated with channels by an authorized user and, in addition or in the alternative, the instances of media may be associated with channels according to a theme and/or genre (e.g., rap, rock, country, etc.) associated with each said instance of media.

A frequency an instance of media in a channel is played may be changed in dependence on one or more control codes received from the authorized user. The control codes may correspond to at least one of: skipping an instance of media being played by the channel, replaying an instance of media being or having just been played by the channel, and removing an instance of media from the channel.

Here, as elsewhere, any suitable combination of programmed logic circuitry (e.g., hardware, software, firmware, and/or any suitable combination thereof) may enable such functionality. Additionally, instructions for carrying out the same may be stored on any suitable computer-readable storage medium.

FIG. 24 is a flowchart showing an exemplary process for accessing a system remotely according one exemplary embodiment. In step S2402, a user logs into a jukebox system from a location and/or device different from a jukebox housed locally in a bar. In a preferred embodiment, the user will access the system through a web browser to gain access to an Internet portal. However, it will be appreciated that other mechanisms for accessing the system are possible, such as, for example, via a mobile device (e.g. a cell phone, PDA, etc.), a stand-alone application to be run on a remote computer, an appropriately configured jukebox, etc. Moreover, it will be appreciated that the features and/or services offered by the system may depend on the mechanism and/or device by which it is accessed. For example, some features may be available for an illustrative embodiment that uses a web browser on a personal computer that are not available over a cell phone, and vice versa. Keeping with this example, but still for non-limiting exemplary purposes only, although an illustrative embodiment that uses a web browser on a personal computer may have access to the full set of features and/or services (e.g. music downloads, participation in forums, chat rooms, etc.), an illustrative embodiment that uses a cell phone may only be able to download music and receive text messages. Of course, as various devices become more interconnected and more powerful, it will be appreciated that such distinctions may no longer hold.

Step S2404 determines whether the user is recognized by the system as an existing user. Users not recognized by the system preferably will be prompted to create an account, as in step S2408. Users may be required to supply, for example,

their names, ages, e-mail addresses, etc. Preferably, users will be able to create profiles including, for example, a name and/or nickname, their age, their location, and an associated image. These profiles may be viewable by other users of the system. In some exemplary embodiments, users may be required to register and/or create profiles before participating in some features of the system, for example. Alternatively, in other exemplary embodiments, users may specify that they want their information, or portions thereof, to remain private. A non-limiting exemplary screen shot for editing account information is provided in FIG. 26C, and an accompanying description thereof is given below. After users create their account, preferably a central account database is updated in step S2410. In other exemplary embodiments, a central account database need not be maintained, and the account information may be stored elsewhere, such as, for example, on jukeboxes, on a user's own personal computer, etc.

If users are recognized in step S2404, they may chose a feature and/or service in step S2406, described in more detail in relation to FIGS. 25 and 26A. In some exemplary embodiments, users may have their access to the system limited by the service to which they subscribe. For example, in some example embodiments, users may have free and unlimited, access to the online system from personal computers, but may have to pay premiums to access the services via PDAs. Thus, in a step not shown, even if an existing user is recognized, if they attempt to access the system in an unauthorized way, they will not be permitted to access the system. This feature may, in some exemplary embodiments, also may block unauthorized access by users, registered or unregistered, who attempt to inappropriately access the system and/or its underlying code, songs, etc.

FIG. 25 is a flowchart showing an exemplary process for selecting one or more of a non-limiting list of services and/or features in a system according to one exemplary embodiment. Other features and/or services in addition to, and in combination with, those described below are possible and contemplated herein. FIG. 25 begins with step S2406 as discussed above in connection with FIG. 24. It will be appreciated that in a preferred embodiment, the steps described below need not be selected in any specific order. For example, a user need not search for music before

communicating with other users. It also will be appreciated that the steps described below may, in some exemplary embodiments, be performed alone or in combination. For example, a user may wish to search for music before communicating with other users.

In step S2502, a user may search for music. Preferably, a user may specify search fields, alone or in combination, such as, for example, artist, song, album, genre, etc. An exemplary screen shot showing the results of a search is provided in FIG. 26E and is discussed below. After searching for music, a user may then, for example, add the found songs to a playlist, purchase the songs, or recommend the songs to others. For example, a typical use might include a user remembering a catchy song at a bar but only remembering that it was a reggae song that had something to do with "Zion" from Damien Marley's latest album. Accordingly, a user might, in step S2502, search with the following parameters: artist, "Damien Marley"; song, "Zion" and genre, "reggae." After the search is performed, the user would be able to select "Road to Zion" from a results list. After finding the song, the user might add it to a playlist for convenience the next time the user visits a bar, and the user might recommend the songs to friends. A user might also purchase the single and/or download a digital copy. A particularly bold user might order the entire "Welcome to Jamrock" album.

In step S2504, a user may manage the user's playlists. An exemplary screen shot showing how a user might manage playlists is provided in FIG. 26D and is discussed below. Briefly, a user may, for example, create a new playlist, delete an existing playlist, and/or edit an existing playlist. In a preferred embodiment, a user might also share a playlist, for example, with friends or the public, and a user might also search for publicly available playlists. Preferably, a playlist will have a unique name (e.g. "Joe's All-Time Favorites," "Ghetto Grooves," "Maximum Chill-Out," etc.), and, preferably, a playlist will have a plurality of songs associated therewith. It will be appreciated that a playlist might contain only one song. Additionally, it will be appreciated that in some embodiments users, professional DJs, or others may create and sell playlists, and that users could pay for and download such playlists through the system.

A user may purchase music in step S2506. In one exemplary embodiment, a user may order a single song, a plurality of songs, an entire album, or multiple albums. In some exemplary embodiments, a user may have his order shipped, while in other exemplary embodiments licensed digital copies may be made available. In still other exemplary embodiments, it may be possible to purchase music and download it directly to a media playback device (e.g. an appropriately configured PDA, cell phone, or the like).

Preferably, a user could communicate with others via e-mail and/or text messages in step S2508. Step S2508 preferably also will allow users to participate in online forums and live chat rooms. Messages sent through this step may be delivered to conventional e-mail inboxes, cell phones, etc. Additionally, messages may be displayed on jukeboxes, for example, when recipient users initially log-in, when messages are sent and/or received, etc. It will be appreciated that online forums and chat rooms may use protocols, such as, for example, newsgroups, Network News Transfer Protocol (NNTP), Usenet, Internet Relay Chat (IRC), etc., or they may use custom protocols.

From time-to-time, proprietors or the managers of the system may sponsor contests, promotions, or giveaways. Users may participate in such activities through step S2510. For example, a proprietor might allow anyone who prints a coupon to avoid a cover charge. As another example, there might be a competition for the best playlist. It will be appreciated that these example promotions are for illustrative purposes only, and that other contests, promotions, giveaways, or the like may be possible.

In step S2512, a user may edit that user's profile. The discussions of step S2408 from FIG. 24 above and FIG. 26C below provide additional detail on the type of information preferably associated with a user profile. A user may logout of the system in step S2514. In some exemplary embodiments, a user may be logged out of the system automatically after a predefined period of inactivity.

FIGS. 26A through 26E, which provide sample screen shots, are for exemplary, illustrative purposes only. The specific fields, buttons, graphics, etc. and

locations of the same are in no way intended to be limiting. On the contrary, Figs. 26A through 26E present only one preferred embodiment of the instant invention. It will be appreciated that other arrangements and combinations are possible, and those arrangements and combinations are contemplated herein. It also will be appreciated that a user may navigate through the following screens in any number of ways, and the use of the term "press" indicates only one such implementation. For example, a user may navigate the screens with a mouse, keyboard, touch screen, stylus pen, etc. Preferably, the user interface will be very smooth and streamlined with a limited amount of advertising or other non-core material. Accordingly, when advertisements, surveys, and the like are accessible through the user interface, they preferably will be linked to jukebox functions and or perks tied to their links. For example, an advertiser may reward users clicking on one of its advertisements with credits, free plays, entries in competitions, etc. Similar rewards may be given to users participating in surveys. FIG 26A is an exemplary screen shot showing the features available before a user logs into a system according one exemplary embodiment. A user will log into the system by entering a username and password into username field 2600 and password field 2602, respectively, and then pressing login button 2604. Alternatively, an unregistered user may create an account by pressing register button 2606. Navigation buttons 2608 help the user navigate the system quickly. In this exemplary embodiment, navigation buttons 2608 include a home button, an about button, a FAQ button, and a Find a juke button. The home button returns the user to an introductory screen. The about button preferably displays information about the system, including, for example, help documentation, terms and conditions of use, etc. The FAQ button preferably provides information complementary to that given by the about button and is customized to provide information regarding the most frequently asked questions about the system and its associated services and features.

The Find a juke button preferably enables a user to search for a location with a jukebox. Users may, for example, search for locations near a certain address, in a particular city to which the user may be traveling, locations that offer special features (e.g. drink specials, guest DJs, etc.), locations likely to play music the user likes (e.g.

a bar with line dancing might be appropriate for a country music fan. etc.), etc. Social networks also may be built based on postal codes, schools (e.g. high schools, colleges, universities, etc), etc. In a preferred embodiment, when a user finds a specific location, the user may have custom information, for example, directions, custom maps, hours of operation, parking information, etc. displayed. It will be appreciated that because jukeboxes located at many locations preferably are connected to the system and that users' playlists preferably are accessible by the system, users may receive recommendations likely to fit their tastes. For example, the system might suggest a specific bar to users of a certain age in the same town, who are known to like a specific type of music. In certain example embodiments, the jukebox can recommend that users form certain connections. These suggested connections between patrons (and thus potential "buddies") preferably may be based on information known to the jukebox though not necessarily known to other users. Such information can include, for example, venues frequented, musical tastes, ages, etc. In certain example embodiments, therefore, users may be connected based on where they go, not necessarily where they are from.

Similarly, a jukebox locator service may be provided, and it may exist on a jukebox, via a webpage, through the MyTouchTunes web and/or jukebox interface(s), etc. The jukebox locator service may receive user input and search accordingly. For example, a user may specify an address or portion thereof (e.g., city, state, and/or zip code), may search by name or keyword, or may search for specially enabled jukeboxes. A map may be displayed, with each matching location being highlighted. Additional information may be displayed for a given location. Such additional information may include, for example, the name and address of the location and/or any other details about the location. For example, the songs most frequently played may be displayed, as may be certain promotional offerings, DJs being present, etc.

Additional search criteria may be entered initially or to refine a search. Such additional search criteria may include the user's age, musical tastes, preferred theme (e.g., Irish bar, country line dancing, etc.), etc. Alternatively, the jukebox locator service may conduct a basic search using only the criteria entered by the user and may then highlight recommended locations based on, for example, information

already known about the user, such as, for example, the above-mentioned additional search criteria.

Figure 26F is an exemplary screenshot showing the results of a search with a jukebox locator service. In Figure 26F, search criteria may be entered in search block 2601. The search may be restricted using the exemplary search criteria 2603a-d. A map 2605 of the area may be displayed, with matching locations 2607 being flagged or otherwise distinguished. A user may select a flag. In this example, when flag 2607' is selected, information block 2609 including information pertaining to the selected location is displayed. As noted above, information block 2609 may present the above-noted or other information pertaining to the location.

Users may enter search terms in search field 2610. Additionally, users may further limit their searches by searching only in one of the categories listed in search parameters list 2612. It will be appreciated that the fields of search parameters 2612 are for illustrative purposes only - other fields may be present in addition to, or in place of, those listed, such as, for example, year, popularity, etc. A user may conduct the search by pressing search button 2614. Users may alternatively search by filling in search field 2616 and selecting a search parameter from the list of search parameters in search parameters dropdown list 2618. The results of a search may be displayed according to FIG. 26E, discussed below.

A non-limiting exemplary list of genres is given in genre list 2620. It will be appreciated that other genres may be present in genre list 2620, and it also will be appreciated that, in this exemplary embodiment, additional genres may be accessed by clicking the "More..." button in genre list 2620. A user pressing a specific genre may receive content tailored to that genre. For example, pressing the "Alternative" genre may, for example, present a list of the most popular alternative albums, a list of forums dedicated to the discussion of alternative music, etc.

Most popular albums list 2622 contains information related to the most popular albums and preferably displays the album cover, the artist name, and the name of the album. Most popular album list 2622 may be generated in any number of ways. For example, most popular album list 2622 may be based on preferences of

users of the system, commercial data regarding album sales, album sales within a particular geographic segment, etc. Preferably, most popular album list 2622 may be changed depending on the genre selected from genre list 2620.

Users may access community-specific content by selecting a specific communications method from communications methods 2624. In this exemplary embodiment, users may communicate in groups and/or forums. It will be appreciated that participation in groups and/or forums may require registration, and that groups and/or forums may be designated public or private. Preferably, groups will comprise users, for example, with similar interests, in near locations, etc. For example, there may be a group for so-called "Dead Heads" to allow them to plan meet-up points, transportation, etc. for upcoming tours of the Grateful Dead. Forums may function similarly, though they preferably will be more static, allowing threaded messages between users.

Breadcrumbs trail 2626 provides a convenient way for navigating through the system. Specifically, breadcrumbs trail 2626 traces the path a user takes in accessing a certain page. Pressing a specific part of breadcrumb from breadcrumbs trail 2626 preferably takes the user to a corresponding location.

Information area 2628 and information area 2630 may be customized to provide additional information to the users. In this exemplary embodiment, information area 2628 suggests reasons for using specific features (here, creating custom playlists), and information area 2630 provides useful tips (here, relating to how to create a custom playlist). It will be appreciated that information areas 2628 and 2630 may change depending on, for example, the user, the time of access, the number of times a user has accessed a given page, etc. Furthermore, it will be appreciated that information areas 2628 and 2630 may contain different information not related to use of the system, such as, for example, advertisements, promotions, etc.

FIG. 26B is an exemplary screen shot showing the features available after a user logs into a system according to one exemplary embodiment. Based on the user's profile information, user information 2632 and user icon 2634 may be

displayed. In this exemplary embodiment, user information 2632 comprises a username and a location, and user icon 2634 is a solid square. It will be appreciated that additional and/or alternate information may be displayed, and it also will be appreciated that only information provided by users may be displayed. Users may edit their profile, including user information 2632 and user icon 2634 by pressing edit profile button 2636. Users may log out of the system by pressing logout button 2638. In some exemplary embodiments, users may be logged out automatically after a predefined period of inactivity has elapsed. User playlists 2640 displays the playlists the user has created and/or imported. It will be appreciated, that user playlists 2640 will be empty if a user has not created or imported any playlists. Preferably, the exemplary embodiment will display the unique name specified by the user for each of the playlists. Users may expand one or more specific playlists from user playlists 2640 to show the songs associated with the playlist. In the example shown in FIG. 26B, the "Alternative" playlist is expanded, and four songs 2642 are shown as being associated with the "Alternative" playlist. It will be appreciated that users may expand multiple playlists, and/or collapse lists after they are expanded.

Preferably, the system will display at least one recommendation 2644. The recommendation preferably will include an album cover, a song name, artist information, and keywords associated therewith. The recommendation preferably will be generated automatically, and may be based on a variety of data. The data for generating a recommendation may include, for example, user demographic data (e.g. age, gender, race, etc.), the number and types of songs in the user's playlists, songs the user has purchased through the system, popular songs, etc. Preferably, the system will allow the user to quickly add the recommended song to a playlist by pressing add button 2646.

FIG. 26C is an exemplary screen shot showing how a user might update account information. Stats box 2648 displays basic information that will be publicly available. In this exemplary embodiment, it includes a username, real name, gender, location, and age. However, it will be appreciated that other embodiments may require additional and/or alternate information. It also will be appreciated that users may elect to keep some information (e.g. real name, age, etc.) private.

The username selected in the initial account setup is displayed as username field 2670. Preferably, a user will not be able to change username field 2670 after initial setup, and, preferably, each username will be unique. The user may enter an e-mail address in e-mail address field 2652. In a preferred embodiment, the system will send important e-mails (e.g. terms and conditions information, notices of giveaways, bills, etc.) to this e-mail address. A real name may be specified in name field 2654. Although this exemplary embodiment shows name field 2654 as a single field, it will be appreciated that multiple fields may instead capture the same information (e.g. fields for first and last names; first, middle, and last names; etc.). A user may choose a gender from gender options 2656.

A location may be specified in location field 2658. Although this exemplary embodiment shows location field 2658 as a single field, it will be appreciated that multiple fields may instead capture the same or additional information (e.g. fields for city and state; city, state, and zip code; city, province, and country; etc.). A date of birth may be specified in date of birth field 2660. Although this exemplary embodiment shows date of birth field 2660 as three fields, it will be appreciated that a single field or multiple fields in a different order may instead capture the same or additional information (e.g. one field for month, date, and year; month and year only; month, date, and year; etc.). A user may enter personal or other information in free-form biography field 2662. Additionally, a user may specify a signature line or tag line in forum signature field 2662. Preferably, the system would automatically append the contents of forum signature field 2662 to users' posts to groups and/or forums. A user may upload a picture or icon by entering a file location in picture field 2666. Preferably, this picture would be displayed on login and/or when a user posts a message to a group or forum.

In a preferred embodiment, required fields may be, for example, marked with an asterisk (*), highlighted, or otherwise designated. In other embodiments, a user may additionally store purchase information, such as, for example, credit card information, billing and shipping addresses, etc. This information could be used, for example, if users decide to order or download songs, albums, or other merchandise,

to automatically debit an account so that users could "play now and pay later" while using jukeboxes in bars and/or clubs, etc.

Playlist manager button 2668 provides quick access to the playlist manager form, discussed in more detail below in connection with FIG. 26D. Collapse and expand buttons 2670 allow a user to quickly change the playlist information showing on the far left of the display in this exemplary embodiment. Recommendations 2672 preferably are generated in the manner described above, and clicking one or more recommendation preferably allows a user to associate them with their own playlists quickly and easily.

FIG. 26D is an exemplary screen shot showing how a user might manage playlists. In this exemplary embodiment, a user's custom playlists are displayed horizontally (as playlist labels 2674), and their respective contents are displayed underneath the appropriate headings (as playlist contents 2676). It will be appreciated, of course, that alternate arrangements are possible. Preferably, each playlist has a unique description associated, therewith, and each entry in the playlist displays the number of the song in the playlist, the name of the song, and the artist. Users preferably may edit playlists by "dragging" and "dropping" songs into and/or out of a playlist. Users may use the search function to add a specific song. In some embodiments, removing all of the songs from a given playlist cause the user to be notified that the playlist is empty, and asks the user whether the playlist should be deleted.

A user may create a new playlist by pressing create new playlist button 2678. Preferably, the user will be prompted to enter a unique name for the new playlist. Then the user may edit the playlist in the above-described manner. Users also may share playlists by pressing sharing button 2680. Sharing button 2680 allows users to designate their playlists as public to allow other users to retrieve their contents, preferably allowing users to drag and drop selected playlists into categories, such as, for example, public, private, etc. Correspondingly, users may search for playlists created by other users, prepared by DJs, etc. and then copy and/or import the playlist. Users may delete playlists by pressing delete playlist button 2682 to display list of playlists 2684 and then selecting the playlist to delete from the list of playlists 2684.

FIG. 26E is an exemplary screen shot showing the results of an illustrative search for music. In this exemplary embodiment, a hypothetical user searched for "portis*", where the asterisk (*) indicates a wildcard. Preferably, search results will be grouped by artist (artist list 2686), track/song (track/song list 2688), and/or album (album list 2690). Preferably, each result within a grouping may have a rank 2692 associated therewith. This rank may be based on, for example, the number of users with the artist/track/album in a playlist, the current position on the Billboard (or other) charts, all-time sales, etc. Additionally, in a preferred embodiment, users may be able to sort the results within a category alphabetically or by rank to facilitate the location of a specific result. Because users' attention spans are finite and because of space limitations, the output of the search, preferably will be limited to a few (preferably at least five and preferably no more than ten) results, while more button 2694 will allow the user to browse an expanded list.

FIG. 27 shows an exemplary system for downloading music to a mobile device. A user indicates via mobile device 2702 that the user is interested in downloading a song or songs after logging into jukebox 2703. It will be appreciated that many types of mobile devices 2702 may communicate with jukebox 2703 over many different communications networks. For example, an appropriately configured cell phone may communicate over the data channel of a cell network, a PDA may communicate over standard TCP/IP, etc. Jukebox 2703 communicates over a network 2704 to interface with music repository 2706. Preferably, music repository 2706 will contain a broad range of potential downloads. For some requests, it may be necessary to confirm the download if, for example, there are multiple possible version of a song (e.g. Van Morrison and Jimmy Buffett both have versions of "Brown Eyed Girl," etc.), or for confirmation purposes. After music repository 2706 receives a request, it will select the appropriate song (or songs) and send it back through network 2704 to mobile device 2702.

Although not reflected in FIG. 27, alternate embodiments may send an appropriate song to another designated device. For example, a user at a bar might hear a song and want it to be sent to his home computer. Users in a car might want the song to be available the next time they visit a particular bar, and thus, for example,

the song may be downloaded to a remote jukebox. In some exemplary embodiments, appropriate song 2708 may be sent through a different communications network, as appropriate.

Users may be billed according to billing information they inputted. For example, a stored credit card may be charged whenever a song is downloaded via the jukebox interface. As another example, a user may setup a special jukebox-specific account, linked to, for example, a Pay-Pal account. In a related aspect, users may use their cell phones (or other suitable portable devices) to purchase media, register with the jukebox, etc. by exchanging codes (e.g. via SMS messages) with the jukebox. For example, a user may first identify himself or herself with the jukebox (e.g. may login on the jukebox or remotely through a mobile device). Depending on whether the user's cell phone number (or e-mail address, etc., depending on the implementation), the user may have to enter the appropriate information. An SMS (or appropriately similar) authorization message and/or code may be sent to the user. The user may then enter that code on the jukebox to begin purchasing music, creating personalized content, accessing features available to recognized users, etc. This method of communication also may in some example embodiments enable users to pre-verify and/or pre-pay for music purchases remotely.

FIG. 28 is a flowchart showing an exemplary process for retrieving music. A user might hear a song being played, for example, at a bar, in a club, on the radio, etc., as in step S2802. A user typically decides to purchase a song in step S2804 after hearing it. However, it will be appreciated that a user might choose to download a song without hearing it because, for example, the user's friends recommended it, the user downloads all of the songs by a particular artist, etc.

Using a mobile device, the user logs into the jukebox system in step S2806 and selects the song to purchase in step S2808. The use of a mobile device allows for quick and convenient "impulse" purchases. In some embodiments, the choice of what song to purchase might be automated, for example, by selecting the song currently playing at a given location, the song currently playing on a certain radio station, the song just played at a given location, etc. The user may specify purchase parameters in step S2810, such as, for example, how to pay for the song, the device to which it

should be downloaded, etc. In step S2812, the song is delivered to the user via the jukebox interface. It will be appreciated that the song may be delivered to other devices, such as, for example, a particular jukebox, a portable media device, a laptop, a home computer, etc.

FIG. 29 is a block diagram of an improved downloading digital jukebox system with remote devices accessing a jukebox from within a particular location. A jukebox 16 is located within a particular location, bar, or the like. Users may access jukebox 16 through user interface 24 running thereon. However, users also may access jukebox 16 remotely. Thus, a plurality of mobile devices 26a-26d are shown located in the particular location. Mobile devices 26a-26d are, respectively, a laptop, a PDA, a cell phone, and other personal devices, though it will be appreciated that other properly configured devices may be used. Each mobile device 26a-26d is equipped with a wireless transmitter 27a-27d, respectively. Mobile devices 26a-26d preferably allow registered users to effectively logon to jukebox devices and access specific content, such as, for example, customized playlists, personalized screens, messages from other recognized users, etc. Preferably, a change made by one of the mobile devices 26a-26d (e.g. to user settings, playlist setup, etc.) will be reflected by all of the mobile devices 26a-26d. For example, if a user creates a new playlist, edits an existing playlist, changes a password, etc. on a jukebox via user interface 24, user interfaces 24a-24f will reflect that change.

Mobile devices 26a-26d may communicate through an external network to communicate with jukebox 16 having communicator 25. It will be appreciated that mobile devices 26a-26d may communicate over a LAN, wireless Internet, Bluetooth, or any other suitable communications network.

Preferably a central database of recognized user information is maintained and accessible by each of the jukebox devices 16, 16a-f and remote devices 26a-d. However, in some exemplary embodiments, local databases of recognized user information may be maintained on devices. Preferably, the devices will communicate with each other through a communications network, such as, for example, the Internet. However, it will be appreciated that other communications

methods are possible, such as, for example, through wired communications over a LAN, wireless communications, etc.

FIGS. 29A-29E provide additional, non-limiting exemplary configurations that remotely access jukeboxes. It will be appreciated that other variations on and combinations of these exemplary configurations are possible and contemplated herein.

FIG. 29A is a block diagram of an improved downloading digital jukebox system with remote devices accessing a jukebox from within a particular location. In this exemplary embodiment, jukebox 16 and the plurality of remote devices are located within a particular location. Jukebox 16 includes a user interface 24 that allows jukebox users to, for example, make selections, create profiles, edit playlists, etc. Jukebox 16 also includes a wireless communications device 25. Remote devices 26a-26b communicate with jukebox 16 through their respective wireless communication devices 27a-c. In this example embodiment, remote devices 27a-c are, respectively, a laptop, a PDA, and a cell phone. It will be appreciated that other remote devices may communicate with jukebox 16. It also will be appreciated that remote devices according to this system may operably communicate with jukebox 16 from outside the particular location, with the range being determined by the hardware on jukebox 16 and remote devices 27a-c and the protocols over which they communicate. Thus, for example, wireless communications over 802.11g connections may allow for users to access jukebox 16 from within a location, tables setup outside the location, and other areas immediately surrounding the area.

Remote devices 26a-c allow users to login to jukebox 16 remotely, without having to access jukebox 16 and user interface 24 directly. Thus, a user can, for example, play songs, edit playlists, and perform other jukebox-related activities without using user interface 24 directly. It will be appreciated that remote devices 27a-c may have their own user interfaces, which may be the same as or different from user interface 24. User profile information may be stored locally on jukebox 16, on a remote server (not pictured), or on a remote device 27. Preferably, a change made by a remote device (e.g. to user profile information, play list contents, purchased media, etc.) would be mirrored on jukebox 16, any other associated remote devices, and/or on other jukeboxes, etc.

FIG. 29B is a block diagram of an improved downloading digital jukebox system with remote devices outside of a particular location accessing the digital jukebox system. In this exemplary embodiment, jukebox 16 with local server 22 and user interface 24 is connected to a network 14. Unlike remote devices 26a-26c which connect directly with jukebox 16 in FIG. 29A, these remote devices 26a-26c also are connected to network 14. Thus, users may login to jukebox 16 over a potentially broad area. For example, remote devices 26a-26c may connect with jukebox 16 over LAN, WAN, Internet connection, or the like. User profile information may be stored on local server 22. Again, preferably, a change made by a remote device (e.g. to user profile information, playlist contents, purchased media, etc.) would be mirrored on jukebox 16, any other associated remote devices, and/or on other jukeboxes, etc.

FIG. 29C is another block diagram of an improved downloading digital jukebox system with remote devices outside of a particular location accessing the digital jukebox system. FIG. 29C is like FIG. 29B, in that a plurality of remote devices 26a-26c are connected to jukebox 16 over network 14. However, FIG. 29C includes a database 29 connected to network 14. Database 29 may store, for example, user profile information, users' playlist definitions, etc. This configuration may be advantageous in some exemplary implementations because a plurality of jukeboxes connected to network 14 may all access the central database 29 without contacting individual jukeboxes or individual remote devices, which may not always be available.

FIG. 29D is a block diagram of an improved downloading digital jukebox system with another downloading digital jukebox system outside of a particular location accessing the first digital jukebox system. In this exemplary embodiment, a jukebox 16a with a local server 22a and a user interface 24a are located within a first location, and a jukebox 16b with a local server 22b and a user interface 24b are located within a second location. It will be appreciated that the locations may be different rooms within a single bar, two separate establishments, etc. Both jukebox 16a and jukebox 16b are connected via network 14. User profile information may be stored at one or both of local server 22a and 22b. In some exemplary embodiments, if user profile information is stored on only one local server, a user logging-in to one

jukebox preferably would be able to access information stored on the other local server. In other exemplary embodiments, if user profile information is stored on both local servers, preferably any changes made on one jukebox would be mirrored on the other. In still other exemplary embodiments, user profile information may be retrieved from a first jukebox and stored to a second jukebox only when a user accessed the second jukebox. These configurations are advantageous because they do not distribute user information to areas where users do not access their information. For example, a user who travels from Washington to Los Angeles on business would be able to access that user's particular information created in Washington by logging-in to a jukebox in Los Angeles; however, because that hypothetical user has not logged-in to a jukebox in Montreal, jukeboxes there would not necessarily have the user's information stored locally.

FIG. 29E is another block diagram of an improved downloading digital jukebox system with another downloading digital jukebox system outside of a particular location accessing the first digital jukebox system. FIG. 29E is like FIG. 29D, in that a jukebox 16a is connected to a jukebox 16b over network 14. However, FIG. 29E includes a database 29 connected to network 14. Database 29 may store, for example, user profile information, users' playlist definitions, etc. This configuration may be advantageous in some exemplary implementations because a plurality of jukeboxes connected to network 14 may all access the central database 29 without contacting individual jukeboxes or individual remote devices, which may not always be available. It will be appreciated that a plurality of jukeboxes may be attached through network 14.

Fig. 30 is an exemplary flowchart showing one way in which users may change their status by becoming "verified users." In step S3002, a potential user creates a jukebox account to become a registered user. As described above, the potential user may create the account on a jukebox at a location, or by remotely accessing a jukebox by, for example, using a personal computer, laptop, mobile device, etc. It will be appreciated that potential users who unexpectedly or spontaneously walk in on a get-together for verified users may wish to become verified users immediately and create their account with little or no prior planning. Registered users receive notifications of get-togethers, special events, and

the like in step S3004. Such notifications may be sent via e-mail, SMS text message, through a message displayed when a registered user logs onto a jukebox locally, etc. In certain preferred embodiments, registered users will be informed of get-togethers only if they meet predefined qualifications. For example, all registered users living in the Boroughs of New York may receive a notice of a get-together in Manhattan, but a registered user in Des Moines may not. As another example, users who specified they like Metal Music may not receive notices of Go-Go get-togethers.

After a registered user receives a notification regarding a get-together, the registered user may participate in that event, as in step S3006. This step might involve attending a bar, event, social event, or the like. In step S3008, the registered user becomes a verified user. This step might require the user to log onto a jukebox during the get-together. Alternatively, registered users might receive vouchers, codes, or the like to later enter into a jukebox (locally or remotely) to change their status to verified. In certain other embodiments, registered users may drop their business cards, sign lists, or otherwise indicate their presence to have their status changed for them. Thus, In step S3010, a user may have privileges granted, access rights changed, etc. based on verified status. For example, a verified user may be gain access to certain groups/discussion forums for verified users only. Verified users also may, in certain exemplary embodiments, make un-moderated postings. In certain other exemplary embodiments, verified users may be able to access special promotions not available to other users. One such special promotion may include the ability to send messages (e.g., SMS messages, email messages, picture messages, etc.) to registered users' mobile phones. In certain exemplary embodiments, the coupons for one or more free credits may be sent to registered users. The coupon itself may come in many forms. For example, a simple message may be provided to show to an authorized person at a location (e.g., a bar manager, proprietor, etc.). The coupon may also include a special validation code, for example, to help reduce the chances of free credits being given away erroneously. The validation code could be shown to an authorized person or could be entered directly into a jukebox. The code could be validated against a database of codes located on the jukebox or remote from the jukebox. The database could be updated (e.g., the validity of the coupon may be revoked so as to prevent

further use(s) of the same coupon, etc.) after a coupon is read. Alternatively or in addition, the code may be created according to a self-checkable format such that a database does not need to be consulted (e.g., the code itself may be encoded with the number of credits and/or additional information according to a predefined format which may or may not be readily determinable by a casual observer).

The validation code may also include a location code that indicates that the coupon is only good for a particular location or group of locations. As an example, a coupon may be provided for plays on all jukeboxes at all locations owned by a single person. Similarly, the validation code may also include a time expiry code, such as, for example, a number of days or an absolute date and/or time in which the recipient of the coupon must redeem the coupon. For example, the code may be good for use within 1 week, only on certain days, etc.

Separate from or in addition to a simple message and/or validation code, the coupon may be provided in the form of a barcode scannable by the manager and/or at the jukebox using appropriate hardware. The coupons may be delivered, for example, as text messages, through the regular mail, through email, etc. In certain exemplary cases, whether the coupon includes a barcode or in another format, it may be necessary to print out a coupon for redemption at a location, whereas in certain other exemplary cases it may be possible to merely bring the mobile device that has received the coupon.

Thus, certain exemplary embodiments provide systems and/or methods of delivering a coupon to a user of a digital downloading jukebox. A validation code is created. Associating with the validation code is a number of free credits for the user. Note that no specific number necessarily need be included in the validation code - for example, the existence of the coupon itself may be good for a predetermined number of credits (e.g., 1 credit, 5 credits, 10 credits, etc.). The validation code is included with the coupon. The validation code is transmitted to the user.

Included in the validation code may be at least one location where the user can redeem the coupon and/or a restriction on the time when the user can redeem the coupon. The validation code may include an encoded, alphanumeric sequence and/or

a barcode. The coupon feature could also be used for coupons that provide discounts on food, drink and/or other products/services offered by a jukebox establishment.

In certain exemplary embodiments, the validation code may be obtained from the coupon. It may then be determined whether the validation code is valid. When the validation code is valid, a number of credits is/are provided to the user in accordance with the number of free credits associated with the validation code. The validation code may be obtained by the jukebox via a barcode scanner, and/or a user may have to input the validation code into the jukebox via a user interface provided to the jukebox. Alternatively or in addition, an authorized user of a jukebox (e.g., a manager, proprietor, etc.) may provide the free credits to the patron.

Here, as elsewhere, any suitable combination of programmed logic circuitry (e.g., hardware, software, firmware, and/or any suitable combination thereof) may enable such functionality. Additionally, instructions for carrying out the same may be stored on any suitable computer-readable storage medium.

FIG. 31 is an exemplary flowchart showing how registered users may receive credits for referring potential users. In step S3102, a user logs onto a jukebox (either locally or remotely). In step S3104, users specify some contact information for potential future users. It will be appreciated that step S3104 may be accessed when users create new accounts, when registered users access a form, etc. It also will be appreciated that contact information may be used, as appropriate, to send, for example, an e-mail, a text message, conventional mail, etc. In step S3106, invitations are sent to potential future users by the jukebox using the provided contact information. Lastly, in step S3108, referring users may receive credits. In certain example embodiments, credits may be awarded based on the number of users referred. In certain example embodiments, credits may be awarded based on a formula (e.g. 1 credit for every five users referred and/or registered), which might include bonuses (e.g. a bonus may be given when 10, 25, and/or 50 referred users become registered users).

FIG. 32 is an exemplary flowchart showing how registered uses may receive credits for creating public playlists. In step S3202, a user logs onto a jukebox, either

locally or remotely. In step S3204, a user creates a public playlist. As described above, a user will associate certain instances of media with a playlist, preferably giving the entire playlist a descriptive name. It will be appreciated that users may create playlists using any of the methods described above, including, for example, drag-and-drop features, search features, etc. In certain exemplary embodiments, it may be necessary to designate playlists as public.

In step 3206, jukebox users may access public playlists and songs as part of their normal jukebox activities. The jukebox monitors user activity, especially with respect to the songs and playlists accessed. In certain exemplary embodiments, the jukebox will maintain counts, percentages, and the like related to the songs and playlists accessed. The jukebox will reward users based on the accessions of other users in step S3210. For example, users may be rewarded if their playlist(s) is/are accessed the most of any playlist, is/are accessed above a certain threshold number of times, is/are selected more than other playlists in a predefined category, etc. In certain exemplary embodiments, users may be rewarded based on the contents of their playlists. For example, users may be rewarded based on whether their playlists contain a predefined number and/or proportion of the most popular songs, as determined by the jukebox. It will be appreciated that users may be rewarded in any number of ways. For example, one obvious way users may be rewarded is by giving them credits for additional plays on the jukebox. Users also may be rewarded by being entered in certain promotions, giveaways, or the like. More generally, a recognized user may earn reward points for conducting arbitrary jukebox-related activities (e.g., on an actual jukebox, via a web interface, etc.). For example, a recognized user may earn points every time the user logs on, sends a message, shares a playlist, etc. Reward points may be redeemed, for example, to purchase plays on a jukebox. Alternatively, points may be redeemed for special merchandise to promote the further use of the jukebox by the recognized user.

Optionally, a user may choose an avatar to represent the user on a jukebox

and/or in any other jukebox-mediated environment. A user's avatar may be depicted graphically, for example, as a picture, as a moving animation, as a character,

or in any other way suitable for identifying the user. The avatar may or may not bear a physical likeness to the registered user.

In one example, when a person operating a jukebox selects the local playlists option from a jukebox, the playlists for each registered user on the jukebox may be made available. This example is depicted in Figure 33, which is an exemplary screenshot of a "Local Playlists" screen in which registered users are represented by avatars. Next to the registered users' names, nicknames, or other identifying marks 3302a-e may be the particular user's avatar 3304a-e. This may provide further customization and personalization to a jukebox by, for example, showing the regulars of a bar whom many patrons may know or recognize, encourage casual jukebox patrons to become more active in creating custom playlists, etc. After the user selects a group of local playlists associated with a registered user and represented by an avatar, that user's playlists may be displayed. The user operating the jukebox may then select an individual playlist associated with the user represented by the avatar. Once selected, the contents of the playlist (e.g., one or more songs) may be displayed, along with the user's avatar, the name of the playlist, and the number of plays required for play.

If the playlist, or a portion thereof, is selected for play by the user, a now playing screen may be displayed, as shown in Figure 34, which is an exemplary screenshot of a "now playing" screen. This screen may display the avatar 3304a, the name of the registered user 3402, the name of the playlist 3404, etc. Additionally, acting on behalf of the registered user, a lead-in audio or video may be played to introduce the songs, transition between songs, and/or conclude the playlist. For example, the avatar itself may be animated. When the avatar is a picture, a slide show may cause the image appear to be animated, or various images may be superimposed over the mouth to make it appear as if the avatar is speaking or singing. The same or similar functionality may be made available if the avatar is an animated character. For example, an avatar may sing and/or dance along with the music (e.g., the avatar may head-bang to heavy metal music, line dance with other real or simulated avatars to country music, etc., as programmed by the registered user or as automatically generated by the jukebox), using, for example, custom animations,

inverse kinematics techniques, etc. The and/or other features may be enabled by any suitable avatar action programmed logic circuitry (e.g., a processor on or accessible by a jukebox or remote jukebox interface comprising any suitable combination of hardware, software, or the like). In general, the avatar action programmed, logic circuitry may cause the avatar to react to the instance of media, before, during, and/or after it is played.

These and/or similar features may be provided by allowing the registered user to upload content (e.g., video and/or audio content) to the jukebox and/or jukebox distribution network. For example, pictures, audio files, movies, and the like may be captured by and/or stored to a single jukebox and associated with a registered user on that jukebox, or may be stored to a central location so as to be available to multiple jukeboxes.

It will be appreciated that these and/or similar features may be provided in a networked jukebox environment so that registered users may have access to their avatars from anywhere. This may be useful when traveling, for example, back to a hometown, university, and/or other place that the user may visit later, for example. In this way, the avatar may be familiar to other patrons who once knew the registered user. It also may be helpful to registered users on the go, who wish to be creative (e.g., to make new friends) or to introduce themselves (e.g., through introductory audio and/or video clips), while also having access to the user's own library of music, etc.

Avatars and media presented by avatars may be forced to comply with local filters applied to the jukebox. For example, playlists may be filtered if a location does not permit rap, pre-recorded messages may be edited for explicit language, inappropriate pictures may be altered, etc.

As described above with reference to the MyTouchTunes services, suitably configured jukeboxes may facilitate users' connections with various virtual communities, including, for example, communities of interest (e.g., organized around a specific subject), communities of fantasies (e.g., around gaming, creative environments, or the like), and/or communities of relationships (e.g., online meeting spaces). Accordingly, the jukeboxes may provide - either directly or through an

online portal -one or more of electronic mailing lists; online chat rooms; bulletin boards, message boards, discussion groups, Internet forums, or the like; blogs; instant messaging services; podcasts; text messaging services; file sharing (e.g., P2P file sharing); multi-user dungeons (MUDs); etc. By connecting with and establishing new jukebox-centered virtual communities, it may be possible to attract new patrons and develop further interest of existing patrons. Indeed, it may be possible to realize increased returns in and through the virtual community by gathering information about members (and targeting advertising and transaction offerings appropriately), drawing vendors to the community to attract members and promote spending, drawing members to the community and generating member-based content, and/or promoting user-to-user interactions to build user loyalty to the community.

The above-described avatars may be used connection with a jukebox-mediated virtual community. For example, avatars may be associated with all actions a user takes within the community (e.g., writing a blog, posting on a bulletin board, sending a message, etc.). As a further, avatars may become more interesting as users begin to create and share their own media. A user may upload music (e.g., pursuant to a standard license agreement), with the avatar being virtually "attached" to or associated with the media. The jukebox-mediated network could track the distribution of the music to websites, P2P file sharing engines, and/or jukeboxes. Royalties may be tracked and returned to the user through its avatar. Additionally, based on the type of user, different fees/royalty structures may be provided. For example, for independent artists, a large fee and small royalty may not be appropriate, whereas an artist associated with a major record label may be more able to bear such direct and indirect costs. To facilitate control, music may be checked for copyright compliance, and this authorization process may be performed automatically (e.g., a comparison to banks of known music, checking for the presence of a digital watermark, etc.) or manually (e.g., by an authorized user, patron, operator, employee of the central distribution network, etc.). When content is posted and/or agreements are ratified, an avatar may be an acceptable form of digital signature attached to a copyright and/or license agreement. An avatar also may be displayed as album art, during the actual playing of a tagged song, etc.

In addition to the above-described toolkit usable by operators, operators, authorized, users, and/or proprietors also may make use of a configuration dashboard to administer, manage, update, and/or otherwise control jukeboxes. Indeed, operators managing one or more jukeboxes may save time and/or money by remotely accessing diagnostic and/or management features provided to the jukebox. In certain exemplary embodiments, an operator may be able to access a management and/or diagnostics module of the dashboard using, for example, a computer, handheld device, or other suitable device to receive notifications and/or to inquire into the status of a jukebox through a proprietary (e.g., web-based) interface. Communications may be bidirectional, thus allowing operators to directly instruct and/or query jukeboxes of interest regarding management features and/or alerts and/or notifications. The features described below may be used alone or in various combinations to allow operators, authorized users, and/or proprietors to focus on particular jukeboxes in need of attention, thus saving time and money, while also increasing the up-time, reliability, and overall appeal of jukeboxes. With respect to such features, each jukebox may gather and/or monitor information in which an operator would be interested, potentially proactively sending this information to the operator. A notification mechanism may send messages (e.g., via email, SMS, or other suitable format) to the operator automatically when predetermined thresholds are exceeded. These predetermined thresholds may be set by the operator, may be preconfigured, etc.

Diagnostics and/alerts may include, for example, the bill rejection rate (e.g., in terms of percentage and/or number), number of bills in the stack, number of reboots, temperature (e.g., hard drive, CPU, system, enclosure, etc.), and the like. Additionally, such alerts/notifications may be set to be sent at a predetermined time. For example, notifications may be sent when the CPU temperature exceeds a threshold for a specified amount of time, when the number of reboots exceeds a threshold within a specified amount of time, etc.

It will be appreciated that such diagnostic techniques may be made available to operators located remote from a jukebox, as well as directly at a jukebox. Furthermore, in certain exemplary embodiments, the diagnostic module of a first

jukebox may be used to remotely check the diagnostic module of a second jukebox, the second jukebox being located remote from the first jukebox.

Figures 35-37 are illustrative screenshots demonstrating how diagnostic alerts may be configured. In particular, Figure 35 is an exemplary screenshot for setting alert preferences. In Figure 35, an email address to which alerts may be sent is specified in block 3502. Weekly reports may be specified by selecting a day from the dropdown box 3504, and daily reports similarly may be sent at a time specified by dropdown box 3506. The bottom portion of the exemplary screenshot lists the various tests and the frequency at which emails should be sent. For example, a selection from dropdown box 3508 may specify whether hard, disk temperature alerts should be sent never, instantly-daily, or weekly. Other alerts may be sent at the same or different intervals by, for example, setting dropdown boxes 3510, 3512 (which are partially blocked by the expanded dropdown box 3508). Of course, it will be appreciated that other intervals may be specified, other tests may be monitored, and that other addresses or notification means may be used (e.g., SMS messages, instant messages, etc.).

Figure 36 is an exemplary screenshot for setting alert thresholds. The left-hand side of Figure 36 notes the particular statistic to be monitored (e.g., bill reject rate, bill stacker fill rate, number of reboots, etc.), while the right-hand side allows the threshold to be specified. The threshold may be set in terms of an absolute number and/or in terms of a percent, as necessary. It will be appreciated that other tests may be monitored, and that any suitable threshold may be specified. Additionally, indicating no threshold may be used to avoid causing alerts to be generated for a particular test, as well. Figure 37 is an exemplary screenshot showing the status of various tests. In Figure 37, the left-hand side of the screen lists the particular test, while the right-hand side of the screen indicates its status. The tests optionally may be categorized for convenience (e.g., temperature, fee collection, software operation, etc.). As shown in Figure 37, status may be indicated by a color (e.g., green may signify no problem or operation within an acceptable range, yellow may signal some cause for concern, and red may signal the occurrence

or imminence of a problem). Of course, other indicators may be used instead of a color-coded scheme (e.g., a number or percentage system, etc.).

An enhanced scheduling module also may be provided to the jukebox. This module may allow operators and/or other authorized jukebox users to schedule single and/or recurring events. For example, a bar owner may schedule morphs, music filters, and/or background music. The event recurrences may be weekly, monthly, yearly, etc. For example, background music may be played during lunch, music filters may be applied during the afternoon and early evening, and an unrestricted playlist may be made available at night. As another example, the jukebox may morph nightly to accommodate an 80s music night, hip-hop night, country line dancing night, disco night, etc. Such advance setup ultimately may save time and improve convenience. The events scheduler may include monthly and/or daily views, with the daily view being broken down, for example, by hours. After a new event is added, the appropriate views may be updated.

Figure 38 is an exemplary screenshot of the morphing capabilities being linked to the enhanced scheduling module. With respect to morphing capabilities, a jukebox may be morphed to a predefined master jukebox (e.g., via input area 3802), to a specific jukebox (e.g., a jukebox previously defined by, for example, a proprietor or operator, etc., via input area 3804), or to a new custom list of music (e.g., via input area 3806). In particular, a snapshot may be taken for each morph of the jukebox, with associated data (e.g., date, time, playlist) being stored therewith. This may facilitate the morph process and also may allow proprietors to "undo" morphs (e.g., when they are less popular, not as successful, etc.). A user may then schedule the morph like any other event by selecting the "schedule this setting" button 3808.

The "schedule this setting" button 3808 may bring up the calendar scheduling component, just as in any other event scheduling. Figure 39 is an exemplary screenshot of a scheduling calendar. The scheduling calendar may have a month-by-month view 3902 as well as a daily view 3904. The daily view 3904 may be broken down by the half-hour, hour, etc., and may show any other events scheduled for a particular day.

Operators also may setup sub-user accounts, with such sub-users having restricted location access.

Other functions may include enhanced location setup features.

The present disclosure has used certain terms that should not be interpreted as limiting the invention to a particular embodiment, hardware components and configurations, software configurations, etc. For example, many features and examples have been described in relation to their existence within a bar, pub, or other environment. However, it will be appreciated that the features present in the exemplary embodiments of the present invention are adaptable for use in any location where a jukebox (or multiple jukeboxes) may be located. Similarly, while certain features and functions are described with reference to usage by "users," "owners," "operators," "patrons," etc., it will be appreciated that these terms are generic and may, in most cases, be used interchangeably depending on the embodiment chosen and the feature employed. For example, while it may be advantageous to limit the initial song selection to owners and/or operators, in certain exemplary embodiments, patrons may play a role in the initial song selection.

Still further, particular hardware combinations and configurations are disclosed which represent only one way which the embodiments may be constructed. Central servers may, in some exemplary embodiments, be comprised of one or more servers acting together or separately to coherently provide the full range of services necessary to enable a functioning jukebox. For example, a cluster of servers may comprise a virtual central server, with one server providing media, another tracking membership, still another processing licensing, etc.

Similarly, the local servers described herein may be incorporated into the jukeboxes. For example, the local servers may appear to function independently, even though they exist as part (e.g. partition) of an integrated mass storage device within the jukebox. Indeed, as hard disks become larger and less expensive, they may preferably serve the functions of local servers. One presently preferred embodiment comprises a 120 GB hard drive containing 25,000 songs as the factory drive. However, flash media or any storage media now known or later developed (or

combinations thereof) may be used as the factory drive, expanded media storage or local server, depending on the desires of a system designer for a particular jukebox application. In other words, the invention is not limited to any particular storage technology or configuration, but instead can be implemented using any suitable storage technology.

Also, although the term "song" has been used sometimes in the above-description, this term is not intended to be limiting to the scope of the invention, and any instance or instances of media (e.g., song, video, song/video combination, data, information etc.) can be used in any embodiment herein and still fall within the intended scope of the invention.

Lastly, it will be appreciated that the screen shots and software arrangements presented herein are only one preferred method for organizing and displaying the features disclosed herein. Other configurations are possible and are therefore contemplated herein.

While the preferred aspects of the invention have been illustrated and described herein, it will be apparent to one of ordinary skill in the art that various changes and/or modifications can be made. Thus, the specific description herein is meant to be exemplary only and is not intended to limit the invention beyond the terms of appended claims.

1. A method of distributing a playlist to one or more digital jukeboxes connected to an audiovisual distribution network, the method comprising:
connecting a peripheral device having a peripheral playlist stored in a computer-readable storage medium thereon to a digital jukebox via a connector;
retrieving the peripheral playlist for the digital jukebox;
converting the peripheral playlist to a jukebox playlist; and
publishing the jukebox playlist at least on the jukebox,
2. The method of claim 1, wherein peripheral device is at least one of: a portable music player, a computer, and a mobile phone.
3. The method of claim 1, further comprising at least temporarily storing the peripheral playlist to a computer-readable storage medium of the digital jukebox.
4. The method of claim 1, further comprising determining whether the jukebox playlist is to be designated a public jukebox playlist or a private jukebox playlist.
5. The method of claim 1, further comprising sending the jukebox playlist to a server of the audiovisual distribution network.
6. The method of claim 5, further comprising making the jukebox playlist available to jukeboxes connected to the audiovisual distribution network.
7. The method of claim 5, further comprising making the jukebox playlist available to remote devices connected to the audiovisual distribution network.
8. The method of claim 1, wherein the connector is a USB connection.
9. The method of claim 1, further comprising requiring a user to log into the jukebox before importing of the peripheral playlist is enabled.

10. A method of distributing a playlist to one or more digital jukeboxes connected to an audiovisual distribution network, the method comprising:

retrieving a software application playlist associated with a software application used on a computer device for use on a the digital jukebox;

converting the software application playlist to a jukebox playlist; and

publishing the jukebox playlist at least on the jukebox.

11. The method of claim 10, further comprising at least temporarily storing the software application playlist to a computer-readable storage medium of the computer device.

12. The method of claim 10, further comprising determining whether the jukebox playlist is to be designated a public jukebox playlist or a private jukebox playlist.

13. The method of claim 10, further comprising sending the jukebox playlist to a server of the audiovisual distribution network.

14. The method of claim 10, further comprising sending the jukebox playlist to a server of the audiovisual distribution network.

15. The method of claim 14, further comprising making the jukebox playlist available to jukeboxes connected to the audiovisual distribution network.

16. The method of claim 14, further comprising making the jukebox playlist available to remote devices connected to the audiovisual distribution network.

17. The method of claim. 10, further, comprising requiring a user to log into the computer device before importing of the software application playlist is enabled.

18. A digital downloading jukebox, comprising:

a communicator configured to distribute a playlist to one or more different digital jukeboxes connected to an audiovisual distribution network;

a connector configured to connect a peripheral device having a peripheral playlist stored in a computer-readable storage medium thereon to the digital jukebox;

a processor configured to retrieve the peripheral playlist for the digital jukebox; and,

converting programmed logic circuitry configured to convert the peripheral playlist. to a jukebox playlist,

wherein the processor is further configured to publish the jukebox playlist at least on the jukebox.

19. The jukebox of claim 18, wherein peripheral device is at least one of: a portable music player, a computer, and a mobile phone.

20. The jukebox of claim 18, further comprising a computer-readable storage medium associated with the digital jukebox configured to at least temporarily store the peripheral playlist.

21. The jukebox of claim 18, wherein the processor is further configured to determine whether the jukebox playlist is to be designated a public jukebox playlist or a private jukebox playlist.

22. The jukebox of claim 18, wherein the communicator is configured to send the jukebox playlist to a server of the audiovisual distribution network.

23. The jukebox of claim 22, wherein a module of the server on the audiovisual distribution network is configured to make the jukebox playlist available to jukeboxes connected to the audiovisual distribution network.

24. The jukebox of claim 22, wherein a module of the server on the audiovisual distribution network is configured to make the jukebox playlist available to remote devices connected to the audiovisual distribution network

25. The jukebox of claim 18, wherein the connector is a USB connection.

26. The jukebox of claim 18, wherein the connector is a proprietary docking station.

27. The jukebox of claim 18, wherein the processor is further configured

to require a user to log into the jukebox before importing of the peripheral playlist.

28. A method for playing instances of media on a digital downloading jukebox, the method comprising:

requiring a user to log into the jukebox;

providing a plurality of selectable instances of media for play on the jukebox;

enabling user selection of an instance of media from the plurality of selectable instances of media via a user interface to the jukebox;

for each user selection, adding the instance of media to a playlist stored in a computer-readable storage medium of the jukebox; and

playing the instances of media in the playlist in order;

wherein the plurality of selectable instances of media is selected from a master group of instances of media in dependence on a previous play of an instance of media by the user.

29. The method of claim 28, further comprising forming the plurality of selectable instances of media from instances of media previously played by the user within a predetermined time period.

30. The method of claim 28, further comprising forming the plurality of selectable instances of media from instances of media previously played by the user at a predetermined location.

31. The method of claim 28, further comprising forming the plurality of selectable instances of media from instances of media stored in a list of recently played instances of media associated with the user.

32. The method of claim 28, further comprising:

determining a number of credits available to the user;

when the number of credits is sufficient to play all of the instances of media in the playlist, playing all instances of media in the playlist; and

when the number of credits is insufficient to play all of the instances of media in the playlist, playing only a portion of the instances of media in the playlist.

33. The method of claim 32, further comprising when the number of credits is insufficient to play all of the instances of media in the playlist, allowing the user to select the portion of the instances of media in the playlist for play.

34. The method of claim 32, further comprising, when the number of credits is insufficient to play all of the instances of media in the playlist, playing the instances of media in the playlist in order until the credits are exhausted.

35. A method of playing an instance of media on a digital downloading jukebox, the method comprising:

providing a plurality of instances of media on the jukebox;

providing an interface to a user of the jukebox to allow the user to search for a particular instance of media for play on the jukebox;

determining whether the instance of media is available for immediate play on the jukebox free from additional processing by the jukebox;

when the instance of media is available for immediate play, charging the user a predetermined first number of credits and queuing up the instance of media for play; and

when the instance of media is not available for immediate play,

determining whether the user is a verified user,

when the user is a verified user, charging a second predetermined number of credits if the instance of media exists in a playlist associated with the user, but otherwise charging a third predetermined number of credits, and

queuing up the instance of media for play,

wherein the second predetermined number of credits is less than the third predetermined number of credits, and

wherein the third predetermined number of credits is greater than the first number of predetermined credits.

36. The method of claim 35, wherein the playlist associated with the user is a playlist previously created by the user.

37. The method of claim 35, wherein the playlist associated with the user

is a playlist including songs recently played by the user.

38. The method of claim 35, wherein the playlist associated with the user is a playlist created by another user but recently played at least in part by the user.

39. The method of claim 35, wherein the determination of whether the instance of media is available for immediate play on the jukebox free from additional processing by the jukebox includes determining whether the instance of media resides on a storage area separate from a dedicated storage area of the jukebox comprising immediately playable instances of media.

40. The method of claim 39, wherein the separate storage area is a remote server.

41. The method of claim 39, wherein the separate storage area is a partition of the dedicated storage area of the jukebox.

42. The method of claim 39, wherein the separate storage area is a separate storage area within the jukebox.

43. The method of claim 35, wherein the determination of whether the instance of media is available for immediate play on the jukebox free from additional processing by the jukebox includes determining the user's search methodology provided by the interface.

44. A method of operating a digital downloading jukebox, the method comprising:

defining a plurality of channels, each said channel having a plurality of instances of media associated therewith;

selecting a channel from said plurality of channels on the part of an authorized user;

responding to one or more patron uses of the jukebox in a first mode, at least some of said uses including adding instances of media to a queue for play on the jukebox and playing said queued instances of media; and

when the jukebox is not responding to a patron request and/or playing a queued instance of media, playing the instances of media in the channel in a second mode,

wherein the first mode is different from the second mode,

and wherein the second mode is not directly controllable by a patron of the jukebox.

45. The method of claim 44, wherein the instances of media are associated with channels by an authorized user.

46. The method of claim 44, wherein the instances of media are associated with channels according to a theme and/or genre associated with each said instance of media.

47. The method of claim 44, further comprising when a control code is received from the authorized user, skipping the instance of media being played by the channel.

48. The method of claim 44, further comprising changing a frequency an instance of media in a channel is played in dependence on one or more control codes received from the authorized user,

wherein the control codes correspond to at least one of: skipping an instance of media being played by the channel, replaying an instance of media being or having just been played by the channel, and removing an instance of media from the channel.

49. The method of claim 44, wherein the second mode is a background music mode.

50. A method of delivering a coupon to a user of a digital downloading jukebox, the method comprising:

creating a validation code;

associating with the validation code a number of free credits for the user;

including the validation code with the coupon; and

transmitting the validation code to the user.

51. The method of claim 50, further comprising including in the validation code at least one location where the user can redeem the coupon.

52. The method of claim 50, further comprising including in the validation code a restriction on the time when the user can redeem the coupon.

53. The method of claim 50, wherein the coupon is transmitted to a mobile phone of the user.

54. The method of claim 50, wherein the validation code is an encoded alphanumeric sequence.

55. The method of claim 50, wherein the validation code is a barcode.

56. The method of claim 50, further comprising:
obtaining the validation code from the coupon;
determining whether the validation code is valid;
when the validation code is valid, providing to the user a number of credits in accordance with the number of free credits associated with the validation code.

57. The method of claim 56, wherein the validation code is obtained by the jukebox via a barcode scanner.

58. The method of claim 56, wherein the validation code is obtained by having the user input the validation code into the jukebox via a user interface provided to the jukebox.

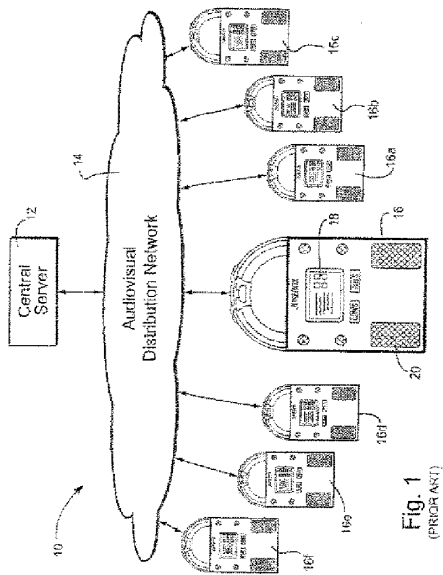
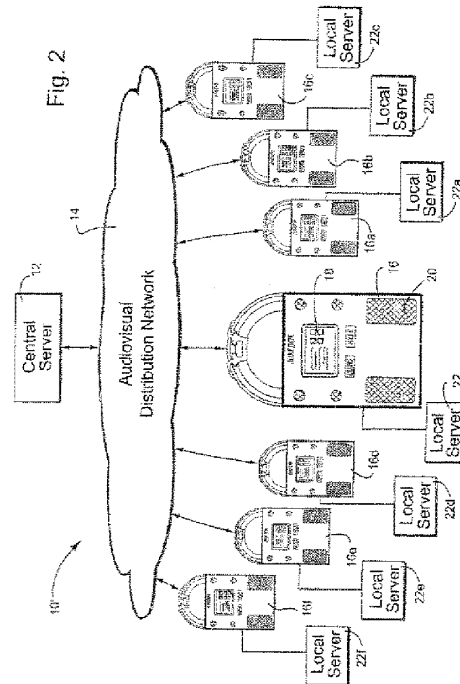
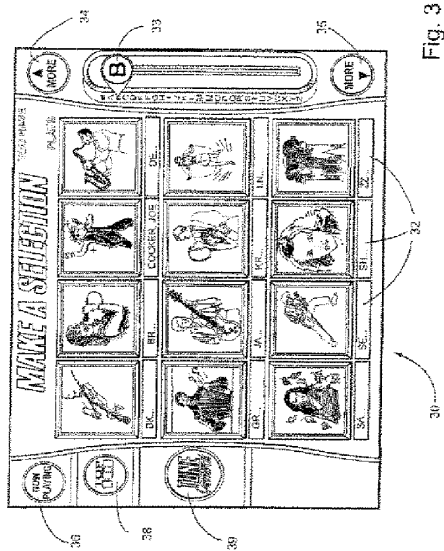
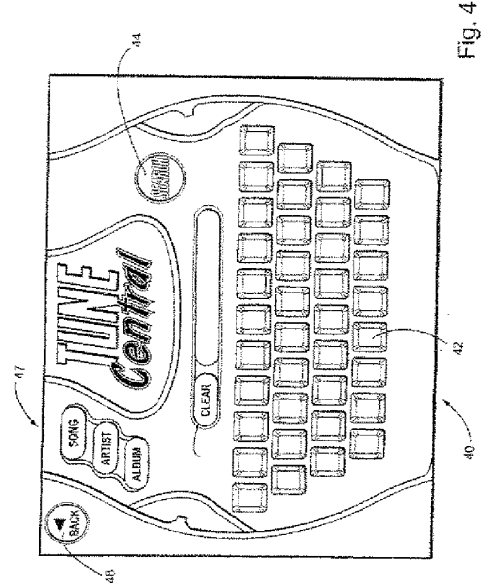
59. The method of claim 56, wherein the validation code is validated against a database of valid validation codes.

1 Abstract

Systems and/or methods for use in connection with digital downloading jukeboxes are provided. Such systems and/or methods may be used to provide enhanced communications capabilities, e.g., to registered users of jukeboxes. In
5 certain exemplary embodiments, users may become registered users directly at a jukebox. Registered users also may, in certain exemplary embodiments, receive coupons for free plays, import playlists from hardware devices and/or software applications, receive special pricing when playing certain instances of media, play
10 playlists in whole or in part, create and/or manage playlists directly at a jukebox, establish and manage connections with other registered users, etc. Still further, in certain exemplary embodiments, one or more channels may be predefined and/or set up for a jukebox.

2 Representative Drawing

Fig. 2



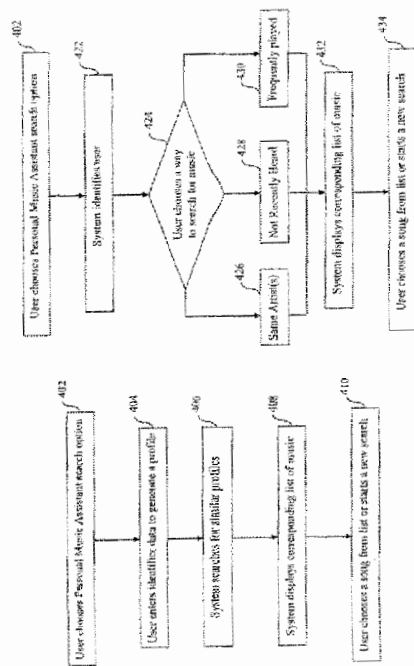


Fig. 4A

Fig. 4B

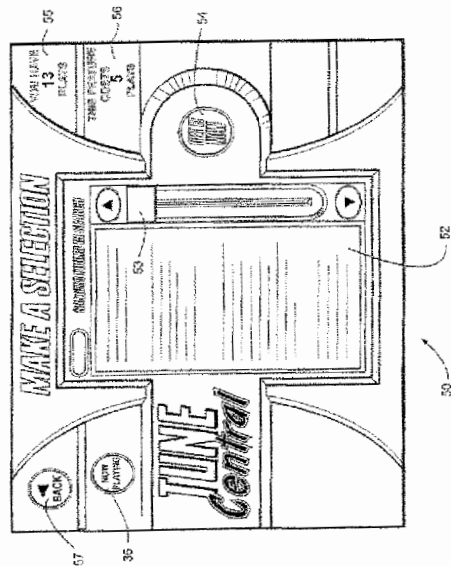


Fig. 5

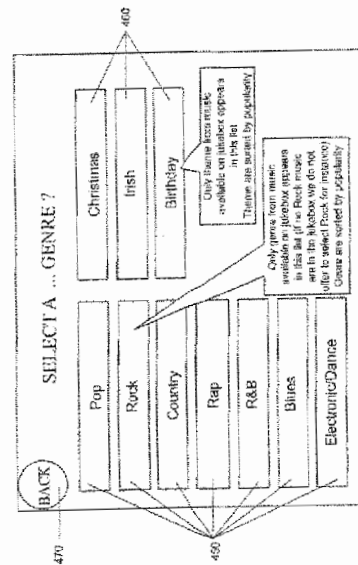


Fig. 4C

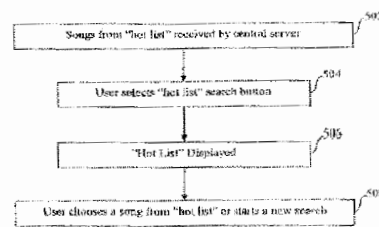


Fig. 5A

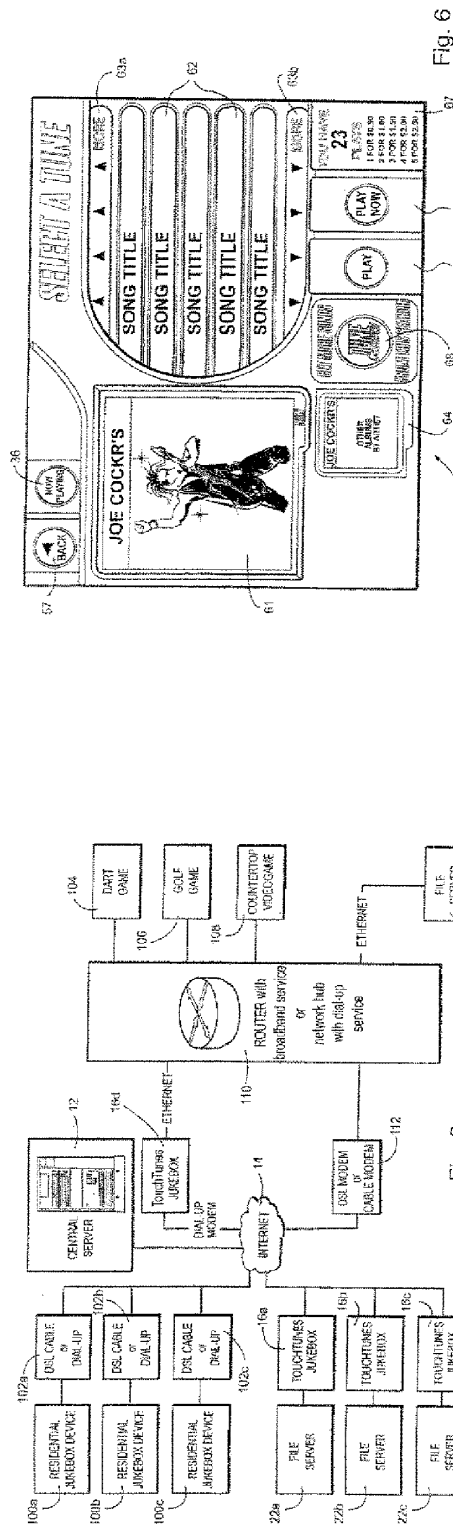


Fig. 6

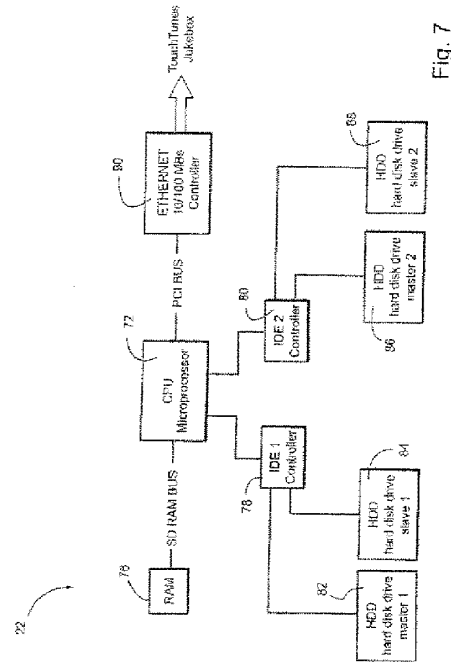


Fig. 7

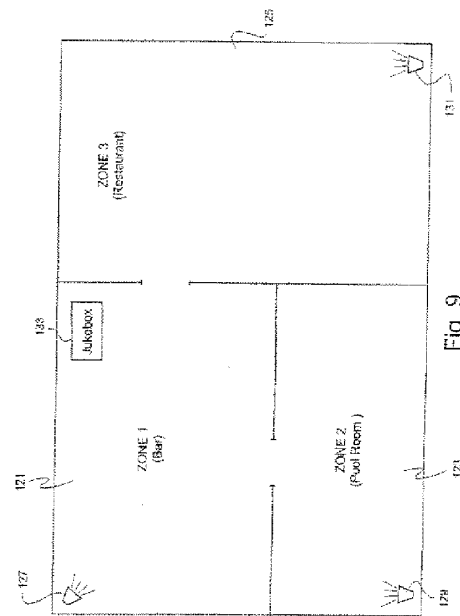


Fig. 9

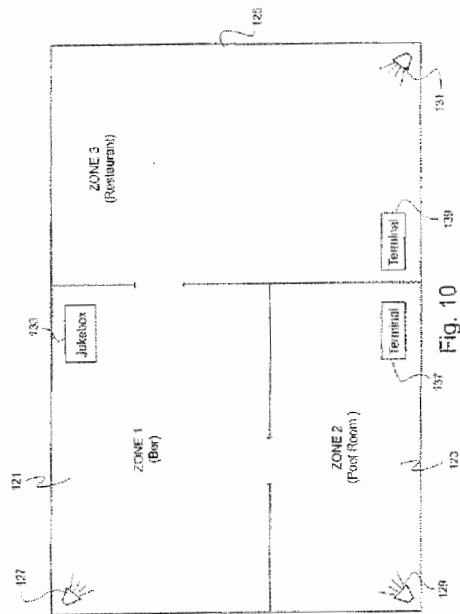


Fig. 10

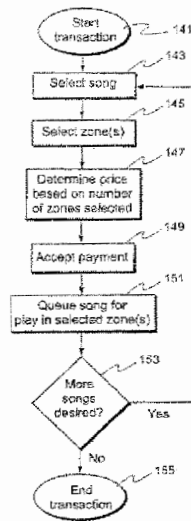


Fig. 11

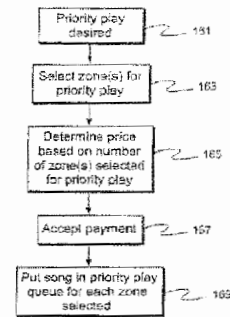


Fig. 12

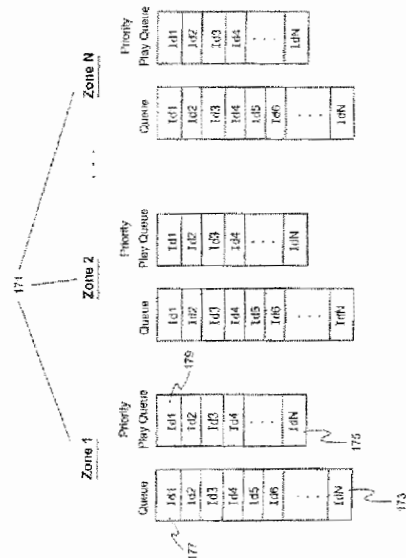


Fig. 13

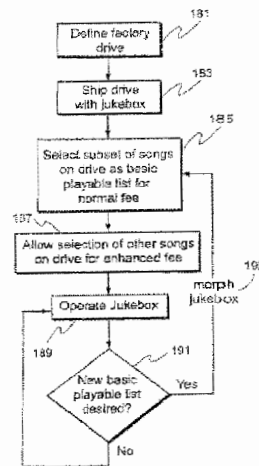


Fig. 14

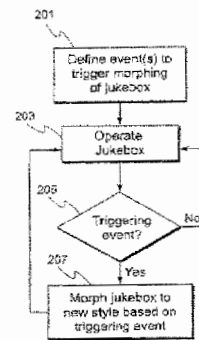
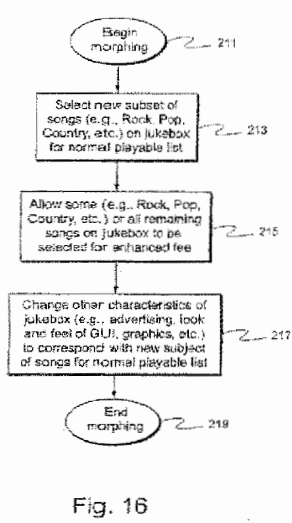
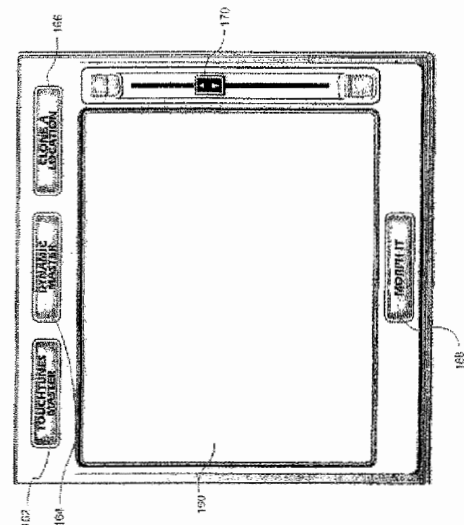
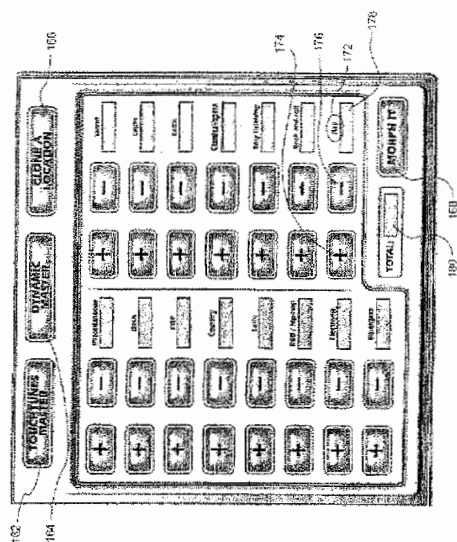
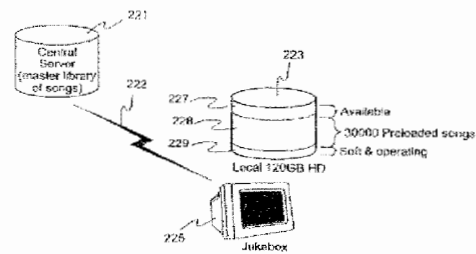


Fig. 15



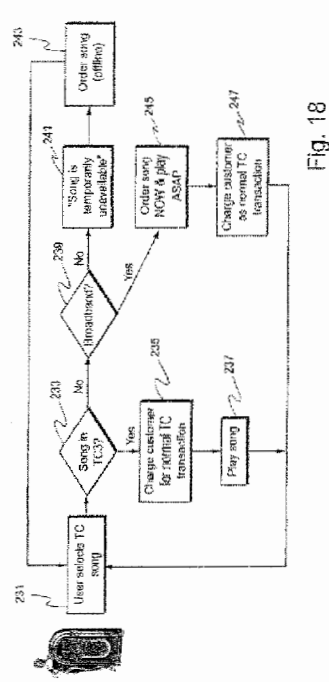


Fig. 18

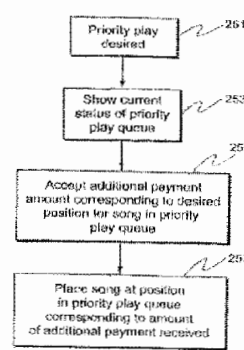


Fig. 19

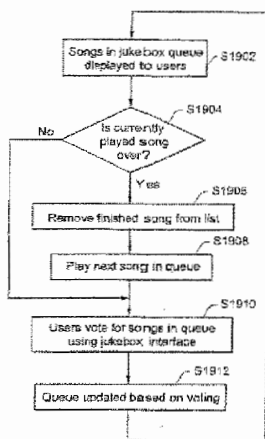


Fig. 19A

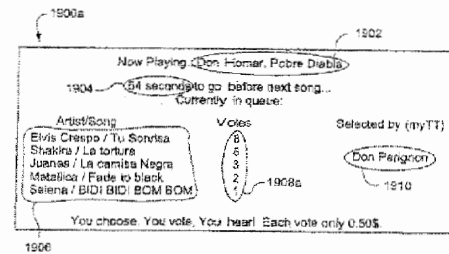


Fig. 19B(1)

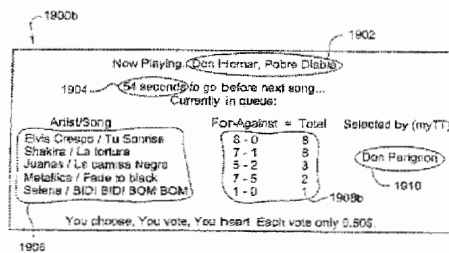


Fig. 19B(2)

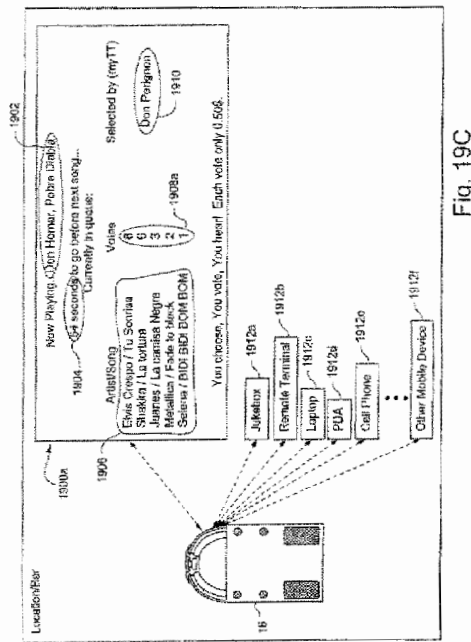


Fig. 19C

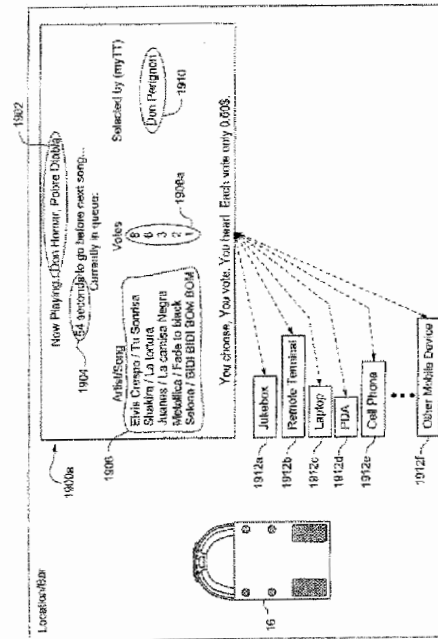


Fig. 19D

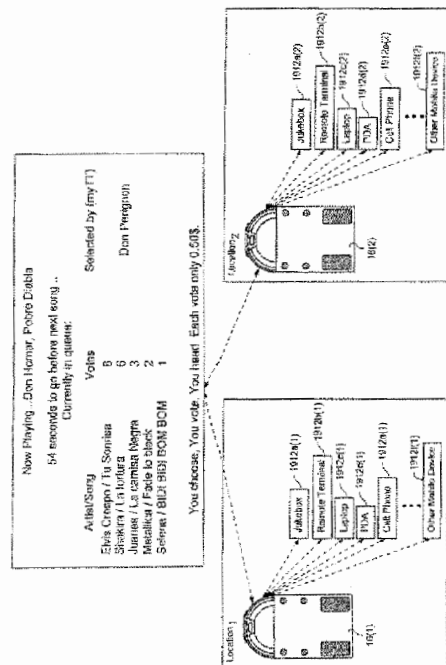


Fig. 19E

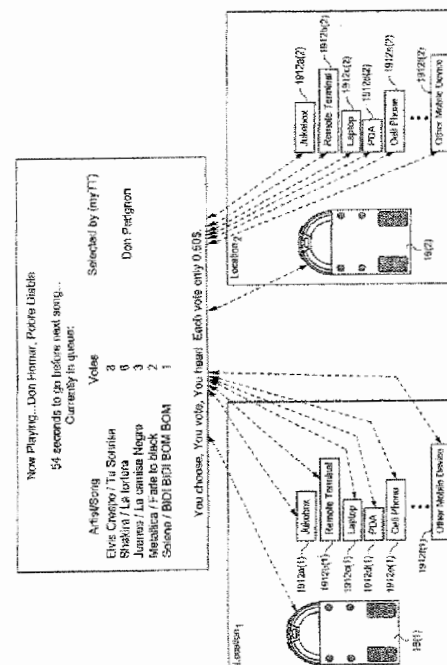


Fig. 19F

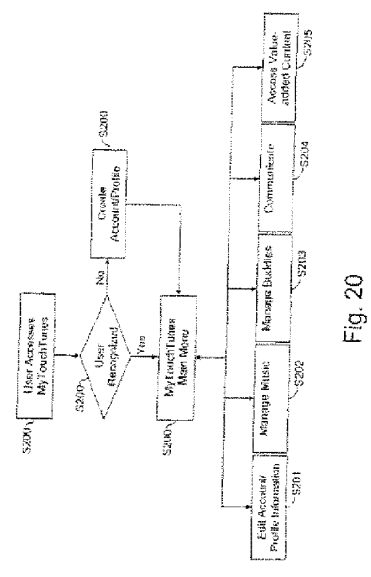


Fig. 20

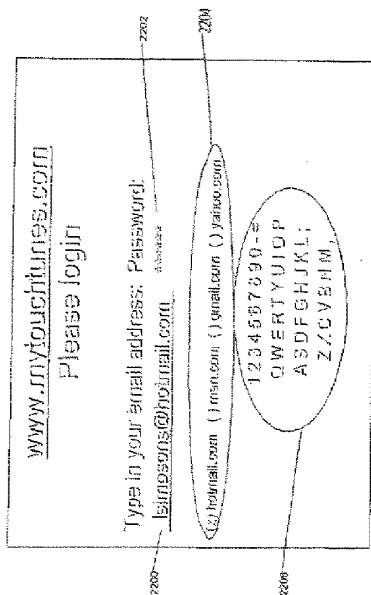


Fig. 22

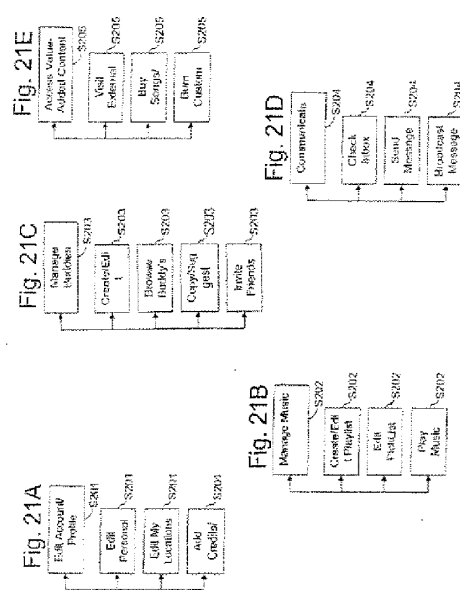
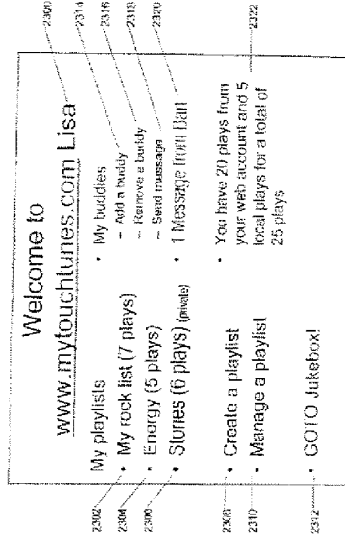


Fig. 23

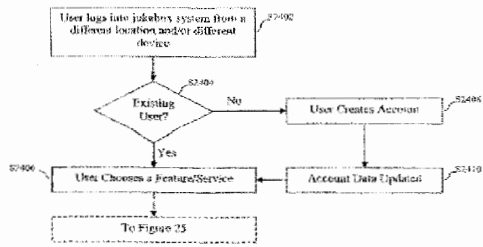


Fig. 24

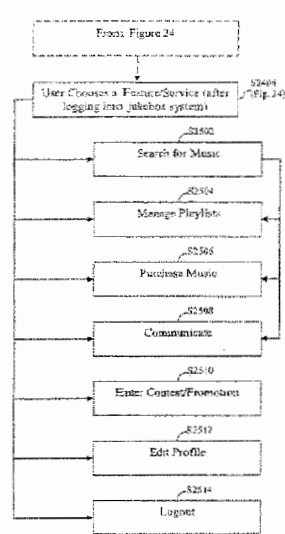


Fig. 25

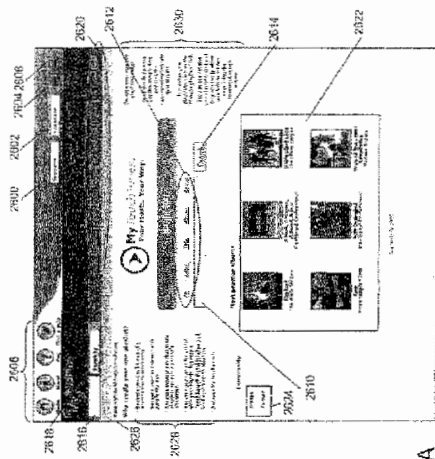


Fig. 26A

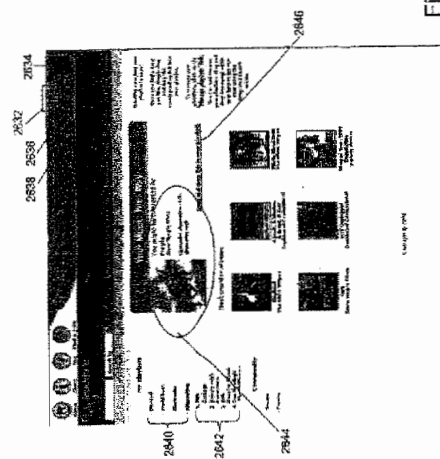


Fig. 26B

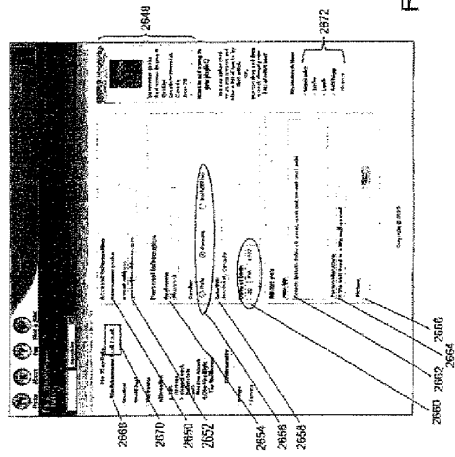


Fig. 26C

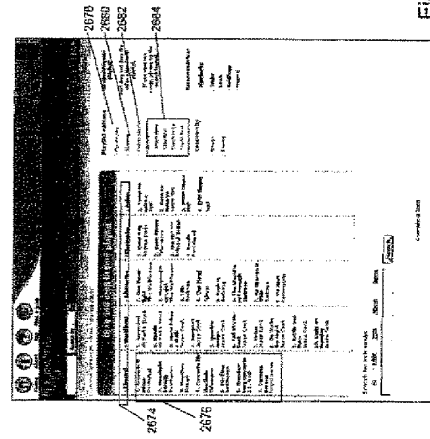


Fig. 26D

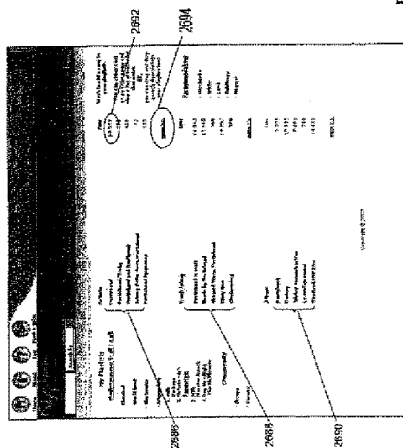


Fig. 26E

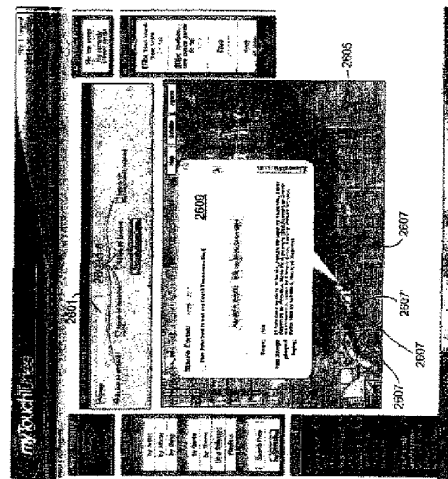


Fig. 26F

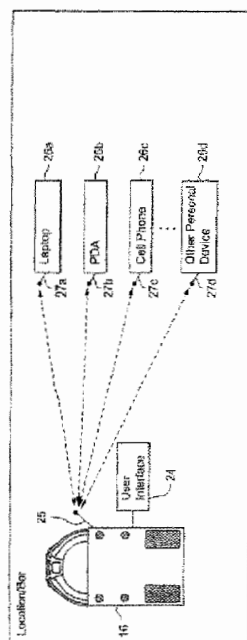


Fig. 29A

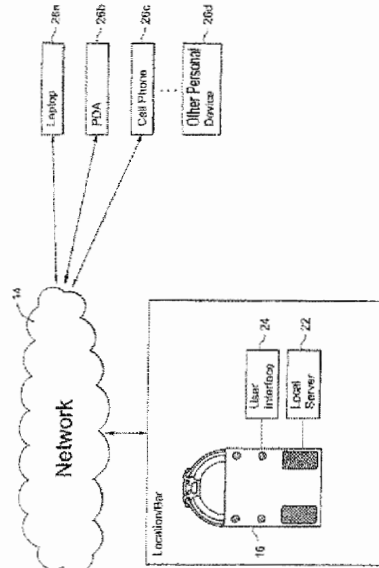


Fig. 29B

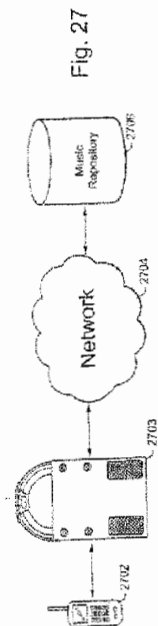


Fig. 27

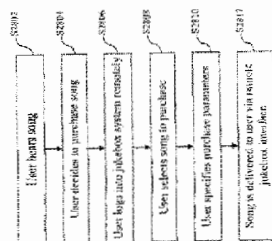


Fig. 28

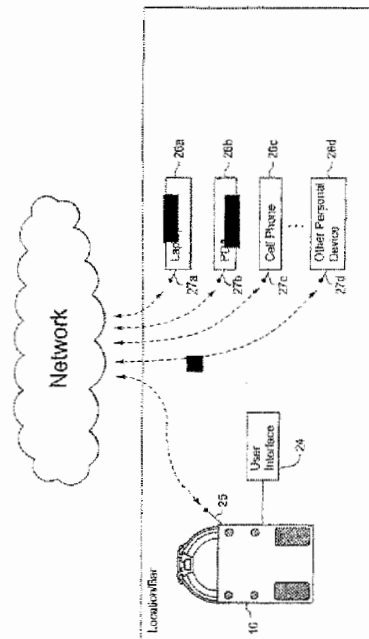


Fig. 29

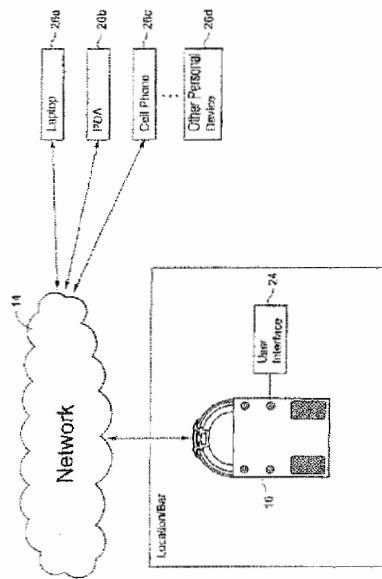


Fig. 29C

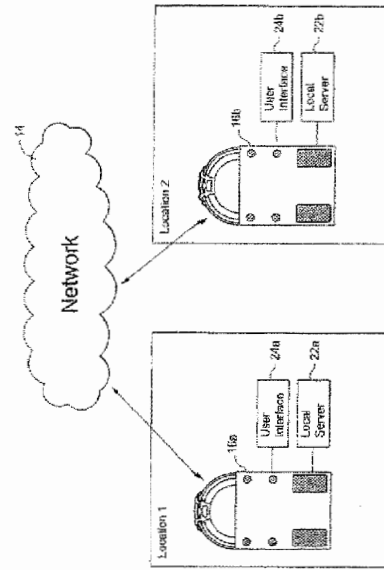


Fig. 29D

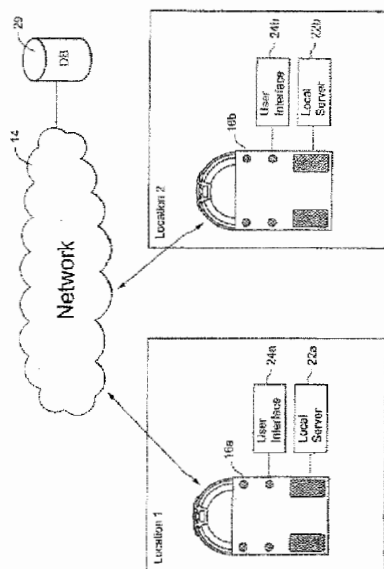


Fig. 29E

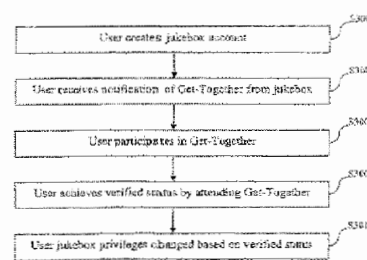


Fig. 30

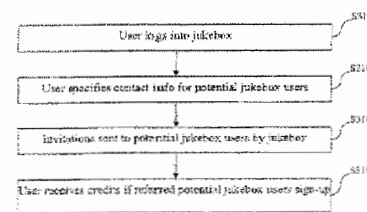


Fig. 31

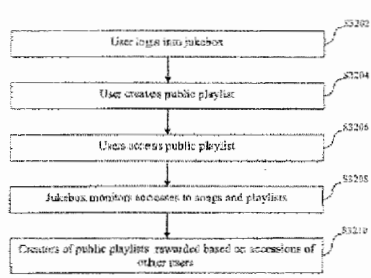


Fig. 32

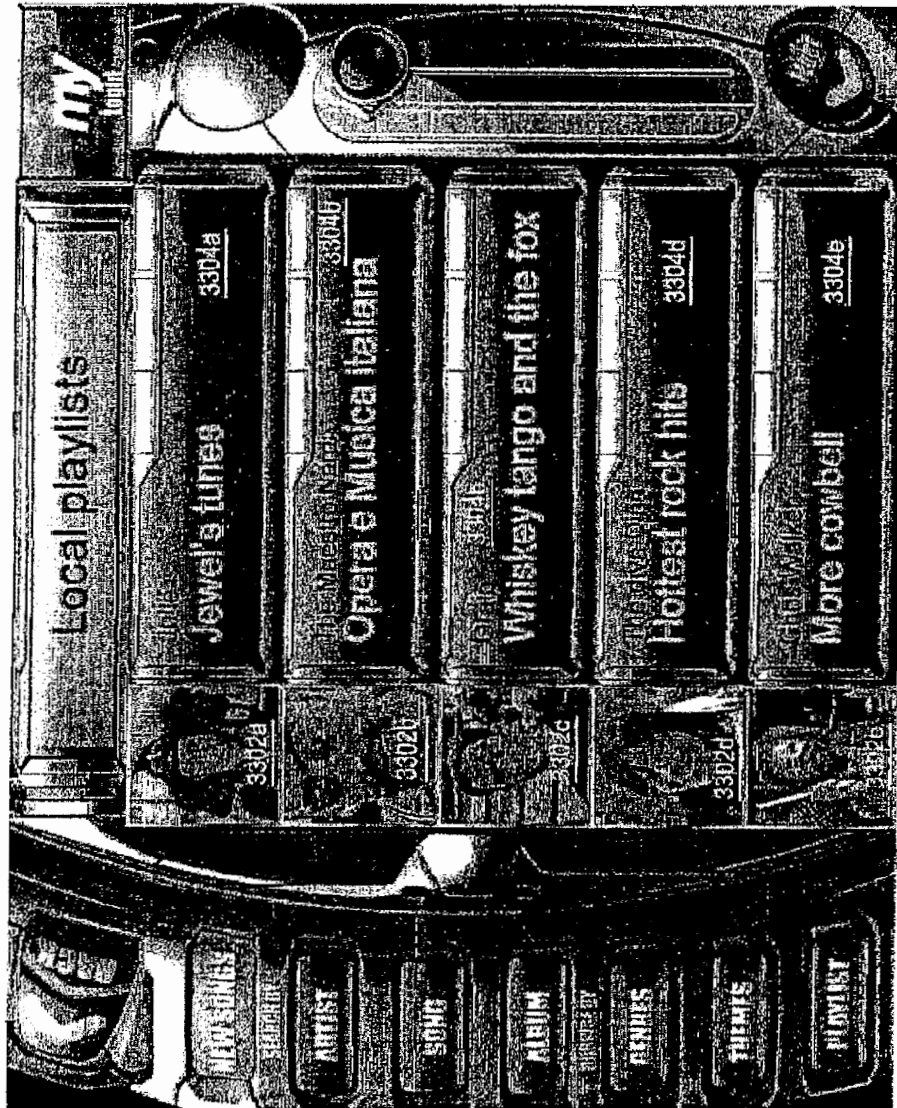


Fig. 33

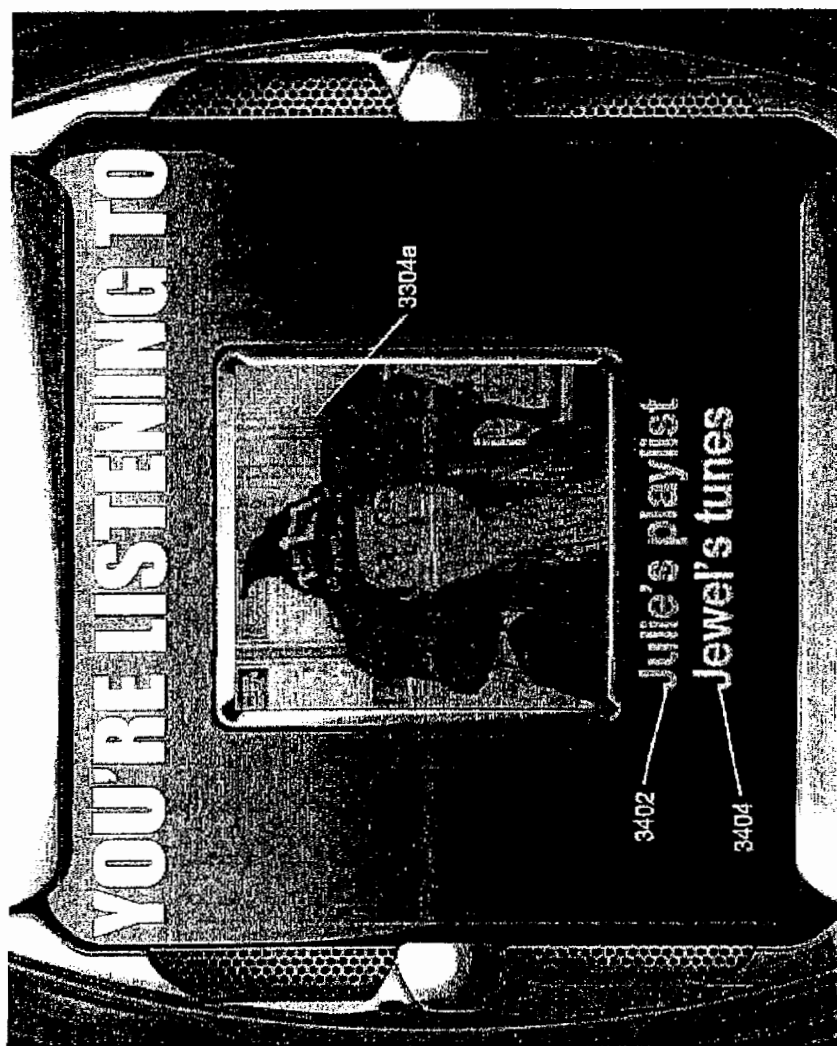


Fig. 34

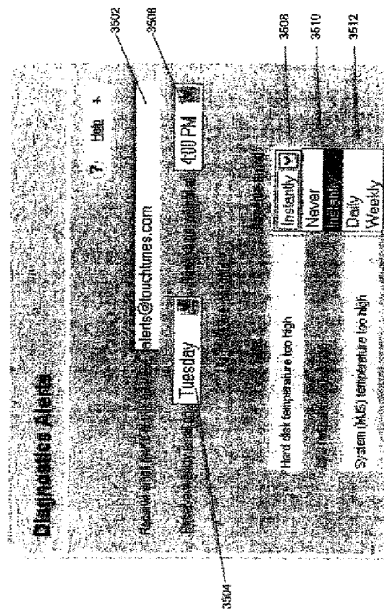


Fig. 35

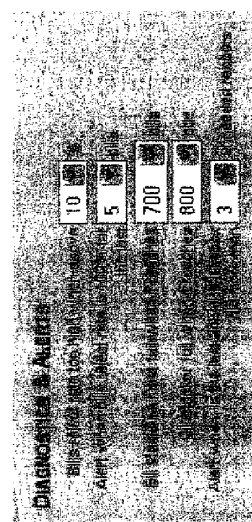


Fig. 36

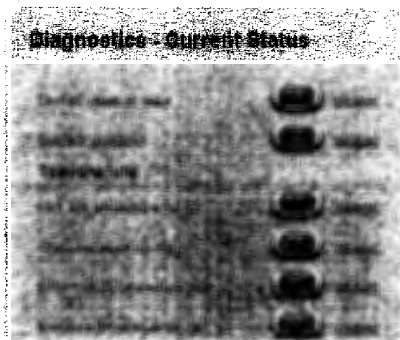


Fig. 37

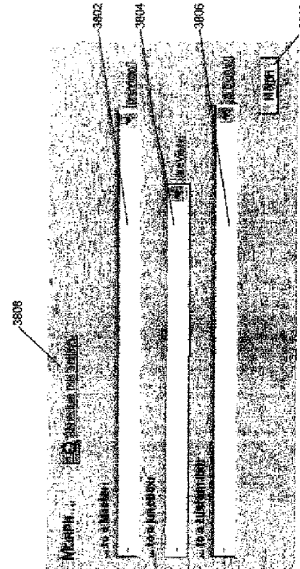


Fig. 38

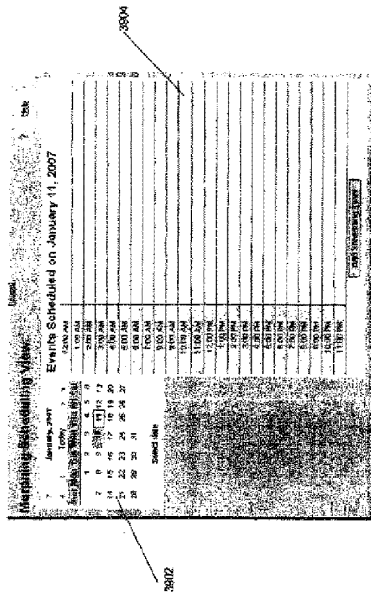


Fig. 39

(2 0 0)

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(71) Applicants and

(72) Inventors: **GOLDBERG, David** [US/US]; 4055 Pinion Drive, Boulder, CO 80303 (US). **SIMON, Neil, R.** [US/US]; 3084 4th Street, Boulder, CO 80304 (US). **GOLDBERG, Martha, B.** [US/US]; 4055 Pinion Drive,

Boulder, CO 80303 (US). **GOLDBERG, Miriam, D.** [US/US]; 4055 Pinion Drive, Boulder, CO 80303 (US). **GOLDBERG, Benjamin, M.** [US/US]; 4055 Pinion Drive, Boulder, CO 80303 (US).

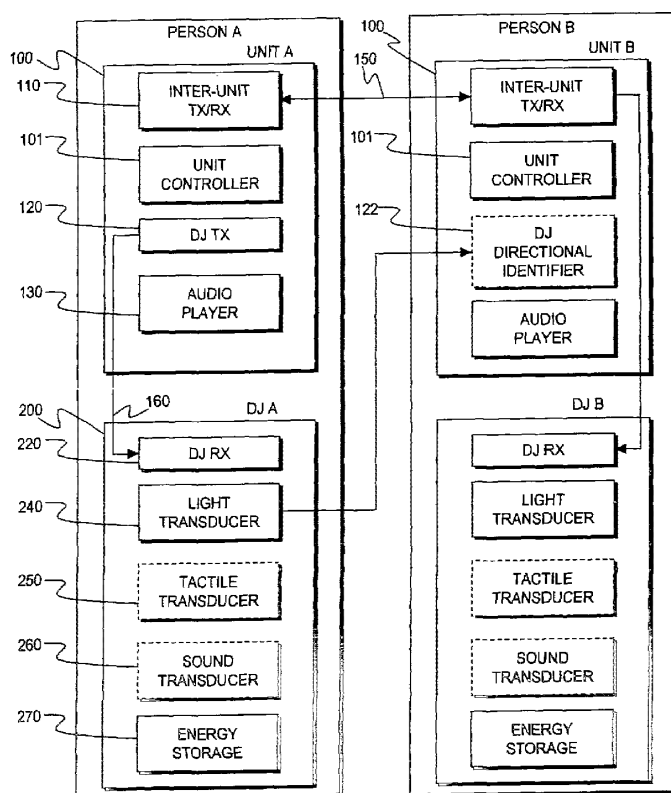
(74) Agent: **McINTOSH, Michael, D.**; 8000 E. Prentice Avenue, Suite B-6, Greenwood Village, CO 80111 (US).

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[Continued on next page]

(54) Title: LOCALIZED AUDIO NETWORKS AND ASSOCIATED DIGITAL ACCESSORIES



(57) Abstract: Individuals share music through portable audio devices that synchronously play music by transmitting music signals from one device to the other. Upon request, other individuals can join such a sharing group as a member, and these new members can either be accepted automatically, or through voting by existing group members. So that members of a group can converse, the fraction of ambient sound admitted by the earphones of the portable device can be manually set. In order to enhance the enjoyment of the music, wearable transducers, such as light emitting diodes, respond to signals related to the beat of the music. Signals can be automatically generated, or manually specified, and signals thereafter stored for playback with the music to which it is related. Wearable transducers associated with individuals at a large music event can be set to receive common signals such that the transducers respond in synchrony with one another



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LOCALIZED AUDIO NETWORKS AND ASSOCIATED DIGITAL ACCESSORIES**Cross Reference To Related Patent Applications**

This application is related to and claims priority from Provisional Patent Application No. 60/378,415, filed May 6, 2002, titled "Localized Audio Networks and Associated Digital Accessories," and from Provisional Patent Application No. 60/388,887, filed June 14, 2002, titled "Localized Audio Networks and Associated Digital Accessories," and from Provisional Patent Application No. 60/452,230, filed March 4, 2003, titled "Localized Audio Networks and Associated Digital Accessories," the contents of each of which are incorporated herein by reference.

Technical Field

The present invention relates to localized wireless audio networks for shared listening of recorded music, and wearable digital accessories for public music-related display, which can be used in conjunction with one another.

Background

Portable audio players are popular consumer electronic products, and come in a variety of device formats, from cassette tape "boom boxes" to portable CD players to digital flash-memory and hard-disk MP3 players. While boom boxes are meant to make music to be shared among people, most of the portable audio players are designed for single person use. While some of this orientation to personal music listening is due to personal preference, other important considerations are the technical difficulties of reproducing music for open area listening with small, portable devices, as well as the social imposition of listening to music in public places with other people who do not wish to listen to the same music, or who are listening to different music that would interfere with one's own music.

There are numerous audio devices that are designed to allow the transfer of music from one portable audio device to another, especially through those that store music in the MP3 audio format. These devices suffer from two main difficulties: firstly, listeners still do not hear the music simultaneously, which is the optimal manner to share music, and secondly, there are serious copyright issues associated with the transfer of music files. Thus, it would be preferable for the transfer of the music for simultaneous enjoyment, and which did not

result in a permanent transfer of the music files between the devices, so as not to infringe on the intellectual property rights of the music owners.

Given the sharing of music, listeners will on occasion want to purchase the music for themselves. In such case, it would be beneficial for the user to have a way to obtain the music with minimal effort. It would further be desirable for there to be a way to keep track of the person from whom the listener heard the music, so that the person could be in some way encouraged or compensated.

The earphones associated with a portable music player admit a relatively constant fraction of ambient sound. If listening to music with a shared portable music device, however, one might at times want to talk with a friend, and at times listen to music without outside audible distraction. In such case, it would be desirable to have an earphone for which the amount of external ambient sound could be manually set.

Furthermore, many people like to show their individual preferences, to exhibit themselves, and to demonstrate their group membership. Furthermore, music preferences and listening to music together are among the more important means by which individuals express their individual and group identities. It would be beneficial for there to be a way for individuals to express themselves through their music, and for groups of individuals listening to music together, to be able to demonstrate their group enjoyment of the music.

One means for a person to express their identity through motion would be through having wearable transducers wherein the transduction signal is related to the music. If the transducer were a light transducer, this would result in a display of light related to the music that was being listened to. It would be further beneficial if there were means by which a person could generate control signals for the transducer so that instead of a wholly artificial response to the music, the transducer showed a humanly interpreted display. It would be preferable if these signals could be shared between people along with music files, so that others could entertain or appreciate the light display so produced.

At popular music concerts, there is often a "light show" that pulsates in rough relation to the music. In contrast to the generally vigorous light show, the patrons at the concerts often have light bracelets or other such static displays which are used to join with the displays on the stage. It would be beneficial for there to be a way in which patrons could participate in the light show in order to enhance their enjoyment of the concert.

It is to the solution of these and other problems that the present invention is directed.

Summary of the Invention

It is an object of the present invention to provide users a means of listening to music together using mobile devices.

It is also an object of the present invention to provide users a means of choosing with
5 whom to listen to music.

It is additionally an object of the present invention to provide users the ability to monitor the people that are listening together.

It is furthermore an object of the present invention to provide users a means of expressing their enjoyment of the music they are listening to through visual displays of
10 wearable accessories.

It is yet another object of the present invention to provide users a means of demonstrating their identity with other people they are listening to music with.

It is still further an object of the present invention to provide users to provide users with means to choreograph the visual displays.

Additional objects, advantages and novel features of this invention shall be set forth in
15 part in the description that follows, and will become apparent to those skilled in the art upon examination of the following specification or may be learned through the practice of the invention. The objects and advantages of the invention may be realized and attained by means of the instrumentalities, combinations, and methods particularly pointed out in the
20 appended claims.

To achieve the foregoing and other objects and in accordance with the purposes of the present invention, as embodied and broadly described therein, the present invention is directed to a method for sharing music from stored musical signals between a first user with a first music player device and at least one second user with at least one second music player
25 device. The method includes the step of playing the musical signals for the first user on the first music player device while essentially simultaneously wirelessly transmitting the musical signals from the first music player device to the at least one second music player device. The method additionally includes receiving the musical signals by the at least one second player device, such that the musical signals can be played on the at least one second player device
30 essentially simultaneously with the playing of the musical signals on the first music player device. In this method, the first and the at least one second users are mobile and maintain less than a predetermined distance.

The present invention is also related to a system of music sharing for a plurality of users. The system includes a first sharing device and at least one second sharing device, each comprising a musical signal store, a musical signal transmitter, a musical signal receiver, and a musical signal player. Furthermore, the system comprises a broadcast user operating the first sharing device and at least one member user operating the at least one second sharing device. The broadcast user plays the musical signal for his own enjoyment on the first sharing device and simultaneously transmits the musical signal to the receiver of the at least one second sharing device of the at least one member user, on which the musical signal is played for the at least one member user. The broadcast user and the at least one member user hear the musical signal substantially simultaneously.

The present invention yet further is related to a wireless communications system for sharing audio entertainment between a first mobile device and a second mobile device in the presence of a non-participating third mobile device. The system includes an announcement signal transmitted by the first mobile device for which the second mobile device and the third mobile device are receptive. In addition, the system includes a response signal transmitted by the second mobile device in response to the announcement signal for which the first mobile device is receptive and for which the third mobile device is not receptive. Also, the system includes an identifier signal transmitted by the first mobile device to the second mobile device in response to the response signal, and which is not receptive to the third mobile device. Finally, the system includes a broadcast signal comprising audio entertainment that is transmitted by the first mobile device, and which is receptive by the second mobile device on the basis of the reception of the identifier signal.

The present invention additionally is related to an audio entertainment device. The device includes a signal store that stores an audio entertainment signal, a transmitter that can transmit the stored audio entertainment signal, a receiver that can receive the transmitted audio entertainment signal from a transmitter of another such device, and a player that can play audio entertainment from a member selected from the group of stored audio entertainment signals or audio entertainment signals transmitted from the transmitter of another such device.

The present invention yet still is related to a system for identifying a first device that introduces a music selection to a second device. The system includes a mobile music transmitter operated by the first device and a mobile music receiver operated by the second device. In addition, the system includes a music signal comprising the music selection transmitted by the transmitter and received by the receiver, an individual musical identifier

that is associated with the music selection, and an individual transmitter identifier that identifies the transmitter. The transmitter identifier and the individual music identifier are stored in association with each other in the receiver.

The present invention is still further related to an audio entertainment device. The device includes a wireless transmitter for the transmission of audio entertainment signals and a wireless receiver for the reception of the transmitted audio entertainment signals from a transmitter of audio entertainment signals. A first manually-separable connector for electrically connecting with an audio player allows transfer of audio entertainment signals from the player to the device. The device also includes a second connector for connecting with a speaker and a control to manually switch between at least three states. In the first state the speaker plays audio entertainment signals from the audio player and the transmitter does not transmit the audio entertainment signals. In the second state the speaker plays audio entertainment signals from the audio player and the transmitter essentially simultaneously transmits the audio entertainment signals. In the third state the speaker plays audio entertainment signals received by the receiver.

The present invention also still is related to a system for the sharing of stored music between a first user and a second user. The system includes a first device for playing music to the first user, comprising a store of musical signals. A first controller prepares musical signals from the first store for transmission and playing, and a first player takes musical signals from the first controller and plays the signals for the first user. A transmitter is capable of taking the musical signals from the controller and transmitting the musical signals via wireless broadcast. A second device for playing music to the second user comprises a receiver receptive of the transmissions from the transmitter of the first device, a second controller that prepares musical signals from the receiver for playing, and a second player that takes musical signal from the second controller and plays the signals for the second user. The first user and the second user hear the musical signals at substantially the same time.

The present invention also is related to an earphone for listening to audio entertainment allowing for the controlled reception of ambient sound by a user. The earphone includes a speaker that is oriented towards the user's ear and an enclosure that reduces the amount of ambient noise perceptible to the user. In addition, a manually-adjustable characteristic of the enclosure adjusts the amount of ambient sound perceptible to the user.

The present invention is further related to a mobile device for the transmission of audio entertainment signals. The mobile device includes an audio signal store for the storage

of the audio entertainment signals, and an audio signal player for the playing of the audio entertainment signals. The device also includes a wireless transmitter for the transmission of the audio entertainment signals and a transmitter control to manually switch between two states consisting of the operation and the non-operation of the audio transmitter.

5 The present invention yet still is related to a mobile device for the reception of digital audio entertainment signals. The mobile device includes an audio signal store for the storage of the digital audio entertainment signals and an audio receiver for the reception of external digital audio entertainment signals from a mobile audio signal transmitter located within a predetermined distance of the audio receiver. The device also includes a receiver control with
10 at least a first state and a second state. An audio signal player plays digital audio entertainment signals from the audio signal store when the receiver control is in the first state, and plays digital audio entertainment signals from the audio receiver when the receiver control is in the second state.

 The present invention furthermore relates to a method for the shared enjoyment of
15 music from stored musical signals between a first user with a first music player device and at least one second user with at least one second music player device. The method includes the step of playing the musical signals for the first user on the first music player device while essentially simultaneously wirelessly transmitting synchronization signals from the first music player device to the at least one second music player device. The method also includes
20 receiving the synchronization signals by the at least one second player device. The synchronization signals allow the musical signals on the at least one second player device to be played essentially simultaneously with the playing of the musical signals on the first music player device. The first and the at least one second users are mobile.

 The present invention yet furthermore relates to a wireless communications system
25 for sharing audio entertainment between a first mobile device and a second mobile device. The system includes a broadcast identifier signal transmitted by the first mobile device to the second mobile device. A personal identifier signal is transmitted by the second mobile device to the first mobile device. A broadcast signal comprising audio entertainment is transmitted by the first mobile device of which the second device is receptive. The first
30 mobile device and the second mobile device have displays which can display the identifier signal that they receive and the second mobile device can play the audio entertainment from the broadcast signal that it receives.

 The present invention also relates to a method for enhancing enjoyment of a musical selection. The method includes the steps of obtaining control signals related to the musical

selection, transmitting the control signals wirelessly, receiving the control signals, and converting the control signals to a humanly-perceptible form.

5 The present invention further yet relates to a method for generating and storing control signals corresponding to musical signals. The method includes the steps of playing musical signals for a user and receiving manual input signals from the user that are produced substantially in synchrony with the music. The method also includes the steps of generating control signals from the input signals, and storing the control signals so that they can be retrieved with the musical signals.

10 The present invention still additionally relates to a wearable personal accessory. The accessory includes an input transducer taken from the group consisting of a microphone and an accelerometer. The transducer generates a time-varying input transduction signal. The accessory also includes a controller that accepts the input transduction signal, and generates an output transducer signal whose signal varies in amplitude with time. An output transducer receptive of the output transducer signal provides a humanly-perceptible signal. An energy
15 source powers the input transducer, controller and output transducer.

The present invention also still relates to a wearable personal accessory controlled via wireless communications. The accessory includes a wireless communications receiver that is receptive of an external control signal. The accessory also includes a controller that accepts the external control signal and that generates a time-varying visual output transducer signal.
20 A visual output transducer is receptive of the output transducer signal, and provides a humanly-perceptible visual signal. An energy store powers the receiver, controller and output transducer. The visual output transducer generates visually-perceptive output.

The present invention still further relates to a device for converting user tactile responses to stored music into a stored control signal. The device includes a player that plays
25 stored music audible to the user and a manually-operated transducer that outputs an electrical signal. The transducer is actuated by the user in response to the music. A controller receives the electrical signal and outputs a control signal and a store receives the control signal and stores it.

30 The present invention furthermore relates to a music player that wirelessly transmits control signals related to the music, wherein the control signals control a wearable electronic accessory. The music player includes a store of music signal files and a controller that reads a musical signal file from the store and generates audio signals. The controller further generates the control signals. A transducer converts the audio signals into sound audible to

the user and a wireless transmitter transmits the control signal to the wearable electronic accessory.

The present invention yet relates to a music player that wirelessly transmits control signals related to the music, wherein the control signals control a wearable electronic accessory. The music player includes a store of music signal files and a second store of control signal files associated with the music signal files. A controller reads a musical signal file from the store and generates audio signals. The controller further reads an associated control signal file. A transducer converts the audio signals into sound audible to the user, and a wireless transmitter transmits the control signals from the associated control signal file to the wearable electronic accessory.

The present invention also relates to a system for exhibition of music enjoyment. The system includes a source of music signals, a controller that generates control signals from the music signals, and a transmitter of the control signals. The transmission of the control signals is synchronized with the playing of the music signals. In addition, the system includes a receiver of the control signals and a transducer that responds to the control signals.

The present invention further relates to a method for transferring a wearable-accessory control file stored on a first device to a second device in which an associated music file is stored. The method includes the steps of storing on the first device the name of the music file in conjunction with the control file with which it is associated and requesting by the second device of the first device for a control file stored in conjunction with the name of the music file. In addition, the method includes the step of transferring the control file from the first device to the second device. The control file is stored on the second device in conjunction with the name of the associated music file.

The present invention also relates to a device for transmitting control signals to a wearable accessory receptive of such control signals. The device includes a manually-separable input connector for connecting to an output port of an audio player. Audio signals are conveyed from the audio player to the device across the connector. The device also includes a controller for generating control signals from the audio signals and a transmitter for transmitting the control signals.

Brief Description Of The Drawings

Figure 1 is a schematic block diagram of a local audio network comprised of two linked audio units operated by two persons, and associated digital jewelry conveyed by the two persons.

Figure 2A is a schematic block diagram of a DJ with multiple independently controlled LED arrays.

Figure 2B is a schematic block diagram of a DJ with an LED array with independently controlled LEDs.

5 Figures 3A-C are schematic block diagrams of unit elements used in inter-unit communications.

Figure 4 is a schematic flow diagram of DJ entraining.

Figures 5A-B are schematic block diagrams of DJs associated with multiple people bound to the same master unit.

10 Figure 6 is a schematic block diagram of a cluster comprising a broadcast unit and multiple receive units, with an external search unit.

Figure 7 is a schematic diagram of a broadcast unit transmission.

Figure 8A is a schematic block diagram of audio units with self-broadcast so that audio output is highly synchronized.

15 Figure 8B is a schematic flow diagram for synchronous audio playing with multiple rebroadcast.

Figures 9A and 9B are schematic block diagrams of hierarchically-related clusters.

Figure 10 is a top perspective view of an earphone with manually adjustable external sound ports.

20 Figures 11 A and B are cross-sectional diagrams of a earpiece with an extender to admit additional ambient sound.

Figure 12A is a schematic diagram of a modular audio unit.

Figure 12B is a schematic diagram of modular digital jewelry.

25 Figure 12C is a schematic block diagram of a modular transmitter that generates and transmits control signals for digital jewelry from an audio player.

Figure 13A is a schematic cross-section through a search unit and a broadcast unit in which communications are provided via visible or infrared LED emission in search transmission mode.

30 Figure 13B is a schematic cross-section through a search unit and a broadcast unit in which communications are provided via a visible or infrared laser in search transmission mode.

Figure 13C is a schematic cross-section through a search unit and a broadcast unit in which communications are provided via visible or infrared emission from a digital jewelry element in broadcast transmission mode.

Figure 13D is a schematic cross-section through a search unit and a broadcast unit in which communications are provided via contact in mutual transmission mode.

Figure 13E is a schematic cross-section through a search unit and a broadcast unit in which communications are provided via sonic transmissions in broadcast transmission mode.

5 Figure 13F is a schematic cross-section through a search unit and a broadcast unit in which communications are provided via radio frequency transmissions in broadcast transmission mode.

Figure 14A is a schematic block diagram of the socket configurations on the broadcast unit and the receive unit.

10 Figure 14B is a schematic block flow diagram of using IP sockets for establishing and maintaining communications between a broadcast unit and the receive unit, according to the socket diagram of Figure 14A.

Figure 15 is a schematic block diagram of the IP socket organization used with clusters comprising multiple members.

15 Figure 16 is a schematic block flow diagram of transfer of control between the broadcast unit and the first receive unit.

Figure 17 is a matrix of DJ and searcher preferences and characteristics, illustrating the matching of DJ and searcher in admitting a search to a cluster.

20 Figure 18A is a screenshot of an LCD display of a unit, taken during normal operation.

Figure 18B is a screenshot of an LCD display of a unit, taken during voting for a new member.

Figure 19 is a table of voting schemes for the acceptance of new members into a cluster.

25 Figure 20 is a time-amplitude trace of an audio signal automatically separated into beats.

Figure 21A is a block flow diagram of a neural network method of creating DJ transducer control signals from an audio signal as shown in Figure 20.

30 Figure 21B is a block flow diagram of a deterministic signal analysis method of creating DJ transducer control signals from an audio signal as shown in Figure 20.

Figure 21C is a schematic flow diagram of a method to extract fundamental musical patterns from an audio signal to create DJ control signals.

Figure 21D is a schematic flow diagram of an algorithm to identify a music model resulting in a time signature.

Figure 22A is a top-view diagram of an audio unit user interface, demonstrating the use of buttons to create DJ control signals.

Figure 22B is a top-view diagram of a hand-pad for creating DJ control signals.

Figure 22C is a schematic block diagram of a set of drums used for creating DJ
5 control signals.

Figure 23A is a schematic block flow diagram of the synchronized playback of an audio signal file with a DJ control signal file, using transmission of both audio and control signal information.

Figure 24 is a schematic block diagram of a DJ unit with associated input transducers.

Figure 25 is a schematic flow diagram indicating music sharing using audio devices,
10 providing new means of distributing music to customers.

Figure 26 is a schematic diagram of people at a concert, in which DJs conveyed by multiple individuals are commonly controlled.

Figure 27 is a schematic block flow diagram of using a prospective new member's
15 previous associations to determine whether the person should be added to an existing cluster.

Figure 28 is a block flow diagram indicating the steps used to maintain physical proximity between the broadcast unit and the receive unit via feedback to the receive unit user.

Figures 29A is a schematic block diagram of the connection of an Internet-enabled
20 audio unit with an Internet device through the Internet cloud, using an Internet access point.

Figures 29B is a schematic block diagram of the connection of an Internet-enabled audio unit with an Internet device through the Internet cloud, with an audio unit directly connected to the Internet cloud.

Figure 30 comprises tables of ratings of audio unit users.

Figure 31 is tables of DJ, song and transaction information according to the methods
25 of Figure 25.

Figure 32A is a schematic block diagram of maintaining privacy in open transmission communications.

Figure 32B is a schematic block diagram of maintaining privacy in closed
30 transmission communication.

Figure 33 is a schematic block diagram of a hierarchical cluster, as in Figure 9A, in which communications between different units is cryptographically or otherwise restricted to a subset of the cluster members.

Figure 34B is a schematic layout of a synchronization record according to Figure 34A.

Figure 36 is a schematic block diagram of mode switching between peer-to-peer and infrastructure modes.

Best Mode for Carrying-Out the Invention

Figure 1 is a schematic block diagram of a local audio network comprised of two linked audio units 100 operated by two persons, and associated digital jewelry 200 conveyed by the two persons. The persons are designated Person A and Person B, their audio units 100 are respectively Unit A and Unit B, and their digital jewelry 200 are denoted respectively DJ A and DJ B. In this patent specification, “DJ” is used to denote either the singular “digital jewel” or the plural “digital jewelry”.

In operation, Unit A audio player 130 is playing recorded music under the control of a person to be designated User A. This music can derive from a variety of different sources and storage types, including tape cassettes, CDs, DVDs, magneto-optical disks, flash memory, removable disks, hard-disk drives or other hard storage media. Alternatively, the audio signals can be received from broadcasts using analog (e.g. AM or FM) or digital radio receivers. Unit A is additionally broadcasting a signal through DJ transmitter 120, which is received by DJ 200 through a DJ receiver 220 that is worn or otherwise conveyed by User A.

It should be noted that the audio signals can be of any sound type, and can include spoken text, symphonic music, popular music or other art forms. In this specification, the terms audio signal and music will be used interchangeably.

5 The DJ 200 transduces the signal received by the DJ receiver 220 to a form perceptible to the User A or other people near to him. This transduced form can include audio, visual or tactile elements, which are converted to their perceptible forms via a light transducer 240, and optionally a tactile transducer 250 or an audio transducer 260. The transducers 240, 250 and 260 can either directly generate the perceptible forms directly from the signals received by the DJ receiver 220, or can alternatively incorporate elements to filter
10 or modify the signals prior to their use by the transducers.

When a second individual, User B, perceives the transduced forms produced by User A DJ 200, he can then share the audio signal generated by the audio player 130 of Unit A, by use of the inter-unit transmitter/receiver 110 of Unit A and a compatible receiver 110 of Unit B. Audio signals received by Unit B from Unit A are played using the Unit B audio player
15 130, so that User A and User B hear the audio signals roughly simultaneously. There are a variety of means by which the Unit B can select the signal of Unit A, but a preferred method is for there to be a DJ directional identifier 122 in Unit B, which can be pointed at the DJ of User A and which receives information needed to select the Unit A signal from the User A DJ, whose transduced signal is perceptible to User B.

20 Given the audio signal now being exchanged between Unit A and Unit B, User A and User B can experience the same audio signal roughly simultaneously. Within the spirit of the present invention, it is preferable for the two users to hear the audio signals within 1 second of one another, and more preferable for the users to hear the audio signals within 200 milliseconds of one another, and most preferable for the users to hear the audio signals within
25 50 milliseconds of one another. Furthermore, DJs 200 being worn by User A and User B can receive signals from their respective units, each emitting perceptible forms of their signals. Preferably, the transduced forms expressed by the DJs 200 are such as to enhance the personal or social experience of the audio being played.

Unit 100 Structure

30 Units 100 comprise a device, preferably of a size and weight that is suited for personal wearing or transport, which is preferably of a size and format similar to that of a conventional portable MP3 player. The unit can be designed on a "base" of consumer electronics products

such as cell phones, portable MP3 players, or personal digital assistants (PDAs), and indeed can be configured as an add-on module to any of these devices.

In general, the unit 100 will comprise, in addition to those elements described in Figure 1, other elements such as a user interface (e.g. an LCD or OLED screen, which can be combined with a touch-sensitive screen, keypad and/or keyboard), communications interfaces (e.g. Firewire, USB, or other serial communications ports), permanent or removable digital storage, and other components.

The audio player 130 can comprise one or more modes of audio storage, which can include CDs, tape, DVDs, removable or fixed magnetic drives, flash memory, or other means. Alternatively, the audio can be configured for wireless transmission, including AM/FM radio, digital radio, or other such means. Output of the audio signal so generated can comprise wireless or wired headphones or wired or wireless external speakers.

It is also within the spirit of the present invention that the unit 100 can have only receive capabilities, without having separate audio information storage or broadcast capabilities. In concept, such a device can have as little user interface as an on/off button, a button to cause the unit 100 to receive signals from a new "host", and a volume control. Such devices can be very small and be built very inexpensively.

Unit 100 Audio Output

One of the goals of the present invention is to assist communications between groups of people. In general, with mobile audio devices, the music is listened to through headphones. Many headphones are designed so as to reduce to the extent possible the amount of sound which is heard from outside of the headphones. This, however, will have the general effect of reducing the verbal communications between individuals.

In order to avoid this potential problem, it is within the teachings of the present invention that headphones or earphones be provided that allow ambient sound, including a friend's voice, to be easily perceptible to the wearer of the headphones, and that such headphones can be provided that variably allow such sound to be accessible for the headphone's wearer. Such arrangement of the headphones can be obtained either through physical or electronic means. If through electronic means, the headphones can have a microphone associated with them, through which signals received are played back in proportion through the headphone speakers, said proportion being adjustable from substantially all sound being from the microphone to substantially no sound being from the microphone. This microphone can also be a part of a noise cancellation system, such that the

phase of the playback is adjustable -- if the phase is inverted relative to the ambient sound signal, then the external noise is reduced, whereas if the phase is coincident with the ambient sound signal, then the ambient sounds are enhanced.

Figure 10 is a top perspective view of an earphone 900 with adjustable external sound ports. A speaker element 940 is centrally located, and the outside circumferential surface is a rotatable sound shield 910 in which sound ports 930 are placed. The sound ports 930 are open holes to admit sound. Beneath the sound shield 930 is a non-rotatable sound shield in which fixed sound ports 920 are placed in a similar arrangement. As the sound shield 910 is rotated manually by the user, the sound ports 930 and the fixed sound ports 920 come into registration, so that open ports between sources of ambient noise and the outer ear chamber is created, increasing the amount of ambient sound that the user perceives.

Figures 11 A and B are cross-sectional diagrams of an earpiece with an extender 980 that admits additional ambient sound. In Figure 11A, the face of a speaker 960 with a cord 970 is covered with a porous foam block 950 that fits snugly into the ear. While some ambient sound is accessible to the ear through the foam block 950, the majority of the sound is input is impeded. In Figure 11B, the foam extender 980 is placed over the foam block 950 so that a formed shape at the distal end of the extender 980 fits snugly into the ear. A hollow cavity 982 can be allowed in the extender 980 so as to reduce the sound impedance from the speaker 960 to the ear. Ambient sound is allowed into the space between the speaker 960 and the distal end of the extender 980 (shown by the arrows).

Many other arrangements are allowed within the spirit of the present invention to allow ambient sound to more easily access the user's ear, including adjustable headphones or earplugs as in Figure 10, or accessories that can modify the structure of existing earphones and headphones, as in Figure 11B. Such effects can include increasing the number of apertures admitting ambient sound, increasing the size of an aperture (e.g. by adjusting the overlap between two larger apertures), changing the thickness or number of layers in the enclosure, or by placing a manually detachable cup that covers the earphone and ear channel so as to reduce ambient sound.

DJ 200 Transducers

DJs 200 will have a number of common elements, including communications elements, energy storage elements, and control elements (e.g. a manual ON/OFF switch or a switch to signal DJ entraining, as will be described below). In this section, the structure and function of transducers will be described.

Light Transducers 240

The DJ 200 transducers are used to create perceptible forms of the signals received by the receiver 220. Light transduction can include the use of one or more light-emitting devices, which can conveniently be colored LEDs, OLEDs, LCDs, or electroluminescent displays, which can be supplemented with optical elements, including mirrors, lenses, gratings, and optical fibers. Additionally, motors, electrostatic elements or other mechanical actuators can be used to mechanically alter the directionality or other properties of the light transducers 240. There can be either a single device or an array of devices, and if more than a single device, can display in synchrony, or can be "choreographed" to display in a temporal and/or spatial pattern.

Figure 2A is a schematic block diagram of a DJ 200 with multiple independently controlled LED arrays, wherein the number of LED arrays is preferably between 2 and 8, and is even more preferably between 2 and 4. The signal received from unit 100 via the DJ receiver 220 is passed to a multi-port controller 242 with two ports 294 and 296 connected respectively with two separate arrays 290 and 292 of LEDs 246. These arrays 290 and 292 can be distinguished by spatial placement, color of emitted light, or the temporal pattern of LED illumination. The signal is converted via analog or digital conversion into control signals for the two arrays 290 and 292, which are illuminated in distinct temporal patterns.

It should be noted that the signal received by receiver 220 from the unit 100 can comprise either a signal already in the form required to specify the array and temporal pattern of LED 246 activity, or it can alternatively be converted from a differently formatted signal into temporal pattern signals. For example, the unit 100 can transmit a modulated signal whose amplitude specifies the intensity of LED light amplitude. For multiple LED arrays, signals for the different arrays can be sent together and decoded by the DJ receiver 220, such as through using time multiplexing, or transmission on different frequencies.

Alternatively, the signal could be not directly related to the transduction intensity, such as in the direct transmission of the audio signal being played by the unit 100. In such case, the controller 242 can modify the signal so as to generate appropriate light transduction signals. For example, low frequency bandpass filters could provide the signals for the first array 290, whereas high-frequency bandpass filters could provide the signals for the second array 292. Such filtering could be accomplished by either analog circuitry or digital software within a microprocessor in the controller 242. It is also within the spirit of the present

invention for the different arrays to respond differently to the amplitude of the signal within a frequency band or the total signal.

An alternative control of LED arrays is presented in Figure 2B, a schematic block diagram of a DJ 200 with an LED array with independently controlled LEDs. In this case, the control signal received by the receiver 200 is passed through a single-port, multiple ID controller 243 to a single array of LEDs, each responsive only to signals with a particular characteristic or identifier. One or more of the LEDs 246 can have the same identifier or be responsive to the same characteristic so as to constitute a virtual array of LEDs.

As mentioned above, the transduced light signal can alternatively or additionally comprise multi-element arrays, such as an LED screen. In such case, the signal received by the receiver 220 can be either a specification of image elements to be displayed on the LED screen, or can be as before, a signal unrelated to the light transduction output. For example, many audio players on computers (e.g. Windows Media player) come with pattern generators that are responsive to the frequency and amplitude of the audio signal. Such pattern generators could be incorporated into the controllers 242 or 243.

Alternatively, the light transducer 240 can be a single color illuminated panel, whose temporal pattern of illumination was similar to that of the LEDs of Figures 2A and 2B. In such case, users can partially cover the panel with opaque or translucent patterns, such as a dog or a skull or a representation of a favorite entertainer.

Whereas the receiver 220 and the light controllers 242 or 243 can be hidden from view, either behind the light transducers or separated from the transducers by a wire, for example, the light transducers are meant to be perceptible to other people. For this purpose, the light transducers can be fashioned into fashion accoutrements such as bracelets, brooches, necklaces, pendants, earrings, rings, hair clips (e.g. barrettes), ornamental pins, netting to be worn over clothing, belts, belt buckles, straps, watches, masks, or other objects. Additionally, the light transducers can be fashioned into clothing, such as arrays of lighting elements sewn onto the outside of articles of clothing such as backpacks, wallets, purses, hats, or shoes. For those articles of clothing that are normally washed, however, the lighting transducers and associated electronics will preferably be able to withstand cleaning agents (e.g. water or dry cleaning chemicals), or will be used in clothing such as scarves and hats that do not need to be washable.

It is also convenient for there to be modular lighting arrangements in which the configuration can easily changed by a user. One example of such a modular arrangement is a light pipe made of a flexible plastic cable or rod, at one or both ends of which is positioned

a light source that directs light into the rod. At predetermined locations along the rod, the rod surface can be roughened so as to allow a certain amount of light to escape, on which transparent glass or plastic pieces can be clipped, and that are lighted when the pipe is lighted. Alternatively, the light can be uniformly smooth, and transparent pieces of roughly
5 index of refraction matching material can be clipped onto the rod, allowing some fraction of the light to be diverted from the rod into the pieces. The light sources and associated energy sources used in such an arrangement can be relatively bulky and be carried in a backpack, pouch or other carrying case, and can brightly illuminate a number of separate items.

It should be noted that the transducers require an energy store 270, which is
10 conveniently in the form of a battery. The size of the battery will be highly dependent on the transduction requirements, but can conveniently be a small "watch battery". It is also convenient for the energy store 270 to be rechargeable. Indeed, all of the electric devices of the present invention will need energy stores or generators of some sort, which can comprise non-rechargeable batteries, rechargeable batteries, motion generators that can convert energy
15 from the motion of the user into electrical energy that can be used or stored, fuel cells or other such energy stores or converters as are convenient.

Sound Transducers 260

Sound transducers 260 can supplement or be the primary output of the audio player of the unit 100. For example, the unit 100 can wirelessly transmit the audio signal to DJ 200
20 comprising a wireless headphone sound transducer. This would allow a user to listen to the audio from the audio player without the need for wires connecting the headphones to the unit 100. Such sound transducers can comprise, for example, electromagnetic or piezoelectric elements.

Alternative to headphone or earphone audio production, external speakers, which can
25 be associated with light transducers 240 or tactile transducers 250, can be used to enhance audio reproduction from external speakers associated with the unit 100. In addition or alternative to simple reproduction of the audio signal output by the audio player 130, the sound transducers 260 can play modified or accompanying signals. For example, frequency filters can be used to select various frequency elements from the music (for low bass), so as
30 to emphasize certain aspects of the music. Alternatively, musical elements not directly output from the audio player 130 can be output to complete all instrumental channels of a piece of music, for example.

Tactile Transducers 250

DJs 200 can be configured with tactile transducers, which can provide vibrational, rubbing, or pressure sensation. As before, signals of a format that control these transducers can be sent directly from the DJ transmitter 120, or can be filtered, modified or generated from signals of an unrelated format that are sent from the transmitter 120. As before, the signal can be the audio signal from the audio player 130, which can, for example, be frequency filtered and possibly frequency converted so that the frequency of tactile stimulation is compatible with the tactile transducer. Alternatively, signals that are of the sort meant for light transduction can be modified so as to be appropriate for tactile transduction. For example, signals for light of a particular color can be used to provide vibrational transduction of a particular frequency, or light amplitudes can be converted into pressure values.

The tactile transducer can comprise a pressure cuff encircling a finger, wrist, ankle, arm, leg, throat, forehead, torso, or other body part. The tactile transducer can alternatively comprise a rubbing device, with an actuator that propels a tactile element tangentially across the skin. The tactile transducer can also alternatively comprise a vibrational device, with an actuator that drives an element normally to the skin. The tactile transducer can further alternatively comprise elements that are held fixed in relation to the skin, and which comprise moving internal elements that cause the skin to vibrate or flex in response to the movement of the internal element.

The tactile transducer can lack any moveable element, and can confer tactile sensation through direct electrical stimulation. Such tactile elements are best used where skin conductivity is high, which can include areas with mucus membranes.

Tactile transduction can take place on any part of the body surface with tactile sensation. In addition, tactile transduction elements can be held against the skin overlying bony structures (skull, backbone, hips, knees, wrists), or swallowed and conveyed through the digestive tract, where they can be perceived by the user.

Input Transducers

It should also be understood that the DJ 200 can comprise input transducers in order to create control signals from information or stimuli in the local environment. Figure 24 is a schematic block diagram of a DJ unit 200 with associated input transducers. The input-enabled DJ 1320 comprises energy storage 270, a controller 1322, output transducers 1324, a

DJ receiver 220 and input transducers 1326. The input transducers 1326 can comprise one or more of a microphone 1328 and an accelerometer 1330.

In operation, the energy storage 270 provides energy for all other functions in the DJ 1320. The controller 1322 provides control signals for the output transducers 1324, which
5 can comprise tactile transducers 250, sound transducers 260, and/or light transducers 240. Input to the controller can be provided via the input transducers 1326, optionally along with input from the DJ receiver 220.

For example, on a dance floor, the microphone 1328 can provide electrical signals corresponding to the ambient music. These signals can be converted into transducer 1324
10 control signals in a manner similar to that described below for the automatic generation of control signals according to Figures 21 A-C, as will be described below. This allows the use of the DJ functionality in the absence of an accompanying audio unit 100, expanding the applications of the DJ 200. An automatic gain filter can be applied so as to compensate for the average volume level – because the user can be close or far from the sources of ambient
15 music and the music can vary in volume, the strength of the DJ 200 transduction can be normalized. In addition, it can also be preferable for there to be a manual amplitude control 1323, such as a dial or two position rocker switch, by which the average intensity of the DJ 200 control signals can be varied to suit the taste of the user. The amplitude control 1323 can operate through modulating the input transducer 1326 output or as an input to the controller
20 1322 as it generates the signals for the output transducers 1324.

Alternatively, the accelerometer 1330 can track the movement of the person wearing the DJ 100, such that a signal indicating acceleration in one direction can be converted by the controller 1322 into signals for a channel of output transducers 1324. The accelerometer
25 1330 can be outfitted with sensors for monitoring only a single axis of motion, or alternatively for up to three independent directions of acceleration. Thus, the controller 1322 can convert sensed acceleration in each direction into a separate channel, horizontal axes of acceleration could be combined into a single channel and the vertical axis into a second channel, or other such linear or non-linear combination of sensed acceleration can be combined in aesthetic fashion.

30 It is also within the spirit of the present invention that multiple input signals be combined by the controller 1322 to create control signals for aesthetic output from the output transducers 1324. For example, one channel can be reserved for control signals generated from accelerometer signals, another channel for control signals generated from microphone signals, and yet a third channel from control signals generated from DJ receiver 220 input. In

general, the information from the DJ receiver 220 and from the microphone 1328 will be of the same type (i.e. generated from audio signals), so that the most common configurations will be control signals from a combination of the microphone 1328 and accelerometer 1330, and signals from a combination of the DJ receiver 220 and the accelerometer 1330.

5 The input transducers 1326 can further comprise a light sensor, such that the DJ would mimic light displays in its environment, making it appear that the DJ is part of the activity that surrounds it. In this case, the controller 1322 would preferably generate control signals based on rapid changes in the ambient lighting, since it would be less aesthetic to have the DJ transducers provide constant illumination. Furthermore, slowly changing light
10 (on the order of tens or hundreds of milliseconds) will be created naturally by the movement of the user, whereas changes in the lighting (e.g. strobes, laser lights, disco balls) will be of much faster change (on the order of milliseconds). Thus, to match the ambient dance lighting, it is aesthetic for the DJ 200 to respond most actively to ambient light that is changing in intensity a predetermined percentage in a predetermined time, wherein the
15 predetermined percentage is at least 20% and the predetermined time is 20 milliseconds or less, and even more preferably for the predetermined percentage to be at least 40% and the predetermined time is 5 milliseconds or less.

Unit to Unit Communication

Units 100 transfer audio signals from the audio player in one unit 100 to the audio
20 player 130 of another unit 100. Figures 3A-C are schematic block diagrams of unit 100 elements used in inter-unit communications. Each diagram presents communications between a Unit A and a Unit B, with Unit A transmitting audio signals to Unit B. Dashed connectors and elements indicate elements or transfers that are not being utilized in that unit 100, but are placed to indicate the equivalence of the transmitting and receiving units 100.

25 In Figure 3A, compressed audio signals (e.g. in MP3 format or MPEG4 format for video transfers, as described below) stored in a compressed audio storage 310 are transferred to a signal decompressor 302, where the compressed audio signal is converted into an uncompressed form suitable for audio output. In Unit A, this decompressed signal is passed both to the local speaker 300, as well as to the inter-unit transmitter/receiver 110. The Unit B
30 inter-unit transmitter-receiver 110 receives the uncompressed audio signal, which is sent to its local speaker for output. Thus, both Unit A and Unit B play the same audio from the Unit A storage, in which uncompressed audio is transferred between the two units 100.

In Figure 3B, compressed audio signals from the Unit A compressed audio storage 310 are sent both to the local signal decompressor 302 and to the inter-unit transmitter/receiver 110. The Unit A decompressor 302 conditions the audio signal so that it is suitable for output through the Unit A speaker 300. The compressed audio signal is sent
5 via Unit A transmitter-receiver 110 to the Unit B transmitter/receiver 110, where it is passed to the Unit B decompressor 302 and thence to the Unit B speaker 300. In this embodiment, because compressed audio signals are transmitted between the units 100 transmitter/receivers 302, lower bandwidth communications means can be used in comparison with the embodiment of Figure 3A.

10 In Figure 3C, compressed audio signals from the Unit A compressed audio storage 310 are sent to the Unit A signal decompressor 302. These decompressed signals are sent to both the local speaker 300 as well as to a local compressor 330, which recompresses the audio signal to a custom format. In addition to decompressed audio signal input, the compressor also optionally utilizes information from a DJ signal generator 320, which
15 generates signals to control DJ transducers 240, 250 and 260, which can be sent in conjunction with the audio signal. The signal generator 320 can include analog and/or digital filtering or other algorithms that analyze or modify the audio signals, or can alternatively take manually input transducer control signals input as described below. The custom compression can include multiplexing of the audio signals with the transducer control signals.

20 The custom compressed audio signals, are then passed to the Unit A inter-unit transmitter/receiver 110, which are then transferred to the Unit B inter-unit transmitter/receiver 110, and thence to the Unit B signal decompressor 302 and speaker 300.

Given the time delays in signal transfer between the units 100, custom compression that takes place in the sending unit, and any subsequent decompression that takes place in the
25 receiving unit 100, it can be convenient to place a delay on the local (i.e. Unit A) speaker output of tens of milliseconds, so that both units 100 play the audio through their speakers at roughly the same time. This delay can include limited local digital storage between the local signal decompression and speaker 300 output.

Various hardware communications protocols will be discussed below with respect to
30 unit-to-unit communications, but in general it is required that the distance between the units that must be maintained be preferably at least 40 feet, and more preferably at least 100 feet, and most preferably 500 feet, in order to allow units 100 sharing music to be able to move reasonably with respect to one another (e.g. for a user to go to the bathroom without losing contact), or to find each other in a large venue such as a shopping mall.

Communications Protocols

Communication between the inter-unit transmitter/receivers 110 can involve a variety of protocols within the teachings of the present invention, and can include IP protocol-based transmissions mediated by such physical link layers as 802.11a, b or g, WDCT, HiperLAN, ultra-wideband, 2.5 or 3G wireless telephony communications, custom digital protocols such as Bluetooth or Millennial Net i-Beans. Indeed, it is not even necessary for the transmissions to be based on Internet protocol, and conventional analog radio-frequency or non-IP infrared transmissions are also within the spirit of the present invention. Each unit 100 will generally have both transmission and reception capabilities, though it is possible for a unit to have only reception capabilities. While the bandwidth of the broadcast is dependent on the compression of the audio signal, it is preferable for the transmission bandwidth to be larger than 100 kb/sec, and even more preferable for the transmission bandwidth to be greater than 250 kb/sec.

While the distance of transmission/reception is not bounded within the teachings of the present invention, it will generally be less than a few hundred meters, and often less than 50 meters. The distance of communication is limited in general by the amount of power required to support the transmission, the size of antennae supported by portable devices, and the amount of power allowed by national regulators of broadcast frequencies. Preferably, however, the range of transmission will be at least 10 meters, and even more preferably at least 30 meters, in order to allow people sharing communications to move some distance from one another without communications being lost.

The unit 100 is characterized generally by four sets of roughly independent characteristics: playing audio or not playing audio, transmitting or not transmitting, receiving or not receiving, searching or not searching.

Units 100 will often function in conditions with large numbers of other units 100 within the communications range. For example, in a subway car, a classroom, bicycling, or at a party, a unit 100 can potentially be within range of dozens of other units. A unit 100 that is playing audio from local compressed audio storage 310 can, at the user's prerogative, choose to broadcast this audio to other units 100. A unit 100 that is currently "listening" to a broadcast or is searching for a broadcast to "listen" to will require a specific identifier roughly unique to a broadcaster in order to select that broadcaster signal from among the other possible broadcasters. Some of the communications protocols listed above, such as those based on IP protocols, 2.5G or 3G wireless, or Bluetooth communications, have such

identifiers as part of the protocols. Custom radio frequency based protocols will require protocols to allow signals to be tagged with specific identifiers.

A unit 100 that is transmitting signals can, within the spirit of the present invention, be prevented from simultaneously receiving signals. Preferably, however, units 100 can both
5 transmit and receive simultaneously. One example of the use of simultaneous transmission and reception is for a unit 100 that is receiving a signal to send a signal indicating its reception to the transmitting unit 100. This allows the transmitting unit to determine the number of units 100 that are currently receiving its broadcast. In return, this information could be sent, along with the audio signal, so that all of the users with units 100 receiving the
10 broadcast can know the size of the current reception group. Alternatively, a user with a unit 100 that is currently broadcasting can be searching for other broadcasting units, so that the user can decide whether to continue broadcasting or whether to listen to the broadcast of another unit.

Unit to DJ Communication

15 Communication between the unit 100 and the DJ 200 can be either through the inter-unit transmitter/receiver 110, or through a separate system. In general, the requirement of the DJ 200 is for reception only, although it is permissible for the DJ 200 to include transmission capabilities (e.g. to indicate to the unit 100 when the DJ 200 energy storage 270 is near depletion).

20 The signals for which the DJ 200 is receptive is dependent on how the transduction control signals are generated. For example, for a controller 242 that incorporates a filter or modifier that takes the audio signal as its input, the DJ receiver 220 would receive all or a large fraction of the audio signal. In this case, the communication between the unit 100 and the DJ 200 would require a bandwidth comparable to that of inter-unit communication, as
25 described above.

However, if the signals are either generated in the unit 100, or pre-stored along with the stored compressed audio signal, then the communications bandwidth can be quite modest. Consider a DJ 200 with 2 arrays 290 and 292 of LEDs 246, which flash with a frequency of no more than 10 Hertz, and that the LEDs are in either an ON or an OFF state, without
30 intermediate amplitudes. In such case, the maximum bandwidth required would be only 20 bits/second, in addition to the DJ control signals.

The range of unit to DJ communications need not be far. In general, the unit 100 and the DJ 200 will be carried by the same user, so communications ranges of 10 feet can be

adequate for many applications. Some applications (see below) can require, however, somewhat larger ranges. On the other hand, longer communications ranges will tend to confer the possibility of overlap and interference between two different units 100 to their respective DJs 200. In general, for the application of unit to DJ communications, it is preferable for the minimum range of communications to be at least 1 foot, and more preferably for the minimum range of communications to be at least 10 feet, and most preferably for the minimum range of communications to be at least 20 feet. Also, for the application of unit to DJ communications, it is preferable for the maximum range of communications to be no more than 500 feet, and more preferably for the maximum range of communications to be no more than 100 feet, and most preferably for the maximum range of communications to be no more than 40 feet. It should be noted that these communications ranges refer primarily to the transmission distance of the units 100, especially with regard to the maximum transmission distance.

Because there can be multiple unit 100/DJ 200 ensembles within a relatively short distance, communications between a unit 100 and a DJ 200 preferably comprise both a control signal as well as a unit identification signal, so that each DJ 200 receives its control signals from the correct unit 100. Because the unit 100 and the DJ 200 will not, in general, be purchased together, or that a user can buy a new unit 100 to be compatible with already owned DJs 200, it is highly useful to have a means of "entraining" a DJ 200 to a particular unit 100, called its "master unit", and a DJ 200 entrained to a master unit is "bound" to that unit.

Figure 4 is a schematic flow diagram of DJ entraining. To entrain a DJ 200, the DJ is set into entraining mode, preferably by a physical switch on the DJ 200. The master unit 100 to which the DJ 200 is to be entrained is then placed within communications range, and the unit 100 transmits through the DJ transmitter 120 an entraining signal that includes the master unit 100 identifier. Even should there be other units 100 transmitting in the vicinity, it is unlikely that they would be transmitting the entraining signal, so that entraining can often take place in a location with other active units 100. Verification that the entraining took place can involve a characteristic sequence of light output (for light transduction), audio output (for sound transduction) or motion (for tactile transduction). After verification, the DJ 100 is reset to its normal mode of operation, and will respond only to control signals accompanied by the identifier of its master unit 200.

It should be noted that there can be multiple DJ's 200 bound to the same master unit 100. Thus, a single person can have multiple light transducing DJs 200, or DJs 200 of various modes (light, sound, tactile) transduction.

While DJs 200 will generally be bound to a master unit associated with the same person, this is not a requirement of the present invention. Figures 5A-B are schematic block diagrams of DJs 200 associated with multiple people bound to the same master unit. In Figure 5A, DJ A 200 and DJ B 200 are both bound to the same DJ transmitter 120, even though DJ A 200 and DJ B 200 are carried by different persons. This is particularly useful if the control signals are choreographed manually or through custom means by one person, so that multiple people can then share the same control signals. Such a means of synchronization is less necessary if the DJ 200 control signals are transmitted between units 100 through the inter-unit transmitter/receiver 110 along with the audio signals. Furthermore, in this case, it is better for the range of unit-to-DJ communication to be in the range of the inter-unit communication described above.

In the case of sound transducers 260, the DJ B 200 can comprise a wireless audio earpiece, allowing users to share music, played on a single unit 100, privately. Consider Figure 5A, configured with sound transducers 260 (see, for example, Figure 1) in DJ A 200 and DJ B 200. Signals from the audio player 130 are transmitted by the DJ transmitter 120, where they are received by DJs 200 – DJ A and DJ B – that are carried by Person A and Person B, respectively. In this case, both persons can listen to the same music.

Figure 5B shows the operation of a wide-area broadcast unit 360, which is used primarily to synchronize control of a large number of DJs 200, such as might happen at a concert, party or rave. In this case, the audio player 130 is used to play audio to a large audience, many of whom are wearing DJs 200. In order to synchronize the DJ output, a relatively high-power broadcast transmitter 125 broadcasts control signals to a number of different DJs 200 carried by Person A, Person B and other undesigned persons. The entraining signal can be automatically sent on a regular basis (e.g. whenever music is not being played, such as between songs, or interspersed within compressed or decompressed songs) so that patrons or partygoers could entrain their DJs 200 to the broadcast unit 360. The broadcast unit 360 can also transmit inter-unit audio signals, or can only play the audio through some public output speaker that both Person A and Person B can enjoy.

Figure 26 is a schematic diagram of people at a concert, in which DJs 200 conveyed by multiple individuals are commonly controlled. At a concert venue 1370, music is produced on a stage 1372, and concert patrons 1376 are located on the floor of the venue.

Many of the patrons have DJs 200 which are receptive to signals generated by a broadcast DJ controller 1374. The broadcast DJ controller creates signals as described below, in which the music is automatically converted into beats, where microphones are used to pick up percussive instruments, and/or where individuals use a hand-pad to tap out control signals.

5 These control signals are either broadcast directly from the area of the broadcast DJ controller 1374, or alternatively are broadcast from a plurality of transmitters 1380 placed around the venue 1370, and which are connected by wires 1378 to the controller 1374 (although the connection can also be wireless within the spirit of the present invention). It should be understood that the protocol for transmitting DJ control signals can be limited either by
10 hardware requirements or by regulatory standards to a certain distance of reception. Thus, to cover a sufficiently large venue, multiple transmitters can be necessary to provide complete coverage over the venue 1370. In general, it is preferable for the maximum transmission distance of transmission from the transmitters to be at least 100 feet, and more preferably at least 200 feet, and most preferably at least 500 feet, so as to be able to cover a reasonable
15 venue 1370 size without needing too many transmitters 1380.

An alternative embodiment of unit 100 to DJ 200 communications is the use of radio frequency transmitters and receivers, such as those used in model airplane control, which comprise multi-channel FM or AM transmitters and receivers. These components can be very small (e.g. the RX72 receivers from Sky Hooks and Riggings, Oakville, Ontario,
20 Canada), and are defined by the crystal oscillators that determine the frequency of RF communications. Each channel can serve for a separate channel of DJ control signals. In such cases, an individual can place a specific crystal in their audio unit 100, and entraining the DJ 200 is then carried out through the use of the same crystal in the DJ 200. Because of the large number of crystals that are available (e.g. comprising approximately 50 channels in
25 the model aircraft FM control band), interference with other audio units 100 can be minimized. Furthermore, control of many DJs 200 within a venue, as described above, can take place by simultaneously transmitting over a large number of frequencies.

As described above, the wide-area broadcast transmitter 125 can transmit entraining signals to which the DJs 200 can be set to respond. However, there are a number of other
30 preferred means by which DJs 200 can be used to respond to control signals to which they have not been entrained. For example, the DJs 200 can be set to respond to controls signals to which they have not been entrained should there be no entrained control signals present (e.g. the corresponding unit 100 is not turned on).

Figure 35 is a schematic block diagram of DJ 200 switch control for both entraining and wide-area broadcast. The DJ 200 comprises a three-way switch 1920. In a first state 1922, the DJ 200 is entrained to the current control signal as described above. Thereafter, in a second state 1924, the DJ 200 responds to control signals corresponding to the entraining signal encountered in the step 1922. In a third state 1926, the DJ 200 responds to any control signal for which its receiver is receptive, and can therefore respond to a wide-area broadcast, thereby providing the user with manual control over the operational state of the DJ 200. It should be noted that the switch 1920 can be any physical switch with at least three discreet positions, or can alternatively be any manual mechanism by which the user can specify at least three states, including a button presses that have a visible user interface or a voice menu.

Figure 12B is a schematic drawing of modular digital jewelry 201. The modular jewelry 201 is comprised of two components: an electronics module 1934 and a display module 1932. These modules 1934 and 1932 can be electrically joined or separated through an electronics module connector 1936 and a display module connector 1938. The value of the modular arrangement is that the electronics module 1934 comprises, in general, relatively expensive components, whose combined price can be many-fold that of the display module 1932. Thus, if a user wants to change the appearance of the jewelry 201 without having to incur the cost of additional electronics components such as the energy storage 270, receiver 220 or controller 1322, they can simply replace the display module 1932 with its arrangement of output transducers 1324 with an alternative display module 1933 with a different arrangement of output transducers 1325.

The transmitter for DJ 200 control signals has been previously discussed primarily in terms of its incorporation within a unit 100. It should be understood, however, that the transmitter can be used in conjunction with a standard audio player unrelated to unit-to-unit communications. Figure 12C is a schematic block diagram of a modular digital jewelry transmitter 143 that generates and transmits control signals from an audio player 131. The modular transmitter 143 is connected to the audio player 131 via audio output port 136 through the cable 134 to the audio input port 138 of the modular transmitter 143. The modular transmitter 143 comprises the DJ transmitter 120, which can send unit-to-DJ communications. The output audio port 142 is connected to the earphone 901 via cable 146. The earphone 901 can also be a wireless earphone, perhaps connected via the DJ transmitter 120.

The audio output from the player 131 is split both to the earphone 901 and to the controller 241 (except, perhaps where the DJ transmitter transmits to a wireless earphone).

The controller 241 automatically generates control signals for the DJ 200 in a manner to be described in detail below. These signals are then conveyed to the DJ transmitter 120. It should be understood that this arrangement has the advantage that the digital jewelry functionality can be obtained without the const of the components for the audio player 131, and in addition, that the modular transmitter 143 can then be used in conjunction with multiple audio players 131 (either of different types or as the audio players are lost or broken).

Inter-unit Audio Sharing

Overview

Inter-unit communication involves the interactions of multiple users, who may or may not be acquaintances of each other. That is, the users can be friends who specifically decide to listen to music together, or it can be strangers who share a transient experience on a subway train. The present invention supports both types of social interaction.

An important aspect of the present invention is the means by which groups of individuals join together. Figure 6 is a schematic block diagram of a cluster 700 of units 100, indicating the nomenclature to be used. The cluster 700 is comprised of a single broadcast unit 710, and its associated broadcast DJ 720, as well as one or more receive units 730 and their associated DJs 740. The broadcast unit 710 transmits music, while the receive unit 730 receives the broadcasted music. A search unit 750 and its associated search DJ 760 are not part of the cluster 700, and comprise a unit 100 that is searching for a broadcast unit 710 to listen to or a cluster 700 to become associated with.

It should be noted that many communications systems can be operated alternatively in two modes: one that supports peer-to-peer communications and one that requires a fixed infrastructure such as an access point. Figure 35 is a schematic block diagram of mode switching between peer-to-peer and infrastructure modes. A mode switch 1950 is made by the user, either manually, or automatically – for example, that the user chooses between different functions (listening or broadcasting, file transfers, browsing the Internet) and the system determine the optimal mode to use. A peer-to-peer mode 1952 is well configured for mutual communications between mobile units 100 that are within a predetermined distance, and is well-suited for short-range wireless communications and audio data streaming 1954. Alternatively, the mode switch 1950 enables an infrastructure mode 1956, which is of particular usefulness in gaining access to a wide area network such as the Internet, through

which remote file transfer 1958 (e.g. downloading and uploading) and remote communications such as Internet browsing can be made through access points to the fixed network.local wireless audio streaming.

It should be noted, however, that certain communications systems, such as many modes of telephony, do not distinguish between mobile communications and communications through fixed access points, and that both file transfer 1958 and audio streaming 1954 can be available through the same mode. Even in those cases, however, it can be convenient to have two modes in order to make optimal use of the advantages of the different modes. In such cases, however, the two modes can alternatively be supported by multiple hardware and software systems within the same device – for example, for remote communications to be made through a telephony system (e.g. GSM or CDMA), while the local audio streaming 1954 can be made through a parallel communications system (e.g. Bluetooth or 802.11) – indeed, the two systems can operate simultaneously with one another.

Inter-unit Transmission Segmentation

Preferably, the broadcast unit 710 and the receive units 730 exchange information in addition to the audio signal. For example, each user preferably has indications as to the number of total units (broadcast units 710 and receive units 730) within a cluster, since the knowledge of cluster 700 sizes is an important aspect of the social bond between the users. This also will help search units 750 that are not part of the cluster determine which of the clusters 700 that might be within their range are the most popular.

The additional information shared between members of a cluster 700 would include personal characteristics that a person might allow to be shared (images, names, addresses, other contact information, or nicknames). For example, the broadcast unit 710 will preferably, along with the music, transmit their nickname, so that other users will be able to identify the broadcast unit 710 for subsequent interactions, and a nickname is significantly easier to remember than a numerical identifier (however, such numerical identifier can be stored in the unit 100 for subsequent searching).

Such additional information can be multiplexed along with the audio signal. For example, if the audio signal is transferred as an MP3 file, assuming that there is additional bandwidth beyond that of the MP3 file itself, the file can be broken into pieces, and can be interspersed with other information. Figure 7 is a schematic diagram of a broadcast unit 710 transmission 820. The transmission is comprised of separate blocks of information, each represented in the figure as a separate line. In the first line, a block code 800 is transmitted,

which is a distinctive digital code indicating the beginning of a block, so that a search unit 750 receiving from the broadcast unit 710 for the first time can effectively synchronize itself to the beginning of a digital block. Following the block code 800 is a MP3 block header 802, which indicates that the next signal to be sent will be from a music file (in this case an MP3 file). The MP3 block header 802 includes such information as is needed to interpret the following block of MP3 file block 804, including such information as the length of the MP3 block 804, and characteristics of the music (e.g. compression, song ID, song length, etc.) that are normally located at the beginning of a MP3 file. By interspersing this file header information at regular intervals, a user can properly handle music files that are first received in the middle of the transmission of an MP3 file. Next, the MP3 block 804 containing a segment of a compressed music file is received.

Dependent on the amount of music compression and the bandwidth of the inter-unit communications, other information can be sent, such as user contact information, images (e.g. of the user), and personal information that can be used to determine the "social compatibility" of the user with the broadcast unit 710 and the receive unit 730. This information can be sent between segments of MP3 files or during "idle" time, and is generally preceded by a block code 800, that is used to synchronize transmission and reception. Next, a header file is transmitted, which indicates the type of information to follow, as well as characteristics that will aid in its interpretation. Such characteristics could include the length of information, descriptions of the data, parsing information, etc. In Figure 7, an ID header 806 is followed by an ID block 808, which includes nicknames, contact information, favorite recording artists, etc. Later, an image header 810 can be followed by an image block with an image of the user. The image header 810 includes the number of rows and columns for the image, as well as the form of image compression.

It should be understood that the communications format described in Figure 7 is only illustrative of a single format, and that a large number of different formats are possible within the present invention. Also, the use of MP3 encoding is just an example, and other forms of digital music encoding are within the spirit of the present invention, and can alternatively comprise streaming audio formats such as Real Audio, Windows Media Audio, Shockwave streaming audio, QuickTime audio or even streaming MP3 and others. Furthermore, these streaming audio formats can be modified so as to incorporate means for transmitting DJ 200 control signals and other information.

Transmitting Dynamic Data and Control Information

As described above, there are benefits to two-way communications between the broadcast unit 710 and the receive unit 730. There are many methods of carrying out this communication, even if the inter-unit transmitter/receiver 110 does not permit simultaneous transmission and reception. For example, additional transmission and reception hardware could be included in each unit 100. Alternatively, in the transmission 820 above, specific synchronization signals such as the block code 800 can be followed by specific intervals during which the inter-unit transmitter/receiver 110 that is transmitting switches into receive mode, while the inter-unit transmitter/receiver 110 that was receiving switches to transmit mode. This switch in communications direction can be for a specific interval, or can be mediated through conventional handshake methods of prior art communications protocols.

It should be noted that in addition to transfer of static information (e.g. identifiers, contact information, or images), dynamic information and control information can also be transferred. For example, the user at the receive unit 730 can be presented with a set of positive and negative comments (e.g. "Cool!" "This is awful!") that can be passed back to the broadcast unit 710 with the press of a button. Such information can be presented to the user of the broadcast unit 710 either by visual icon on, for example, an LCD screen, by a text message on this screen, or by artificial voice synthesis generated by the broadcast unit 710 and presented to the user in conjunction with the music.

Alternatively, the user of the receive unit 730 can speak into a microphone that is integrated into the receive unit 730, and the user voice can be sent back to the broadcast unit 710. Indeed, the inter-unit communications can serve as a two-way or multi-way communications method between all units 100 within range of one another. This two-way or multi-way voice communication can be coincident with that of the playing of the audio entertainment, and as such, it is convenient for there to be separate amplitude control over the audio entertainment and the voice communication. This can be implemented either as two separate amplitude controls, or alternatively as an overall amplitude control, with a second control that sets the voice communications amplitude as a ratio to that of the audio entertainment. In this latter mode, the overall level of audio output by the unit is relatively constant, and the user then selects only the ability to hear the voice communication over the audio entertainment.

In order to express their feelings and appreciation about the music they are hearing, users within a cluster 700 can also press buttons on their units 100 that will interrupt or

supplement the control signals being sent to their respective DJs 200, providing light shows that can be made to reflect their feelings. For example, it can be that all lights flashing together (and not in synchrony with the music) can express dislike for music, whereas intricate light displays could indicate pleasure.

5 It is also possible to send control requests between units 710. For example, a receive unit 730 can make song requests (e.g. "play again", "another by this artist") that can show on the broadcast unit 710 user interface. Alternatively, the user of a receive unit 730 can request that control be switched, so that the receive unit 730 becomes the broadcast unit 710, and the broadcast unit 710 becomes a receive unit 730. Such requests, if accepted by the initial
10 broadcast unit 710 user, will result in the memory storage of the identifier of the broadcast unit 710 being set in all units in the cluster 700 to that of the new broadcast unit 730. Descriptions of the communications resulting in such a transfer of control will be provided below.

Additionally, it is also possible for users of units 100 to privately "chat" with other
15 users while they are concurrently receiving their audio broadcasts. Such chat can be comprised of input methods including keyboard typing, stylus free-form writing/sketching, and quickly selectable icons.

It should be understood that within the spirit of the present invention that the functional configuration can be supported by the extension of certain existing devices. For
20 example, the addition of certain wireless transmitter and receiver, as well as various control and possibly display functionality to a portable audio player would satisfy some embodiments of the present invention. Alternatively, by the addition of music storage and some wireless transmitter and receiver functionality, a mobile telephone would also allow certain embodiments of the present invention. In such case, the normal telephony communications,
25 perhaps supported by expanded 3G telephony capabilities, could serve to replace aspects of the IP communications described elsewhere in this specification.

IP Socket Communication Embodiments

A standard set of protocols for inter-unit communications is provided through IP socket communications, which is widely supported by available wireless communications
30 hardware, including 802.11a, b and g (Wi-Fi). An embodiment of inter-unit communications is provided in Figures 14A-B. Figure 14A is a schematic block diagram of the socket configurations on the broadcast unit 710 and the receive unit 730.

In the discussion below, transfer of the different messages and audio information are provided, generally but not always, through an Internet protocol. At the transport layer of such protocols, there will generally be used either a connectionless protocol or a connection-oriented protocol. Among the most common of these protocols are respectively the User
5 Datagram Protocol (UDP) and the Transmission Control Protocol (TCP), and wherever these protocols are used below, it should be noted that any like protocol (connectionless or connection-oriented), or the entire class of protocol can generally be substituted in the discussion.

The broadcast unit 710, prior to the membership of the receive unit 730, broadcasts
10 the availability of the broadcast on a broadcast 1050, which is generally a TCP socket. The annunciator 1050 broadcasts on a broadcast address with a predetermined IP address and port. The receive unit 730 has a client message handler 1060 that is also a TCP socket that is looking for broadcasts on the predetermined IP address and port. When it receives the broadcast, a handshake creates a private server message handler 1070 on a socket with a new
15 address and port on the broadcast unit 710. The broadcast unit 710 and the receive unit 730 can now exchange a variety of different messages using the TCP protocol between the server message handler 1070 and the client message handler 1060. This information can comprise personal information about the users of the broadcaster unit 710 and the receive unit 730. Alternatively or additionally, the broadcast unit 710 can transfer a section of the audio signal
20 that is currently being played, so that the user of the receive unit 730 can "sample" the music that is being played on the broadcast unit 710. It should be noted that, in general, the broadcast unit 710 continues its broadcast on the broadcast annunciator 1050 for other new members.

Once it is established that the broadcast unit 710 and the receiver unit 730 are
25 mutually desirous of providing and receiving an audio broadcast, respectively, sockets optimized for broadcast audio are created both on the broadcast unit 710 and the receiver unit 730. These sockets will often be UDP sockets – on the broadcast unit 710, a multicast out socket 1080 and on the receiver unit 730, a multicast in socket 1090.

Figure 14B is a schematic block flow diagram of using IP sockets for establishing
30 and maintaining communications between a broadcast unit 710 and the receive unit 730, according to the socket diagram of Figure 14A. In a step 1100, the broadcast annunciator 1050 broadcasts the availability of audio signals. In a step 1102, the receiver unit 730 searches for a broadcast annunciator 1050 on the client message handler 1060 socket. Once a connection is initiated in a step 1104, the broadcast unit 710 creates the message handler

socket 1070 in a step 1106, and the receiver unit 730 retasks the message handler socket 1060 for messaging with the broadcast unit 730. The broadcast annunciator 1050 continues to broadcast availability through the step 1100.

5 In a step 1110, the broadcast unit 710 and the receiver unit 730 exchange TCP messages in order to establish the mutual interest in audio broadcasting and reception. Should there not be mutual acceptance, then the system returns to the original state in which the broadcast unit 710 is transmitting the broadcast annunciation in the step 1100, and the receive unit 730 searches for broadcasts in the step 1102. Given that the receive unit 730 and the broadcast unit 710 will be within communications distance, and that the broadcast unit
10 710 is transmitting an annunciation for which the receive unit 730 is receptive, the broadcast unit 710 will be set into a state where it will not establish communications with the receive unit 730 in the step 1106. This can occur either by not creating the message socket in the step 1106 when connection is made with the receiver unit 730, or that the annunciator 1050 remains silent for a predetermined period, perhaps for a period of seconds.

15 If the broadcast unit 710 and the receiver unit 730 do mutually accept a multicasting relationship, the broadcast unit 710 creates the multicast out UDP socket 1080 in a step 1112 and the receiver unit 730 creates the multicast in UDP socket 1090 in the step 1114, and multicast audio transmission and reception is initiated in a step 1116. It should be noted that should the broadcast unit 710 already be multicasting audio to a receiver unit 730 prior to the
20 step 1112, the multicast out socket 1080 is not created, but that the address of this existing socket 1080 is communicated to the new cluster member.

Given that a cluster can comprise many members, the system of Figures 14A-B must be able to expand to include multiple members. Figure 15 is a schematic block diagram of the IP socket organization used with clusters comprising multiple members. The broadcast
25 unit 710 includes a broadcast annunciator 1050 indicating broadcast availability for new members. For each member in the cluster, the broadcast unit further comprises a message handler 1070 dedicated to the specific member, whose receive unit 730 in turn comprises a message handler 1060, generally in a one-to-one relationship. The broadcast unit comprises N messaging sockets 1070 for the N receive units of the cluster, while each member has only
30 a single socket 1060 connected to the broadcast unit. Thus, when a member wishes to send a message to the other members of the cluster, the message is sent via the receive unit message handler 1060 to the broadcast unit message handler 1070, and which is then multiply sent to the other receive unit message handlers 1060. It is also within the teachings of the present invention for each member of the cluster to have direct messaging capabilities with each

other member, assisted in the creation of the communications by the broadcast unit 710, which can share the socket addresses of each member of the cluster, such that each member can assure that it is making connections with other members of the cluster rather than units of non-members. The broadcast unit 710 also comprises a multicast out socket 1080 which
5 transfers audio to individual receiver sockets 1090 on each of the members of the cluster.

Members of the cluster may come and go, especially since members will frequently move physically outside of the transmission range of the broadcast unit 710. In order for the broadcast unit 710 to determine the current number of members of its cluster, it is within the teachings of the present invention for the broadcast unit 710 to use the messaging sockets
10 1060 and 1070 to "ping" the receive units 730 from time to time, or otherwise attempt to establish contact with each member of the cluster 700. Such communications attempts will generally be done at a predetermined rate, which will generally be more frequent than once every ten seconds. Information about the number of members of a cluster can be sent by the broadcast unit 710 to the other members of the cluster, so that the users can know how many
15 members there are. Such information is conveniently placed on a display on the unit (see, for example, Figures 18A-B).

Music Synchronization

It will be generally desirable that the synchronicity of the audio playback on the broadcast unit 710 and the receive units 730 be highly synchronized, preferably within 1
20 second (i.e. this provides a low level functionality of listening to music together), more preferably within 100 milliseconds (i.e. near-simultaneous sharing of music, but an observer would be able to hear – or see through DJ 200 visible cues – the non-synchronicity), and most preferably within 20 milliseconds of one another. In a simple embodiment of the present invention, all members of a cluster 700 must communicate directly with the broadcast unit
25 710, without any rebroadcast. In such cases, making playback on the two units 710 and 730 as similar as possible will tend to synchronize their audio production.

Figure 8A is a schematic block diagram of audio units 100 with self-broadcast so that audio output is highly synchronized. Two audio units 100 are depicted, including a broadcast unit 710 and a receive unit 730. The organization of audio unit 100 elements is chosen to
30 highlight the self-broadcast architecture. The audio media 1500, which can be compressed audio storage 310, stores the audio signals for broadcast. The output port 1502, which can comprise the inter-unit transmitter/receiver 110, transmits a broadcast audio signal, provided by the audio media 1500. The audio media comprise a variety of different storage protocols

and media, including mp3 files, .wav files, or .au files which are either compressed or uncompressed, monoaural or stereo, 8-bit, 16-bit or 24-bit, and stored on tapes, magnetic disks, or flash media. It should be understood that the spirit of the present invention is applicable to a wide variety of different audio formats, characteristics, and media, of which
5 the ones listed above are given only by way of example. This broadcast audio signal transmitted from the output port 1502 is received at the input port 1504, which can also comprise aspects of the inter-unit transmitter/receiver 110. The signal so received is then played to the associated user via the audio output 1508.

It should be noted that the audio output is normally connected to the audio media
10 1500 for audio playing when the unit 710 is not broadcasting to a receive unit 730. In such case, there is no need for the audio signals to go to the output port 1502 and thence to the input port 1504. Indeed, even when broadcasting, the audio signal within the broadcast unit 710 can go both directly to the audio output 1508 as well as to be broadcast from the output port 1502.

However, in order to assure the synchronicity of the audio output on the broadcast
15 unit 710 and the receive unit 730, the broadcast unit 710 can present all audio signal from the audio media 1500 for output on the output port 1502. The signal will be received not only on the receiver 730 input port 1504, but also on the input port 1504 of the broadcast unit 710. This can take place either through the physical reception of the broadcast audio signal on a
20 radio frequency receiver, or through local feedback loops within the audio unit 100 (e.g. through employment of IP loopback addresses).

In the receive unit 730, the audio signal received at the input port 1504 goes directly to the audio output 1508, and the other elements of the unit 100 depicted are not active. In the broadcast unit 710, however, if means are used to transfer audio signal between the output
25 port 1502 and the input port 1504 are utilized, and if such transfer means requires less time than that taken for transmitting signal from the output port 1502 of the broadcast unit 710 to the input port 1504 of the receive unit 730, then a delay means 1506 is introduced to provide a constant delay between the input port 1504 and the audio output 1508. This delay can comprise a digital buffer if the signal is digitally encoded, or an analog delay circuit if the
30 signal is analog. Generally, the delay introduced into the audio playback will be a predetermined amount based on the characteristics of the unit hardware and software.

Alternatively, in the case of a digital signal, the delay can be variably set according to the characteristics of the communications system. For example, if there are IP-based communications between the units, the units can "ping" one another in order to establish the

time needed for a "round-trip" communications between the systems. Alternatively, each receive unit 730 of a cluster 700 can transmit to the broadcast unit 710 a known latency of the unit based on its hardware and transmission characteristics. It should be noted that in order to handle different delays between multiple members of a cluster, a delay can be introduced into both the broadcast unit 710 and the receive unit 730, should a new member to the cluster have a very long latency in communications.

Note that the delay 1506 can serve a second purpose, which is to buffer the music should there be natural interruptions in the connections between the members of the cluster 700 (for example, should the receive units 730 move temporarily outside of the range of the broadcaster unit 710). In such case, should enough audio signal be buffered in the delay 1506, there would not be interruption of audio signal in the receive unit 730. Even in such cases, however, in order to accommodate the differences in time to play audio between units and within a unit, the delays in the broadcast unit 710 can be larger than those in the receive unit 730.

If the music compression and the bandwidth of the inter-unit communications are large enough, it can be that the broadcast unit 710 will broadcast less than half of the time. This will generally allow the receive unit 730 to rebroadcast the information from an internal memory store, allowing the effective range of the broadcast signal to potentially double. This can allow, through multiple rebroadcasts, for a very large range even if each individual unit 100 has a small range, and therefore for a potentially large number of users to listen to the same music.

In order to synchronize those that listen to the music through first, second and Nth rebroadcast, a scheme for multi-broadcast synchronization is presented in Figure 8B, a schematic flow diagram for synchronous audio playing with multiple rebroadcast. In such a case, the cluster 700 is considered to be all units 100 that synchronize their music, whether from an original broadcast or through multiple rebroadcasts. In a first step 780, a unit 100 receives a music broadcast along with two additional data. The first data is the current "N", or "hop" of the broadcast it receives, where "N" represents the number of rebroadcasts from the original broadcast unit 710. Thus, a unit 100 receiving music from the original broadcast unit 710 would have an "N" of "1" (i.e. 1 hop), while a unit 100 that received from that receiving unit 100 would have an "N" of "2" (2 hops), and so on. A second piece of information would be the "largest N" that was known to a unit 100. That is, a unit 100 is in contact generally with all units 100 with which it either receives or transmits music, and each send the "largest N" with which it has been in contact.

In a second step 782, the unit 100 determines the duration between signals in the broadcasts it is receiving. Then, two actions are taken. In a step 786, the unit 100 rebroadcasts the music it has received, marking the music with both its "N" and the largest "N" it knows of (either from the unit from which it received its broadcast or from a unit to which it has broadcast).

Also, in a step 784, the music that has been received is played after a time equal to the duration between signals and the "largest N" minus the unit's "N". This will allow for all units 100 to play the music simultaneously. Consider, for example the original broadcast unit 710. It's "N" is "0", and its "largest N" is the maximum number of rebroadcasts in the network. It will store music for a period of "largest N" (equals "largest N" minus "0") times the duration of a rebroadcast cycle, and then play it. For a unit 100 at the furthest rebroadcast, it's "N" and "largest N" will be equal to one another, so that it will store music for no time (i.e. "largest N" minus "N" = 0), but will play it immediately. This will allow all units 100 in the cluster to play music simultaneously. The limitation, however, is that there is memory in each unit 100 to store the music for a sufficient period of time. The units 100 on the system, however, can transfer the amount of storage that is available with the other information, and the number of rebroadcasts can be limited to the amount of memory available within the units 100 that comprise the cluster 700.

As the size of this multi-broadcast cluster 700 changes, the "largest N" can vary, and it will take generally on the order of "largest N" steps for the system to register "largest N". In such cases, there can be temporary gaps in the music on the order of the duration between signals, which will generally be on the order of tens of milliseconds, but which can be longer.

It should be noted that the synchronization of music does not need to accompany the transfer of an actual music signal. Figure 34A is a schematic block flow diagram of the synchronization of music playing from music files present on the units 100. In this embodiment, in a step 1900, the broadcast unit establishes the presence or absence of the music file comprising the music signals to be played on the receive unit. The music file can be referenced either with respect to the name of the file (e.g. "Ooops.mp3"), or a digital identifier that is associated with the music file.

If the music file is not present, then transfer of the music file from the broadcast unit to the receive units can automatically proceed through a file transfer mechanism such as peer-to-peer transfer in a step 1904. If the file was already present, or if the file has been transferred, or alternatively, if the file transfer has begun and enough of the file is present to

allow the simultaneous playing of music between the two units 100, transmission of synchronization signals between the two units 100 can commence in a step 1902.

These synchronization signals can comprise many different forms. For example, the synchronization signal can be the time stamp from the beginning of the music file to the current position of the music file being played on the broadcast unit. Alternatively, the broadcast unit can send the sample number that is currently being played on the broadcast unit 100. In order to allow receiving units to begin synchronous playing in the middle of a transmission from a broadcast unit, the synchronization signals will preferably include information about the song being played, such as the name of the file or the digital identifier associated with the file.

Transmission of this synchronization signal continues until the termination of the song, or until a manual termination (e.g. by actuating a Pause or Stop key) is caused (the frequency of transmission of the synchronization signal will be discussed below). At this point, the broadcast unit can send a termination, pause or other signal in a step 1906. Note that this method of synchronization can operate when the receiving unit establishes connection with the broadcast unit even in the middle of a song.

Figure 34B is a schematic layout of a synchronization signal record 1910 according to Figure 34A. The order and composition of the fields can vary according to the types of music files used, the means of establishing position, the use of digital jewelry, the desire for privacy, and more.

The position field 1912 (SAMPLE#) which contains an indicator of position in a music file – in this case the sample number within the file. The music file identifier field 1914 (SONGID) comprises a textual or numerical identifier of the song being played. The third field is the sample rate field 1916 (SAMPLERATE), and is primarily relevant if the position field 1912 is given in samples, which allows a conversion into time. Given that the same audio entertainment can be recorded or saved at different sample rates, this allows the conversion from a potentially relative position key (samples) to one independent of sample rate (time). The jewelry signal field 1918 (JEWEL SIGNAL) is used to encode a digital jewelry 200 control signal for controlling the output of the digital jewelry 200, should the receiver unit be associated with jewelry 200. The order and composition of the fields can vary according to the types of music files used, the means of establishing position, the use of digital jewelry, the desire for privacy, and more.

The frequency with which the record 1910 is broadcast can vary. The time of reception of the record 1910 sets a current time within the song that can adjust the position of

the music playing on the receiver unit. It is possible for the record to be broadcast only once, at the beginning of the song, to establish synchronization. This, however, will not allow others to join in the middle of the music file. Furthermore, if the record 1910 is received or processed at different times for the single record, the music can be poorly synchronized.

- 5 With multiple synchronization signals, the timing can be adjusted to account for the most advanced reception of the signal – that is, the music playing will be adjusted forward for the most advanced signal, but not be adjusted back for a more laggard signal.

If the record further contains a jewelry signal field 1918, the frequency with which the record 1910 should be sent should be comparable or faster than the rate with which these
10 signals change, and should be preferably at least 6 times a second, and even more preferably at least 12 times a second. If less frequent record 1910 transmission is desired, then multiple jewel signal fields 1918 can be included in a single record 1910.

It should be noted that given units 100 of different design or manufacture, there can be different intrinsic delays between reception of music and/or synchronization signals and
15 the playing of the music. Such delays can result from different speeds of MP3 decompression, different sizes of delay buffers (such as delay 1506), different speeds of handling wireless transmission, differing modes of handling music (e.g. directly from audio media 1500 to audio output 1508 on the broadcast unit, but requiring transmission through an output port 1502 and input port 1504 for the receiver unit), and more. In such cases, it is
20 preferable for receiver units to further comprise a manual delay switch that can adjust the amount of delay on the receiver unit. This switch will generally have two settings: to increase the delay and to decrease the delay, and can conveniently be structured as two independent switches, a rocker switch, a dial switch or equivalent. It is useful for the increments of delay determined by the switch be adjustable so as to allow users to sense the
25 music from the broadcast unit and the receiver unit as being synchronous, and it is preferable for the increments of delay to be less than 50 milliseconds, and even more preferable for the increments of delay to be less than 20 milliseconds, and most preferable for the units of delay to be less than 5 milliseconds.

Creation and Maintenance of Clusters

- 30 Search units 750 can be playing music themselves, or can be scanning for broadcast units 710. Indeed, search units 750 can be members of another cluster 700, either as broadcast unit 710 or receive unit 730. To detect a different cluster 700 in which it might desire membership, the search unit 750 can either play the music of the broadcast unit 710 to

the search unit 750 user, or it can scan for personal characteristics of the broadcast unit 710 user that are transmitted in the ID block 808. For example, a user can establish personal characteristic search criteria, comprising such criteria as age, favorite recording artists, and interest in skateboarding, and respond when someone who satisfies these criteria approaches.

5 Alternatively, the search unit 750 user can also identify a person whose cluster he wishes to join through visual contact (e.g. through perceiving the output of the person's light transducer 240).

Before a search unit 750 user can establish contact, it is preferable for a broadcast unit 710 user, or a receive unit 730 user, to provide permissions for others to join the cluster. For
10 example, each unit 100 will generally be able to changeably set whether no one can join with their unit 100, whether anyone can join with their unit 100, or whether permission is manually granted for each user who wishes to join with their unit into a cluster. For a cluster 700, membership in the cluster can be provided either if any one member of the cluster 700 permits a search unit 750 user to join, or it can be set that all members of a cluster 700 need
15 to permit other users to join, or through a variety of voting schemes. The permissions desired by each member will generally be sent between units 100 in a cluster as part of the ID block 808 or other inter-unit communications. Furthermore, these permissions can be used to establish the degree to which others can eavesdrop on a unit 100 transmission. This can be enforced either through the use of cryptography, which can only provide decryption keys as
20 part of becoming a cluster 700 member, through provision of a private IP socket address or password, through standards agreed by manufacturers of unit 100 hardware and software, or by unit 100 users limiting the information that is sent through the ID block 808 through software control.

The search unit 750 user can then establish membership in the group in a variety of
25 ways. For example, if the search unit 750 is scanning music or personal characteristics of the unit 100 user, it can alert the search unit 750 user about the presence of the unit 100. The search unit 750 user can then interact with the search unit 750 interface to send the unit 100 user a message requesting membership in the cluster 700, which can be granted or not. This type of request to join a cluster 700 does not require visual contact, and can be done even if
30 the search unit 750 and cluster are separated by walls, floors, or ceilings.

Another method of establishing contact between a search unit 750 user and a cluster 700 member is for the search unit 750 user to make visual contact with the cluster 700 member. In such case that physical contact or physical proximity is easily made between the unit 100 of the cluster member and the search unit 750, digital exchange can be easily made

either through direct unit 100 contact through electrical conductors, or through directional signals through infra-red LEDs, for example. For example, the search unit 750 user can point his unit 100 at the cluster 700 member unit, and then if the cluster member wishes the search unit 750 user to join the cluster, could point his unit 100 at the search unit 100, and with both pressing buttons, effect the transfer of IDs, cryptography keys, IP socket addresses or other information that allows the search unit 750 user to join the cluster 700.

Alternatively, the broadcast DJ 720 (or the receive DJ 740) can present digital signals through the light transducer. For example, most DJ 720 light transduction will be modulated at frequencies of 1-10 Hz, with human vision not being able to distinguish modulation at 50 Hz or faster. This means that digital signals can be displayed through the light transducer 240 at much higher frequencies (kHz) that will not be perceived by the human eye, even while lower frequency signals are being displayed for human appreciation. Thus, the broadcast DJ 720 can receive a signal from the broadcast unit 710 DJ transmitter 120 containing information needed for a search unit 750 to connect to the broadcast unit's cluster 700. This information will be expressed by the light transducer 240 of the broadcast DJ 720 in digital format. The search unit 750 can have an optical sensor, preferably with significant directionality, that will detect the signal from the light transducer 240, so that the search unit 750 is pointed in the direction of the broadcast DJ 720, and the identifier information required for search unit 750 to become a member of cluster 700. This optical sensor serves as the DJ directional identifier 122 of Figure 1. At this point, if desired, the broadcast unit 710 user can determine if they want the search unit 750 user to join the cluster 700.

A summary of means to effect joining of a cluster is provided in Figures 13A through E, which display means for a search unit 750 to exchange information prior to joining a cluster 700 via a broadcast unit 710. It is also within the teachings of the present invention for the search unit 750 to institute communications with a receive unit 730 for the purposes of joining a cluster in a similar fashion, particularly since it may be difficult for a person outside of the cluster 700 to determine which of the cluster 700 members is the broadcast unit 710, and which is a receiver unit 730.

It should be noted in the Figures 13A-G that limited range and directionality are preferred. That is, there can be a number of broadcast units 710 within an area, and being able to select that one broadcast unit 710 whose cluster one wishes to join requires some means to allow the search unit 750 user to select a single broadcast unit 710 among many. This functionality is generally provided either by making a very directional communication between the two devices, or by depending on the physical proximity of the search unit 750

and the desired broadcast unit 710 (i.e. in a greatly restricted range, there will be fewer competing broadcast units 710). In the following description, the “broadcaster” denotes the user using the broadcast unit 710, and the “searcher” denotes the user using the search unit 750.

5 In the Figures 13A-G, the selection of the cluster by the searcher occurs in three ways, that will referred to as “search transmission mode”, “broadcast transmission mode”, and “mutual transmission mode”, according to the entity that is conveying information. In search transmission mode, the searcher sends an ID via the search unit 750 to the broadcast unit 710. This ID can comprise a unique identifier, or specific means of communication (e.g. an IP
10 address and port for IP-based communication). With this ID, the broadcast unit can either request the searcher to join, or can be receptive to the searcher when the searcher makes an undifferentiated request to join local units within its wireless range. In broadcast transmission mode, the broadcaster sends an ID via the broadcast unit 710 to the search unit 750. With this ID, the searcher unit can then make an attempt to connect with the broadcast
15 unit 710 (e.g. if the ID is an IP address and port), or the search unit can respond positively to a broadcast from the broadcast unit 710 (e.g. from a broadcast annunciator 1050), wherein the ID is passed and checked between the units early in the communications process. Mutual transmission mode comprises a combination of broadcast transmission mode and search transmission mode, in that information and communication is two way between the
20 broadcaster and the searcher.

Figure 13A is a schematic cross-section through a search unit 750 and a broadcast unit 710 in which communications are provided via visible or infrared LED emission in search transmission mode. On the right of the figure, a LED 1044 with an associated lens 1046 (the two of which can be integrated) transmits a directional signal from the unit case 1000. This
25 light can optionally pass thorough a window 1048 that is transparent to the light. On the left of the figure, a lens element 1040 collects light through a broad solid angle and directs it onto a light sensing element 1042, which is conveniently a light-sensing diode or resistor. The directionality of the communication is conferred by the transmitting lens 1046 and the collecting lens 1040.

30 Alternatively, the LED 1044 can be replaced by a visible laser. Figure 13B is a schematic cross-section through a search unit 750 and a broadcast unit 710 in which communications are provided via a visible or infrared laser in search transmission mode. The search unit 750 comprises a diode laser 1041 that is conditioned by a lens 1043 to form a beam that is sensed by the light sensing element 1042 on the broadcast unit 710. Because a

collimated laser beam can be difficult to aim with precision at a photosensing element carried by a person, the optics can comprise a two focus lens 1043 that has a portion that produces a collimated beam 1045, and a second portion that produces a diverging beam 1047. The collimated beam is used by the user of the search unit 750 as a guide beam to direct the pointing of the unit 750, while the divergent beam provides a spread of beam so that the human pointing accuracy can be relatively low. The means for creating the two focus lens 1043 can include the use of a lens with two different patterns of curvature across its surface, or the use of an initial diverging lens whose output intersects a converging lens across only a part of its diameter, where the light that encounters the second lens is collimated, and the light that does not encounter the second lens remains diverging. It is also within the teachings of the present invention for the lens to be slowly diverging without a collimating portion, such that the user does not get visible feedback of their pointing accuracy. In such case, the laser can emit infrared rather than visible wavelengths.

Figure 13C is a schematic cross-section through a search unit 750 and a broadcast unit 710 in which communications are provided via visible or infrared emission from a digital jewelry element 200 in broadcast transmission mode. The digital jewelry 200 is carried by the broadcaster on a chain 1033, with the digital jewelry 200 visible. The digital jewelry is emitting through a light transducer 1031 a high frequency signal multiplexed within the visible low frequency signal. The search unit 750 is pointed in the direction of the digital jewelry 200, and receives a signal through the light-sensing element 1042. This manner of communication is convenient because the searcher knows, via the presence of the visible signal on the digital jewelry 200, that the broadcaster is receptive to cluster formation.

Figure 13D is a schematic cross-section through a search unit 750 and a broadcast unit 710 in which communications are provided via contact in mutual transmission mode. In this case, the broadcast unit 710 and the search unit 750 both comprise a contact transmission terminus 1030, and electronic means by which contact transmission is performed. This means can operate either inductively (via an alternative current circuit), through direct electrical contact with alternating or direct current means, or other such means that involves a direct physical contact (indicated by the movement of the search unit 750 to the position of the unit depicted in dotted lines). The search unit 750 or the broadcast unit 710 can, via automatic sensing of the contact or manual control, initiate communications transfer. Given the mutuality of contact as well as the physical equivalence of the two units 710 and 750, information transfer is possible in both directions. It should be noted that in the case of direct

current connection, the termini 1030 will comprise two contact points, both of which must make electrical contact in order for communications to occur.

Figure 13E is a schematic cross-section through a search unit 750 and a broadcast unit 710 in which communications are provided via sonic transmissions in broadcast transmission mode. The broadcaster (or receivers) will be listening to the audio information generally through headphones 1020 or earphones, all of which comprise speakers 1022 that, to one extent or another, leak sonic energy. The use of audio output devices as depicted in Figure 10 and Figures 11A and 11B that admit external sound, will also increase the amount of sound energy lost. This sound energy can be detected by the searcher via a directional speaker comprising a sound collector 1024 and a microphone 1026. This system requires that the sound output of the broadcast unit 710 and the receiver unit 750 also output an ID encoded in the sound. Such sound can be conveniently output at inaudible frequencies, such as 3000-5000 Hz, which carry sufficient bandwidth to encode short messages or identifiers (e.g. an IP address and port number can be carried in 5 bytes). Sound energy, especially at higher frequencies, can be quite directional, depending on the shape of the collector 1024 and the structure of the microphone 1024, allowing good directional selection by the searcher.

Figure 13F is a schematic cross-section through a search unit 750 and a broadcast unit 710 in which communications are provided via radio frequency transmissions in broadcast transmission mode. The radio frequency transmissions are not strongly directional (and for the purposes of the broadcast of audio information, are designed to be as directionless as possible). In order to distinguish a desired cluster 700 to join and an undesired cluster 700, a number of strategies can be employed. For example, the strengths of the various signals can be measured and the strongest chosen for connection. Alternatively, if there are multiple broadcast connections available, the search unit 750 can sequentially attempt a connection with each broadcast unit 710. When the attempt is made, the broadcast unit 710 can, prior to alerting the broadcaster of the attempted joining by a new member, cause the digital jewelry 200 associated with the broadcast unit 710 to visibly flash a characteristic signal. The searcher can then verify by pressing the appropriate button on the search unit 750 his desire to join the cluster 700 of the broadcast digital jewelry 200 that had just flashed. If the searcher decided not to join that cluster 700, the search unit 750 could search for yet another unit broadcast unit 750 within range, and attempt to join.

At any time, the members of a cluster 700 can share personal characteristics (nickname, real name, address, contact information, face or tattoo images, favorite recording artists, etc.) through selection of choices of the unit 100 interface, with all such characteristics

or a subset thereof to be stored on the units 100. In order to assist cluster 700 members in determining whether or not to accept a person into their cluster 700, a search unit 750 member can display either the total number of people with whom he has shared personal characteristics, or he can alternatively allow the cluster members to probe his store of persons with whom personal characteristics have been stored to see whether a particular trusted person or group of common acquaintances are present therein. It is also within the spirit of the present invention for individuals to rate other individual members of their cluster, and such ratings can be collated and passed from person to person or cluster to cluster, and can be used for a cluster 700 to determine whether a search unit 750 person should be added to the cluster 700.

Figure 17 is a matrix of broadcaster and searcher preferences and characteristics, illustrating the matching of broadcaster and searcher in admitting a searcher to a cluster. A broadcaster preference table 1160 includes those characteristics that the broadcaster wishes to see in a new member of a cluster. These characteristics can include gender, age, musical "likes" and "dislikes", the school attended, and more. The searcher similarly has a preference table 1166. The searcher preference table 1166 and broadcaster preference table 1160 are not different in form, as the searcher will at another time function as the broadcaster for another group, and his preference table 1166 will then serve as the broadcaster preference table.

The broadcaster preference table 1160 can be automatically matched with a searcher characteristics table 1162. This table 1162 comprises characteristics of the searcher, wherein there will be characteristics that overlap in type (e.g. age, gender, etc.) which can then be compared with the parameters in the broadcaster preference table. This matching occurs during the period when the searcher is interrogating the cluster with interest in joining. Similarly, there is a broadcaster characteristics table 1164 indicating the characteristics of the broadcaster, which can be matched against the searcher preferences table 1166.

The algorithm used in approving or disapproving of an accord between a preference table and a characteristics table can be varied and set by the user – whether by the broadcaster to accept new members into a cluster, or by a searcher to join a new cluster. For example, the user could require that the gender be an exact match, the age within a year, and the musical preferences might not matter. The user can additionally specify that an accord is acceptable if any one parameter matches, specify that an accord be unacceptable if any one parameter does not match, specify an accord be acceptable based on the overlap of a majority of the individual matches, or other such specification.

It should be noted that the broadcaster preferences table 1160 and the broadcaster characteristics table 1164 (and likewise with the searcher tables 1162 and 1166) can be a single table, according to the notion that a person will prefer people who are like themselves. Each user could then express the acceptable range of characteristics of people with which to
 5 join as a difference from their own values. For example, the parameter “same” could mean that the person needs to match closely, whereas “similar” could indicate a range (e.g. within a year) and “different” could mean anyone. In this way, there would not be the burden on the user to define the preference table 1160 or 1166 in a very detailed manner.

In the case of a cluster, the transfer of information between the searcher and the
 10 cluster can, as mentioned above, involve not only the broadcaster, but also other members of the cluster (especially since the searcher may not know the identity of a cluster’s broadcaster from external observation). The cluster can also make communal decisions about accepting a new member. That is, if there are 4 members of a cluster, and a searcher indicates an interest in joining the cluster, there can be voting among the members of a cluster regarding the
 15 acceptance of the new member. The procedure of voting will normally be done by messaging among the members, which can be assisted by structured information transfer as will be described below.

A number of such voting schemes are described in Figure 19, a table of voting schemes for the acceptance of new members into a cluster. The first column is the name of
 20 the rule, and the second column describes the algorithm for evaluation according to the rule. In the “BROADCASTER” rule, the broadcaster decides whether or not the new member will be accepted. The new member is accepted when the broadcaster indicates “yes” and is otherwise rejected.

In the “Majority” rule, the members are polled, and whenever a majority of the
 25 members vote either acceptance or rejection, the new member is accordingly accepted or rejected. It should be noted that this rule (as well as the rules to follow) depends on the broadcaster or other member of the cluster having knowledge of the number of members in the cluster, which will generally be the case (e.g. in an IP socket based system, the broadcaster can simply count the number of socket connections). Thus, if the number of
 30 members in a cluster is given as N_{mem} , as soon as $(N_{\text{mem}}/2) + 1$ members have indicated the same result, that result is then communicated to the broadcaster, the members and the prospective new member. If the number of members is even, and there is a split vote, the result goes according to the broadcaster’s vote.

According to the “Unanimous” rule, a new member is accepted only on unanimous decision of the members. Thus, the prospective new member is rejected as soon as the first “no” vote is received, and is accepted only when the votes of all members of the cluster are received, and all of the votes are positive.

5 The “Timed Majority” rule is similar to that of the “Majority” rule, except that a timer is started when the vote is announced, the timer being of a predetermined duration, and in a preferred embodiment, is indicated as a count down timer on the unit 100 of each member of the cluster 700. The vote is completed when $(N_{\text{mem}}/2)+1$ members vote with the same indication (“yes” or “no”) if the timer has not completed its predetermined duration. If all of
10 the members have voted, and the vote is a tie, the result goes in accordance with that of the broadcaster. If the timer has expired, and the vote has not been decided, the number of members that have voted is considered a quorum of number Q . If $(Q/2)+1$ members have voted in some fashion, that is the result of the vote. Otherwise, in the case of a tie, the result goes according to the vote of the broadcaster. If the broadcaster has not voted, the vote goes
15 according to the first vote received.

 The “Synchronized Majority” rule is similar to the Timed Majority rule, but instead of initiating the vote, and then waiting a predetermined period for members to vote, the vote is announced, and then there is a predetermined countdown period to the beginning of voting. The voting itself is very limited in time, generally for less than 10 seconds, and preferably for
20 less than 3 seconds. Counting votes is performed only for the quorum of members that vote, and is performed according to the rules for the Timed Majority.

 There are many different voting schemes consistent with creating, growing and maintaining clusters within the spirit of the present invention. For instance, in cases where there are close votes, the voting can be reopened for individuals to change their vote. In
25 cases, members can request a new round of voting. Furthermore, the voting can be closed ballot, in which the votes of individuals are not known to the other members, or open voting, in which the identity of each member’s vote is publicly displayed on each unit 100.

 In addition, the voting can be supported and enhanced by information made available to each member through displays on the units 100. Figure 18A is a screenshot of an LCD
30 display 1170 of a unit 100, taken during normal operation. The display 1170 is comprised of two different areas, an audio area 1172 and a broadcaster area 1174. The audio area 1172 includes information about the status of the audio output and the unit 100 operation, which can include battery status, the name of the performer, the title of the piece of music, the time the audio has been playing, the track number and more. The broadcaster area 1174 comprises

information about the status of the cluster 700. In the example given, the broadcaster area includes the number “5”, which represents the number of people current in the cluster, the text “DJ”, which indicates that the unit 100 on which the display 1170 is shown is currently the broadcaster of the cluster 700, and the text “OPEN”, which indicates that the cluster is open for new members to join (the text “CLOSED” would indicate that no new members are being solicited or allowed).

Figure 18B is a screenshot of an LCD display 1170 of a unit 100, taken during voting for a new member. The audio area 1172 is replaced by a new member characteristics area 1176, in which characteristics of the prospective new member are displayed. Such characteristics can include the name (or nickname) of the prospective new member, their age, and their likes (hearts) and dislikes (bolts). In the broadcaster area 1174, the digit “3” indicates that there are three current members of the cluster 700, and an ear icon indicates that the current unit 100 is being used to receive from the broadcaster rather than being a broadcaster, and the name [ALI] indicates the name of the current broadcaster. The text “VOTE-MAJ” indicates that the current vote is being done according to the Majority rule. The broadcaster area 1174 and the new member characteristics areas 1176 provide the information needed by the existing member to make a decision about whether to allow the prospective new member to join.

The displays 1170 of Figures 18A-B are indicative only of the types of information that can be placed on a display 1170, but it should be appreciated that there are many pieces of information that can be placed onto the displays 1170 and that the format of the display can be very widely varied. Furthermore, there need not be distinct audio areas 1172 and broadcaster areas 1174, but the information can be mixed together. Alternatively, especially with very small displays 1170, the display 1170 can be made to cycle between different types of information.

It is also within the spirit of the present invention for individuals to rate other individual members of their cluster, and such ratings can be collated and passed from person to person or cluster to cluster, and can be used for a cluster 700 to determine whether a search unit 750 person should be added to the cluster 700. Figure 27 is a schematic block flow diagram of using a prospective new member’s previous associations to determine whether the person should be added to an existing cluster.

In a step 1400, from a search unit 750, the prospective new member places an external communication request with an operational broadcast annunciator 1050 by a broadcast unit 710. In a step 1402, a temporary message connection is established through which

information can be passed mutually between the search unit 750 and the broadcast unit 710. The broadcast unit 710 requests personal and cluster ID's from the search unit 750. The personal ID is a unique identifier that can be optionally provided to every audio unit 100, and which can further be optionally hard-encoded into the hardware of the unit 100. The cluster
5 IDs represent the personal ID's of other units 100 with which the search unit 750 has been previously associated in a cluster. In a step 1406, the broadcast unit 710 matches the incoming personal IDs and cluster IDs with personal ID's and cluster IDs that are stored in the memory of the broadcast unit 710. If there exist a sufficient number of matches, which can be computed as a minimum number or as a minimum fraction of the IDs stored in the
10 broadcast unit 710, the new member of the search unit 750 can be accepted into the cluster. In a step 1412, the search unit 750 can then store the ID of the broadcast unit 710 and the other members of the existing cluster 700 into his cluster IDs, and the broadcast unit 750 and the other receive units 730 of the cluster can then store the personal ID of the search unit 750 into their cluster IDs. If there does not exist a sufficient number or quality of matches, the
15 broadcast unit 710 will reject the prospective new member, optionally send a message of rejection, and then close the socket connection (or other connection that had been created) between the broadcast unit 710 and the search unit 750. No new IDs are stored on either unit 710 or 750.

It is also within the spirit of the present invention for other information associated
20 with the personal and cluster IDs to be shared and used in the algorithm for determining whether to accept or reject a prospective new member into a cluster 700. This information can include rating information, the duration of association with another cluster 700 (i.e. the longer the association, the more suitable the social connection of that person with the cluster 700 would have been), the size of the cluster 700 when the searcher was a member of a
25 particular cluster 700, the popularity of a cluster 700 (measured by the number of cluster IDs carried by the broadcast unit 710), and more. The matching program, likewise, would weight the existence of a match by some of these quality factors in order to determine the suitability of the searcher to join the cluster.

While the comparisons can be made between a search unit 750 personal and cluster
30 IDs and those from the broadcast unit 710, representing the personal experience of the owners of the respective units, it is also possible that the reputation or desirability of individuals with a given personal ID can be posted to or retrieved from trusted people. For example, two friends can swap the information of which IDs are to be trusted or not between two units 100, or alternatively, can be posted onto or retrieved from the Internet. For

example, after a bad personal experience with a unit 100 with a personal ID of 524329102, a person could post that ID on the Internet to share with friends, so that the friends could avoid allowing that person to join, or avoid joining a cluster with that person.

It should be noted that publishing a list of personal IDs allows people to establish the
5 breadth of their contacts. By posting their contacts on web sites, people can demonstrate their activity and popularity. This also encourages people to join clusters, in order to expand the number of people with whom they have been associated. Furthermore, the personal ID serves as a “handle” by which people can further communicate with one another. For example, on the Internet, a person can divulge a limited amount of information (e.g. an email
10 address) that would allow other people with whom they have been in a cluster together to contact them.

It should be noted that the formation and maintenance of a cluster 700 requires the initial and continued physical proximity of the broadcast unit 710 and the receive unit 730. In order to help maintain such physical proximity conducive to cluster maintenance, feedback
15 mechanisms can be used to alert the users to help them maintain the required physical proximity, as will be discussed below.

Figure 28 is a block flow diagram indicating the steps used to maintain physical proximity between the broadcast unit 710 and the receive unit 730 via feedback to the receive unit user. In a step 1530, the wireless connection between the broadcast unit 710 and the
20 receive unit 730 is established. In a step 1532, the connection between the two units 710 and 730 is tested. There are a number of different means by which this testing can take place. For example, in IP-based communications, the receive unit 730 can from time to time – though generally less than every 10 seconds, and even more preferably less than every 1 second – use the “ping” function to test the presence and speed of connection with the
25 broadcast unit 710. Alternatively, the receive unit 730 will be receiving audio signals wirelessly almost continuously from the broadcast unit 710, and a callback alert function can be instituted such that loss of this signal determined at a predetermined repeat time – which is conveniently less than 5 seconds, and even more preferably less than every 1 second – and which is then reported to the system.

30 While the methods above determine the absolute loss of a signal, they do not anticipate loss of signal. A method that does anticipate signal issues prior to loss is the measurement of signal strength. This can be done directly in the signal reception hardware by measuring the wireless signal induced current or voltage.

In a step 1534, the results of the connection testing performed in the step 1532 is analyzed in order to determine whether the signal is adequate. It should be noted that a temporary loss of signal, lasting even seconds, may or may not be of importance. For example, the broadcast unit 710 user and receive unit 730 users could walk on opposite sides
5 of a metallic structure, enter a building at different times, change their body posture such that the antennae are not optimally situated with respect to one another, etc. Thus, an algorithm is generally used to time average the results of the step 1532, with the results conveniently time averaged over a matter of seconds.

Whatever the results of the signal test of the step 1534, the step 1532 is continuously
10 repeated as long as the connection between the broadcast unit 710 and the receive unit 730 is present. If the signal is deemed inadequate, however, feedback to that effect is provided to the receive unit 730 user in a step 1536. The user feedback can occur through a variety of mechanisms, including visual (flashing lights) and tactile (vibration) transducers, emanating either from the audio unit 100 or the digital jewelry 200. For example, the receiver unit 730
15 can send a signal to the associated digital jewelry 200 to effect a special sequence of light transducer output.

It is most convenient, however, for the audio output of the receiver unit 730 as heard by the user to be interrupted or overlain with an audio signal to alert the user to the imminent or possible loss of audio signal. This audio signal can include clicks, beeps, animal sounds,
20 closed doors, or other predetermined or user selected signals heard over silence or the pre-existing signal, with the signal possibly being somewhat reduced in volume such that the combination of the pre-existing signal and the feedback signal is not unpleasantly loud.

It should be noted that the flow diagram of Figure 28 refers specifically to alerting the receive unit 730 user of potential communications issues. Such alerting can also be usefully
25 transferred to or used by the broadcast unit 710. For example, with knowledge of the communications issues, the broadcast unit 710 user can move more slowly, make sure that the unit is not heavily shielding, that any changes in posture that could relate to the problems are reversed, etc. The broadcast unit 710 can perform communications tests (as in the step 1532) or analyze the tests to determine if the communications are adequate (as in the step
30 1534) – particularly through use of the messaging TCP channels. Given that there can be multiple receive units 730 connected to a single broadcast unit 710, it is generally preferable for the tests to be performed on the receive units 730, and problems to be communicated to the broadcast unit 710 – provided, however, that communications still exist for such communication.

In order to overcome this deficiency, it is possible for the receiver unit 730 to communicate potential problems in communications to the broadcast unit 710 at an early indication. The broadcast unit 710 then starts a timer of predetermined length. If the broadcast unit 710 does not receive a "release" from the receive unit 730 before the timer has
5 completed its countdown, it can then assume that communications with the receive unit 730 have been terminated, and it can then send feedback to the broadcast unit 710 user.

It is also within the teachings of the present invention for both the broadcast unit 710 and the receive unit 730 to independently monitor the connections with each other, and alert their respective users of communications problems.

10 It should be noted that the use of audio alerts can be used more generally within the user interface of the audio units 100. Thus, audio alerts can be conveniently used to inform the user of the joining of new members to the cluster 700, the initiation of communications with search units 750 outside of the group, the leaving of the group by existing cluster 700 members, the request by a receive unit 730 to become the broadcast unit 710, the transfer of
15 cluster control from a broadcast unit 710 to a receive unit 730, and more. These alerts can be either predetermined by the hardware (e.g. stored on ROM), or can be specified by the user. Furthermore, it can be convenient for the broadcast unit 710 to temporarily transfer to new members of the cluster custom alerts, so that the alerts are part of the experience that the broadcast unit 710 user shares with the other members of the cluster. Such alerts would be
20 active only as long as the receive units were members of the cluster 700, and then would revert back to the alerts present before becoming cluster members.

Cluster Hierarchy

A receive unit 730 can also be the broadcast unit 710 of a separate cluster 700 from the cluster 700 of which it is a member. This receive unit is called a broadcasting receiver
25 770. In such case, it is convenient for the receive units 730 that are associated with the broadcasting receiver 770 to become associated with the cluster 700 of which the broadcasting receiver 770 is a member. This can conveniently be accomplished in two different ways. In a first manner, the receive units 730 that are associated with the broadcasting receiver 770 can become directly associated with the broadcast unit 710, so that
30 they are members only of the cluster 700, and are no longer associated with the broadcasting receiver 770. In a second manner, the receive units 730 associated with the broadcasting receiver 770 can remain primarily associated with the broadcasting receiver 770, as shown in Figures 9A and 9B, which are schematic block diagrams of hierarchically-related clusters. In

Figure 9A, the receive units 730 that are members of a sub-cluster 701 of which the broadcast unit is a broadcast receive unit 770, can receive music directly from the broadcast receive unit 710, while retaining their identification with the broadcasting receiver 770, such that if the broadcasting receiver 770 removes itself or is removed from the cluster 700, these receive units 730 similarly are removed from the cluster 700. In order to provide this form of hierarchical control, the sub-cluster 701 receive units 730 can obtain an identifier, which can be an IP socket address, from the broadcast receive unit 770, indicating the desired link to the broadcast unit 710. The sub-cluster receive units 730, however, maintain direct communications with the broadcast receive unit 770, such that on directive from the unit 770, they break their communications with the unit 710, and reestablish normal inter-unit audio signal communications with the broadcast receive unit 770. In an embodiment using IP addressing and communications, this can involve the maintenance of TCP messaging communications between the sub-cluster 701 receive units 730 with the broadcast receive unit 770, during the time that the sub-cluster 701 is associated with the cluster 700.

In Figure 9B, the receive units 730 of the sub-cluster 701 receive music directly from the broadcasting receiver 770, which itself receives the music from the broadcast unit 710. In such case, as the broadcasting receiver 770 is removed from the cluster 700, the receive units 730 of the sub-cluster 701 would also not be able to hear music from the cluster 700.

It would be apparent that such an arrangement can be hierarchically arranged, such that the receive unit 730 of the sub-cluster 701 can itself be the broadcast receiver 770 of another sub-cluster 701, and so forth. The advantage of this arrangement is that people that are associated with one another, forming a cluster 700, can move as a group from cluster to cluster, maintaining a separate identity.

It should be also noted that the configuration of communications between members of a hierarchical cluster can be variously arranged, not only as shown in Figures 9A and 9B. For example, every member of the cluster 700 can have a direct link between every other member of the cluster 700, such that no re-broadcast of messages needs to take place. Furthermore, given that there are different inter-unit communications (for example, messaging versus audio broadcast), it is within the teachings of the present invention that the configuration for the different modes of communication can be different – for example direct communications between the broadcast unit 710 for audio broadcast, but peer-to-peer communications between individual units for messaging.

Maintaining Private Communications

In order to restrict membership in a cluster 700, either the information transfer must be restricted, such as by keeping private the socket IP addresses or passwords or other information that is required for a member to receive the signal, or the signal can be transmitted openly in encrypted form, such that only those members having been provided with the encryption key can properly decode the signal so sent. Both of these mechanisms are taught within the present invention, and are described at various points within this specification.

Figure 32A is a schematic block diagram of maintaining privacy in open transmission communications. In this case, the transmission is freely available to search units 750 in a step 1830, such as would occur with a digital RF broadcast, or through a multicast with open a fixed, public socket IP address available in certain transmission protocols. In this case, the broadcast audio signal or information signal is made in encrypted form, and membership in the cluster is granted through transfer of a decoding key in a step 1832.

Figure 32B is a schematic block diagram of maintaining privacy in closed transmission communication. In a step 1834, the broadcast unit 710 makes a closed transmission broadcast, such as through a socket IP address, that is not publicly available. In a step 1836, the broadcast unit 710 provides the private address to the search unit 750, which can now hear the closed transmission from the step 1834, which is not encrypted. Alternatively, or in addition to the provision of the private address in the step 1836, the establishment of the connection through the private, closed transmission is effected via a password provided in a step 1838. This password can, for example, be used in the step 1110 (e.g. see Figure 14B) to determine whether the broadcast unit 710 accepts the search unit 750 for audio multicasting.

In this section, the encryption of the musical signal and/or associated information about personal characteristics of members of the cluster 700 is described. The custom compressor 330 of the unit 100 can perform the encryption. In such a case, before joining a cluster, the search unit 750 can only receive some limited information, such as characteristics of the music being heard or some limited characteristics of the users in the cluster 700. If the search unit 750 user requests permission to join the cluster 700 and it is granted, the broadcast unit 710 can then provide a decryption key to the search unit 750 that can be used to decrypt the music or provide a private IP address for multicasting, as well as supply additional information about the current members of cluster 700.

It should be noted that in certain cases, it can be useful to have multiple forms of privacy protection. For example, a broadcast unit 710 can provide a search unit 750 access to audio signals and information for the cluster 700, but can reserve certain information based on encryption to only some members of the cluster 700. For example, if a group of friends
5 comprise a cluster 700, and accept some new members into the cluster 700, access to more private information about the friends, or communications between friends, can be restricted on the basis of shared decryption keys.

Figure 33 is a schematic block diagram of a hierarchical cluster, as in Figure 9A, in which communications between different units is cryptographically or otherwise restricted to
10 a subset of the cluster members. Thus, there are three types of communication that are used in the communication: channel A, which takes place between the members of the original cluster; channel B, which takes place between the members of the original cluster (mediated through the broadcast unit 710) and members of the sub-cluster 701; and channel C, which takes place between the members of the sub-cluster 701. Thus, a communications originating
15 from the broadcast unit 710 can be directed either through channel A or channel B, and likewise, a communications originating from the broadcast receive unit 770 can be directed either at members only of the sub-cluster 701 through channel C, or to all members of the cluster 700 through both channels C and B, which is then communicated through channel A.

A number of means can be used to maintain such independent channels. For example,
20 separate socket communications can be established, and the originators of the communications can determine that information which is carried on each separate channel. For example, given an open transmission scheme such as digital RF signal, the information can be encoded with separate keys for the different channels of communication – thus, the cryptographic encoding determines each channel. A given unit 100 can respond to more than
25 one encoding. Indeed, a channel identifier can be sent with each piece of information indicating the ID of the decoding key. If a unit 100 does not have the appropriate decoding key, then it is not privy to that channel communications.

Alternatively, if the communications is IP socket based, then each channel is determined by IP socket addresses. Furthermore, access to those addresses can be, for
30 example, password controlled. Also, the socket communications can be broadcast so that any unit 100 can receive such broadcast, but that decoding of the broadcast can be mediated through cryptographic decoding keys.

It should be noted that there can be multiple forms of communication, which can comprise messaging communications using the TCP/IP protocols, versus multicasting using

UDP protocols, and also DJ 200 control signals using yet another protocol. The access to each of these communications can be controlled via different privacy hierarchies and techniques. For example, the audio multicasting will be available to all members within a cluster, while the messaging may retain different groupings of privacy (e.g. hierarchical), while the DJ control signals will generally be limited to communications between a given unit 100 and its corresponding DJs 200.

Broadcast Control Transfer

The dynamics of cluster 700 can be such that it will be desirable for a receive unit 730 to become the broadcast unit for the cluster. Such a transfer of broadcast control will generally require the acquiescence of the broadcast unit 710 user. To effect such a transfer, the user of the receiver unit 730 desiring such control will send a signal to the broadcast unit 710 expressing such intention. If the user of the broadcast unit 710 agrees, a signal is sent to all of the members of the cluster indicating the transfer of broadcast control, and providing the identifier associated with the receive unit 730 that is to become the broadcast unit 710. The broadcast unit 710 that is relinquishing broadcast control now becomes a receive unit 730 of the cluster 700.

It should be noted that the transfer of control as described above requires the manual transfer of control, such as actuation of a DJ switch. This switch can be limited to this function, or can be part of a menu system, in which the switch is shared between different functions. It is also within the spirit of the present invention that there be voice-activated control of the unit 100, in which the unit 100 further comprises a microphone for input of voice signals to a suitable controller within the unit 100, wherein the controller has voice-recognition capabilities.

In the case of a cluster 700 whose broadcast unit 710 is no longer broadcasting (e.g. it is out of range of the receive units 730, or it is turned off), the cluster can maintain its remaining membership by selecting one of the receive units 730 to become the new broadcast unit 710. Such a choice can happen automatically, for example by random choice, by a voting scheme, or by choosing the first receive unit 730 to have become associated with the broadcast unit 710. If the users of the cluster-associated units deem this choice to be wrong, then they can change the broadcast unit 710 manually as described above.

The receive unit 730 that is chosen to become the broadcast unit 710 of the cluster 700 will generally prompt its user of the new status, so that the newly designated broadcast unit 710 can make certain that it is playing music to the rest of the cluster 700. It can be

further arranged so that a newly-designated broadcast unit 710 will play music at random, from the beginning, or a designated musical piece in such case.

An embodiment of a transfer of broadcast control using IP socket communications protocols is described here. Figure 16 is a schematic block flow diagram of transfer of control between the broadcast unit 710 and the first receive unit 730. In a step 1130, the receive unit 730 requests broadcast control (designated here as "DJ" control). In a step 1132, the user of the broadcast unit 710 decides whether control will be transferred. The decision is then transferred back to the first receive unit 730 via the TCP messaging socket. If the decision is affirmative, the first receive unit 730 severs its UDP connection to the broadcast unit 710 multicast. The reason for this is to allow the receive unit 730 opportunity to prepare the beginning of its broadcast, if such time is required, and the user cannot both listen to the multicast as well as prepare its own audio selections, which occurs in a step 1136. In a step 1138, the receive unit 730 creates a multicast UDP socket with which it will later broadcast audio to other members of the cluster, while in a step 1140, the receive unit 730 creates a broadcast annunciator TCP socket with which to announce availability of the cluster, as well as to accept transfers of members from the broadcast unit 710 to itself as the new broadcast unit.

When the two new sockets (multicast and annunciator) are created, the receive unit 730 transmits the new socket addresses to the broadcast unit 710 in a step 1142. Since the other members of the cluster are guaranteed to be in contact with the broadcast unit, they can get addresses of the new, soon-to-be broadcast unit from the existing broadcast unit. In a step 1144, the original broadcast unit 710 transmits to the other cluster members (receive units 730 numbers 2-N) the addresses of the sockets on the receive 1 unit 730 that is now the new broadcast unit 710, and terminates its own multicast. The termination is performed here because the other receive units will be transferring to the new multicast, and because the original broadcast unit 710 is now becoming a receive unit 730 in the reconstituted cluster. In the step 1148, multicast of audio is now provided by the receive 1 unit 730 that has now become the new broadcast unit 710), and the original broadcast unit is listening to audio provided not by itself, but rather by the new broadcast unit.

In a step 1146, performed roughly synchronously with the step 1144, the original broadcast unit 710 transmits the socket addresses of the message handler TCP sockets of the other members of the cluster 700 (i.e. the receive units 730 numbers 2-N). In the subsequent step 1150, the original broadcast unit 710 and the receive units 730 numbers 2-N establish new messaging connections with the receive 1 unit 730 that is now the new broadcast unit

710. While there can be a set of criteria for the acceptance of a new member to a cluster, because the receive 1 unit 730 has received the message socket addresses of the other members of the cluster in the step 1144, the receive 1 unit 730 accepts new members with the socket addresses received. It should be noted that instead of socket addresses being the
5 identifiers passed, the identifiers can also be unique machine IDs, random numbers, cryptographically encoded numbers, or other such identifiers that can be transmitted from one member of the cluster to another.

It should be noted in certain embodiments, that there can be insufficient time for the new broadcast unit 710 to determine a set of music to broadcast to the members of its cluster.
10 It is within the spirit of the present invention for a user to set a default collection of music that is broadcast when no other music has been chosen. This set of music can comprise one or more discrete audio files.

Audio and DJ Choreography

One of the attractions of the present invention is that it allows users to express
15 themselves and share their expressions with others in public or semi-public fashion. Thus, it is highly desirable for users to be able to personalize aspects of both the audio programming as well as the displays of their DJs 200.

Audio

Audio personalization comprises the creation of temporally linked collections of
20 separate musical elements in "sets." These sets can be called up by name or other identifier, and can comprise overlapping selections of music, and can be created either on the unit 100 through a visual or audio interface, or can be created on a computer or other music-enabled device for downloading to the unit 100.

In addition, the unit 100 or other device from which sets are downloaded can
25 comprise a microphone and audio recording software whereby commentary, personal music, accompaniment, or other audio recordings can be recorded, stored, and interspersed between commercial or pre-recorded audio signals, much in the manner that a radio show host or "disc jockey" might alter or supplement music. Such downloads can be accessible from a variety of sources including Internet web sites and private personal computers.

Automatic Generation of DJ 200 Control Signals

In this section, we will describe the automatic and manual generation of control signals for the DJ 200 transducers. The control signals are generally made to correspond to

audio signals played on the units 100, although it is within the spirit of the present invention for such control signals to be made separate from audio signals, and to be displayed on the digital jewelry independently of audio signals played on the unit 100. Figure 20 is a time-amplitude trace of an audio signal automatically separated into beats. Beats 1180, 1182 and 1183 are denoted by vertical dashed and dotted lines and, as described below, are placed at locations on the basis of their rapid rise in low-frequency amplitude relative to the rest of the trace. As can be seen, the beats 1180 are generally of higher amplitude than the other beats 1182 and 1183, and represent the primary beats of a 4/4 time signature. The beat 1183 is of intermediate nature between the characteristics of the beats 1180 and 1182. It represents the third beat of the second measure. Overall, the audio signal thus displayed can be orally represented as ONE-two-Three-four-ONE-two-Three-four ("one" is heavily accented, and the "three" is more lightly accented), which is common in the 4/4 time signature.

Processing of this data can proceed via a number of different methods. Figure 21A is a block flow diagram of a neural network method of creating DJ 200 transducer control signals from an audio signal as shown in Figure 20. In a step 1200, audio data is received either at the unit 100 or the DJ 200. It should be noted that the creation of control signals from audio signals can, within the present invention, take place at either the unit 100 or the DJ 200, or even at a device or system not part of or connected to the unit 100 or DJ 200 (as will be described in more detail below). In an optional step 1202, the data is low pass filtered and/or decimated so that the amount of data is reduced for computational purposes. Furthermore, the data can be processed for automatic gain to normalize the data for recording volume differences. Furthermore, the automatic gain filtering can provide control signals of significant or comparable magnitude throughout the audio data.

In general, the creation of the audio signal depends on audio representing a period of time, which can be tens of milliseconds to tens of seconds, depending on the method. Thus, the audio data from the step 1202 is stored in a prior data array 1204 for use in subsequent processing and analysis. At the same time, the current average amplitude, computed over an interval of preferably less than 50 milliseconds, is computed in a step 1208. In broad outline, the analysis of the signal compares the current average amplitude against the amplitude history stored in the prior data analysis. In the embodiment of Figure 21A, the comparison takes places through neural network processing in a step 1206, preferably with a cascading time back propagation network which takes into account a slowly varying time signal (that is, the data in the prior data array changes only fractionally at each computation, with most of the data remaining the same). The use of prior steps of neural network processing in the

current step of neural network processing is indicated by the looped arrow in the step 1206. The output of the neural network is a determination whether the current time sample is a primary or a secondary beat. The neural network is trained on a large number of different music samples, wherein the training output is identified manually as to the presence of a beat.

5 The output of the neural network is then converted into a digital jewelry signal in a step 1210, in which the presence of a primary or secondary beat determines whether a particular light color, tactile response, etc., is activated. This conversion can be according to either fixed, predetermined rules, or can be determined by rules and algorithms that are externally specified. Such rules can be according to the aesthetics of the user, or can
10 alternatively be determined by the specific characteristics of the transducer. For example, some transducers can have only a single channel or two or three channels. While light transducers will generally work well with high frequency signals, other transducers, such as tactile transducers, will want signals that are much more slowly varying. Thus, there can be algorithm parameters, specified for instance in configuration files that accompany DJ 200
15 transducers, that assist in the conversion of beats to transducer control signals that are appropriate for the specific transducer.

Figure 21B is a block flow diagram of a deterministic signal analysis method of creating DJ 200 transducer control signals from an audio signal as shown in Figure 20. The data is received in the step 1200. In this case, a running average over a time sufficient to
20 remove high frequencies, and preferably less than 50 milliseconds, is performed in a step 1212. Alternatively, a low pass filter and/or data decimation as in the step 1202 can be performed.

In a step 1214, the system determines whether there has been a rise of X-fold in average amplitude over the last Y milliseconds, where X and Y are predetermined values.
25 The value of X is preferably greater than two-fold and is even more preferably three-fold, while the value of Y is preferably less than 100 milliseconds and is even more preferably less than 50 milliseconds. This rise relates to the sharp rises in amplitude found in the signal at the onset of a beat, as shown in Figure 20 by the beat demarcations 1180, 1182, and 1183. If there has not been a rise meeting the criteria, the system returns to the step 1200 for more
30 audio input.

If the signal does meet the criteria, it is checked to ensure that the rise in amplitude is not the "tail end" of a previously identified beat. For this, in a step 1216, the system determines whether there has been a previous beat in the past Z milliseconds, where Z is a predetermined value preferably less than 100 milliseconds, and even more preferably less

than 50 milliseconds. If there has been a recent beat, the system returns to the step 1200 for more audio input. If there has not been a recent beat, then a digital jewelry signal is used to activate a transducer. The level of transduction can be modified according to the current average amplitude which is determined in a step 1208 from, in this case, the running average
5 computed in the step 1212.

The embodiment of Figure 21B provides transducer activation signals at each rapid rise in amplitude, with the activation signal modulated according to the strength of the amplitude. This will capture much of the superficial musical quality of the audio signal, but will not capture or express more fundamental patterns within the audio signal.

10 Figure 21C is a schematic flow diagram of a method to extract fundamental musical patterns from an audio signal to create DJ 200 control signals. In the step 1200, the audio data is received into a buffer for calculations. In a step 1220, a low pass filter is applied to remove high frequency signal. Such high frequency signals can alternatively be removed via decimation, running averages, and other means as set forth in the embodiments of Figures
15 21A and B. As in the embodiment of Figure 21B, beat onsets are extracted from the audio signal in the steps 1214 and 1216, and a current average amplitude is computed in a step 1208.

The amplitudes and times of the onsets of beats are placed into an array in a step 1222. From this array, a musical model is created in a step 1224. This model is based on the
20 regularity of beats and beat emphasis – as seen in the amplitudes – that is independent of the beats and amplitudes in any one short section of music (corresponding, for instance, to a measure of music).

In general, music is organized into repeating patterns, as represented in a time signature such as 3/4, 4/4, 6/8 and the like. Within each time signature, there are primary and
25 secondary beats. In general, the downbeat to a measure is the first beat, representing the beginning of the measure. The downbeat is generally the strongest beat within a measure, but in any given measure, another beat may be given more emphasis. Indeed, there will be high amplitude beats that may not be within the time signature whatsoever (such as an eighth note in $\frac{3}{4}$ time that is not on one of the beats). Thus, by correlating the beats to standard
30 amplitude patterns, the output to the music model identifies the primary (down) beats, secondary beats (e.g. the third beat in 4/4 time) and the tertiary beats (e.g. the second and fourth beats in 4/4 time).

Figure 21D is a schematic flow diagram of an algorithm to identify a music model, resulting in a time signature. In a step 1600, the minimum repeated time interval is

determined, using the array of beat amplitude and onset 1222. This is, over a period of time, the shortest interval for a quarter note equivalent is determined, wherein the time signature beat frequency (i.e. the note value of the denominator of the time signature, such as 8 in 6/8) is preferably limited to between 4 per second and one every two seconds, and even more preferably limited to between 3 per second and 1.25 per second. This is considered the beat time.

From the array of beat amplitudes and onsets 1222, the average and maximum amplitudes over a time period of preferably 3-10 seconds is computed in a step 1604. For the beginning of the audio signal, shorter periods of time can be used, though they will tend to give less reliable DJ 200 control signals. Indeed, in this embodiment, the initial times of an audio signal will tend to follow audio signal amplitude and changes in amplitude more than fundamental musical patterns until the patterns are elicited.

In a step 1606, the amplitude of a beat is compared with the maximum amplitude determined in the step 1604. If the beat is within a percentage threshold of the maximum amplitude, wherein the threshold is preferably 50% and more preferably 30% of the maximum amplitude, the beat is designated a primary beat in a step 1612. In a step 1608, the amplitude of non-primary beats is compared with the maximum amplitude determined in the step 1604. If the beat is within a percentage threshold of the maximum amplitude, wherein the threshold is preferably 75% and more preferably 50% of the maximum amplitude, and the beat is greater than a predetermined fraction of the average amplitude, wherein the fraction is preferably greater than 40% and even more preferably greater than 70% of the average beat amplitude, the beat is designated a secondary beat in a step 1614. The remaining beats are denoted tertiary beats in the step 1610.

In a step 1616, the sequence of the three types of beats is compared with that of established time signatures, such as 4/4, 3/4, 6/8, 2/4 and others, each with their own preferred sequence of primary, secondary and tertiary beats, in order to determine the best fit. This best fit is identified as the time signature in a step 1618.

Returning to Figure 21C, the channels of the DJ are pre-assigned to four different beats in a step 1225. Thus, if there are four channels, each channel is given a separate assignment. With a smaller number of channels, a single channel is assigned multiple beats. Some beats can also be unassigned, thus not being represented in a DJ 200 transducer output. Thus, a high jewelry signal, medium jewelry signal, low jewelry signal and an amplitude dependent signal are each assigned to a channel for DJ 200 transduction.

In a step 1226, a beat determined to be a primary/down beat is assigned to a high jewelry signal 1228. In a step 1230, a beat determined to be a secondary beat is assigned to a medium jewelry signal 1232. In a step 1234, a beat determined to be a tertiary beat is assigned to a low jewelry signal 1236. Beats which are then unassigned, and which will
5 generally be beats that occur not within the music model of the step 1224 (e.g. rapid beats not falling on beats of the time signature) are then assigned in a step 1238 to an amplitude dependent (and not music model dependent) signal 1240.

It should be noted that the computations performed in the flow methods of Figures 21A-C may take time on the order of milliseconds, such that if the computations are made in
10 real time during the playing of music, the activation of the transducers in the DJ 200 are "behind" in time relative to the audio playing of the corresponding music in the audio unit 100. This can be compensated for by carrying out the computations while the audio signal is still in buffers prior to being played in the unit 100, as is described above for numerous embodiments of the present invention. Thus, signals to the DJ 200 can then be made
15 simultaneously with respect to the audio signal to which it corresponds.

It should be noted that many of the parameters described above can conveniently be affected by manual controls either on the DJ 200 or the unit 100 that transmits signals to the DJ 200. For example, it can be convenient for the user to be able to set, for a given DJ 200 response amplitude, the threshold audio amplitude level at which the output transducer (e.g.
20 light transducer 240) responds, or to set the output transducer amplitude corresponding to a maximum audio amplitude, or to set the frequency bands for which different DJ 200 channels respond, or to set other similar parameters. The manual controls for such parameters can comprise dials, rocker switches, up/down button, voice or display menu choices, or other such controls as are convenient for users. Alternatively, these choices can be set on a
25 computer or other user input device, for download onto the unit 100 or DJ 200.

A preferable means of setting the parameters is for the parameters to be stored in a configuration file that can be altered either on the unit 100, the DJ 200 or a computer, so that the same DJ 200 can take on different characteristics dependent on the configuration settings within the file. The configuration settings can then be optimized for a particular situation, or
30 set to individual preference, and be traded or sold between friends or as commercial transactions, for instance over the Internet. For a most preferable use of these configuration files, each file with its set of configurations can be considered to represent a "mode" of operation, and multiple configuration files can be set on the DJ 200 or the unit 100, depending on where the automatic generation of control signals is performed. The user can

then select from the resident configuration files, appearing to the user as different modes, for use of his system, and can change the mode at will. This can be arranged as a series of choices on a voice or display menuing system, as a list toggled through by pressing a single button, or through other convenient user interfaces.

5 Manual Generation of DJ 200 Control Signals

 In the description above, the use of filtering and digital modification of audio signals can be used to create control signals for DJ 200 transducers 240, 250, and 260. In addition, manual choreography of DJ 200 signals can be accomplished. For example, buttons or other interface features (e.g. areas on a touch-screen) on the unit 100 can correspond to different
10 arrays of transducers, such as the LED arrays 290 and 292 of Figure 2A. While playing the audio signal, the user can press the buttons, where pressing of the buttons can correspond to a control signal for a transducer being ON, and otherwise the signal can be off. To aid in choreography where rapid changes in transducers are desired, the audio can be played at less than normal speed.

15 Figure 22A is a top-view diagram of an audio unit 100 user interface 1250, demonstrating the use of buttons to create DJ 200 control signals. The interface 1250 comprises a display screen (e.g. LCD or oLED), which can display information to the user, such as shown in Figures 18 A-B. Standard music control buttons 1254 for playing, stopping, pausing, and rewinding allow the user to control the audio signal musical output. Buttons
20 1252 further control aspects of the music output, such as volume control, musical tracks, and downloading and uploading of music. The number of buttons 1252 is conveniently three as shown, but can be more or less than three.

 In addition, buttons are provided to allow the user to input DJ 200 control signals, comprising a record button 1256, a first channel button 1258, a second channel button 1260
25 and a third channel button 1262. The channel buttons 1258, 1260 and 1262 are prominent and accessible, since the user will want to easily depress the buttons. A record button 1256 allows the user to activate the channel buttons 1258, 1260 and 1262, and has a low profile (even below the nominal surface of the interface 1250) so that it is not accidentally activated. The record button can serve various purposes, including recording into a permanent storage
30 file the sequence of DJ control signals relative to music being played, or controlling the DJ transducers in realtime, synchronously with music being played on the audio unit 100.

 Pressing the buttons 1258, 1260 and 1262 create DJ control signals for the corresponding channels. The number of buttons is conveniently three as shown, but can also

be two or four or more buttons. If a telephone is being used as the unit 100, keys on the telephone keypad can alternatively be used. The channel buttons will generally be used with thumbs, and the buttons are spaced so that two of the buttons can be depressed with a single thumb, so that all three buttons can be activated with only two fingers. In is also convenient
5 for the two secondary buttons 1260 and 1262 to be spaced more closely together, as it will be a preferred mode of operation that the secondary buttons be operated together from time to time.

To further aid in the choreography of the DJs 200, a separate "keyboard" with the number of keys related to the number of possible arrays can be used. The amplitude of the
10 corresponding transducer signal can be modified either according to the pressure on the keys, according to the length of time that a key is depressed, or according to a foot pedal. Figure 22B is a top-view diagram of a hand-pad 1270 for creating DJ control signals. The hand-pad 1270 comprises a platform 1271, a primary transducer 1272, a secondary transducer 1274 and a tertiary transducer 1276. The platform 1271 has a generally flat top and bottom, and can
15 conveniently be placed on a table, or held in the user's lap. The size of the platform is such that two hands are conveniently placed across it, being preferably more than 6 inches across, and even more preferably more than 9 inches across. The pressure transducers 1272, 1274 and 1276 respond to pressure by creating a control signal, with said control signal preferably capturing both the time and amplitude of the pressure applied to the corresponding
20 transducer. The primary transducer 1272 creates a primary control signal, the secondary transducer 1274 creates a secondary control signal and the tertiary transducer 1276 creates a tertiary control signal. The sizes and placements of the transducers can be varied within the spirit of the present invention, but it is convenient for the primary transducer 1272 to be larger and somewhat separate from that of the other transducers 1274 and 1276. In one more
25 method of user interaction, both hands can be rapidly and alternately used to make closely spaced control signals on the primary transducer 1272. In addition, it can be convenient on occasion for the user to activate both the secondary transducer 1274 and the tertiary transducer 1276 with different fingers on one hand, and thus these can be conveniently placed relatively near to one another. In general, while a single transducer will provide minimal
30 function, it is preferable for there to be at least two transducers, and even more preferable that there be three transducers.

The control signals can be transferred to the audio unit 100 for playing and/or storage, or to the DJ 200 unit directly for playing, either wirelessly, or through wired communication. In addition, the hand-pad can also be configured to create percussive or other sounds, either

directly through the incorporation of hollow chambers in the manner of a drum, or preferably by the synthesis of audio waveform signals that can be played through the audio unit 100 (and other audio units 100 participating in a cluster 700), or directly through speakers within the hand-pad 1270 or attached to the hand-pad 1270 through wired or wireless communications. Such audible, percussive feedback can aid the user in the aesthetic creation of control signals.

It is within the spirit of the present invention for the hand-pad to take on various sizes and configurations. For instance, it is also convenient for the hand-pad 1270 to be configured for the use of index and middle fingers, being of dimensions as small as two by four inches or less. Such a hand-pad is highly portable, and can be battery powered.

Additionally, DJ 200 control signals can also be manually generated live, during broadcast at a party, for example, by a percussionist playing a set of digital drums. Figure 22C is a schematic block diagram of a set of drums used for creating DJ control signals. The set of drums comprises four percussive instruments 1280, 1282, 1284 and 1286, which can include snare drums, foot drums, cymbals, foot cymbals and other percussive musical instruments, such as might be found with a contemporary musical "band". Microphones 1290 are positioned so as to receive audio input primarily from instruments to which they are associated. One microphone can furthermore be associated with multiple instruments, as with the drums 1282 and 1284. The microphones 1290 are connected with a controller 1292 that takes the input and creates DJ control signals therefrom. For example, the drums 1282 and 1284 can be associated with the primary channel, the drum 1280 can be associated with the secondary channel, and the drum 1286 can be associated with the tertiary channel. The association of the microphone input with the channel can be determined in many ways. For example, the jack in the controller 1292 to which each microphone 1290 attaches can correspond to a given channel. Alternatively, the user can associate the jacks in the controller to different channels, with such control being manual through a control panel with buttons or touch control displays, or even through prearranged "sets". That is, a set is a pre-arranged configuration of associations of microphones to channels, and thus a set can be chosen with a single choice that instantiates a group of microphone-channel associations.

In general, the inputs from the microphones 1290 will be filtered in frequency and also to enhance audio contrast. For instance, control signals can be arranged to be the highest when the low-frequency envelope is rising the quickest (i.e. the beat or sound onset). The algorithms for conversion of audio signal to DJ control signal can be pre-configured in the controller 1292, or can be user selectable.

It should be noted that the methods and systems of Figures 22 A-C need to synchronize the control signals so generated with the audio files to which they correspond. This can be accomplished in many ways. For example, the first control signal can be understood to correspond to the first beat within the audio file. Alternatively, the audio unit
5 100 or other device that is playing the audio signal to which the control signal is to correspond can send a signal to the device that is creating the control signals indicating the onset of playing of the audio file. The control signal can then be related to the time from the onset of the audio file. In addition, with regards to this synchronization, the user manually inputting the control signals will always be listening to the music during the control signal
10 input. If the device on which control signals are being input is the same as the device that is playing the music, a control signal input can be easily related to the sound that is currently being played by the audio output – many such devices allow information to within less than a millisecond of what sample or time within the audio files is currently being output by the audio device. With the arrangement of the control signal input device being also an audio
15 player, close calibration of the control signals and the audio output is easily accomplished.

DJ 200 Control Signal Files

The control signals can be in a variety of formats within the spirit of the present invention. Such formats include pairs of locations within the associated music file and the corresponding amplitudes of the various DJ channels, and pairs of locations and the
20 amplitudes of those DJ channels which are different from before. The locations can be either time from the start of the song (e.g. in milliseconds) or in terms of sample number. If the location is given in terms of sample number, the sample rate of the music will generally also be provided, since the same song can be recorded at different sample rates, and the invariant in terms of location will generally be time from onset of the music.

Other formats include an amplitude stream, corresponding to each DJ channel,
25 provided in a constant stream with a fixed sample rate, which may be equal to or different from that of the corresponding music file. This format can be stored, for example, as additional channels into the music file, such that one channel corresponds to monoaural sound, two channels correspond to stereo sound, three channels correspond to stereo sound
30 and one channel of control signals, and additional channels correspond to stereo sound plus additional channels of DJ control signals. Another arrangement is to allow for only a small number of states of the transduction in the control signal, so that multiple channels of control signal can be multiplexed into a single transmitted channel for storage and transmission with

the audio signal. For example, if the audio is stored as a 16-bit signal, 3 channels of 5 bit DJ 200 control signal could be stored in a single channel along side the one or two audio channels normally used.

It should be appreciated that these different control signal storage formats are largely
5 interchangeable. For instance, as described above, control signals can be stored as if they are additional audio channels within a music file, but then be extracted from the file for separate transfer (e.g. over the Internet), and then be reintegrated into an audio file at the destination location.

It should be appreciated that there are a number of means by which DJ 200 control
10 signals can be generated, either automatically or manually, and can include the use of devices other than the unit 100 that can have sophisticated digital or analog filtering and modification hardware and software. The control signals so created can be stored in files that are associated with the music files (e.g. MP3) that the control signals are meant to accompany. To aid in their distribution, particularly in reference to limitations on the commercial and
15 private distribution of the corresponding music files, the signal files will generally be separate from the music files, and transferable between units 100 either through inter-unit communication mediated by the inter-unit transmitter/receiver 110, or alternatively through computers or computer networks to which the unit 100 can be connected.

The audio signals and the DJ control signals should also be well synchronized during
20 playback. Figure 23 is a schematic block flow diagram of the synchronized playback of an audio signal file with a DJ control signal file, using transmission of both audio and control signal information. For purposes of convenience in discussion, the audio signal file will be called a "song file" and the "control signal file" will be called a "dance file." In a step 1300, the user is provided a list of song files for display, preferably on the display 1170. In a step
25 1302, the user then selects a song from the display to play. In a step 1304, the dance files that are associated with the selected song file from the step 1302 are displayed for the user. These song files can be either locally resident on the unit 100, or can alternatively be present on other audio units 100 to which the audio unit 100 is connected, as in a cluster, or can alternatively be on the Internet, if the audio unit 100 is connected to the Internet. If there is a
30 dance file that has been previously preferred in association with the song file, this file can be more prominently displayed than other associated dance files.

In a step 1306, the user selects the dance file to play along with the song file. This association is stored in a local database of song file/dance file associations in a step 1307, to be later used in a subsequent step 1304, should such an association not have been previously

made, or if the preferred association is different from the previously preferred association. If the dance file is not locally resident, it can be copied to the audio unit 100 to ensure that the dance file is available throughout the duration of the song file playback.

5 In a step 1308, a timer is initialized at the beginning of the song file playback. In the step 1310, the song file is played on the local unit 100, and is also streamed to the other units 100 within the cluster 700. The corresponding DJ control signal accompanies the streaming song, either multiplexed within the song file audio signal, on another streaming socket, or through other communications (e.g. a TCP socket) channels between the two units. In a step 1312, the time advances along with the playback of the music. In a step 1314, this timer
10 information is used to obtain current control signals from the dance file – that is, the dance file is arranged so that at each moment, the status of the different transducer channels can be determined. The control signals to be streamed along with the song file information can be either the current status of each transducer, or alternatively, can only send changes from the current transducer state.

15 The matching of the files in the database of song file and dance file associations of the step 1307 can be performed both within a machine, but also over a local or wide area network. In such cases, the association can either be external to the file – that is, using the name of the file, that is available the normal system file routines – or can use information internal to one or both files. For example, the dance file can have stored within it a reference
20 to the song to which it is associated, either as the name of the song file, the name and/or other characteristics of the song (such as the recording artist, year of publication, music publisher) or alternatively as a numerical or alphanumerical identifier associated with the song. Then, given a song file, the relationship of the dance file with the song file can be easily determined.

25 For ease in creating an association, it is convenient for the names of the song files and the associated dance files to have a relationship with one another that is easily understood by casual users. For example, given a song file with the name “oops.mp3”, it is convenient for an associated dance file to share the same root (in this case “oops”) with a different extension, creating for example the dance file name “oops.dnc”. Because of the multiplicity
30 of dance files that will often be associated with a particular song file, the root itself can be extended to allow for either a numerical or descriptive filename, which can be preferably done in conjunction with a known punctuation mark to separate the song file root from the dance file description, such as the file names “oops.david2.dnc” or “oops\$wild.dnc”. It is

preferable to use a punctuation mark that is allowed within a range of different operating systems.

Dance files can be stored on the Internet or other wide area network in a store for access by users who want dance files associated with a particular song file. In such case, if the storage is through the root of the filename, the user, requesting dance files corresponding to "oops.mps" would then be returned the names of related files such as "oops\$wild.dnc". If the dance file internally carries the relationship with "oops.mps" as described above, either through the name or other characteristics, or alternatively, through a numerical or alphanumeric identifier, it is preferable to store the information in a database on the storage computer or unit 100, so that it is not necessary to open the file each time for perusal of the dance file information. Thus, if the music file has a substantially unique identifier associated with it internally, it is also useful for the dance file to also have the same identifier associated internally as well. In such case, the identifier is conveniently used to reference both files within a database.

In operation, a remote user would request a dance file for a particular song file by providing the name of the song file, along possibly with other information about the song file, which could include the name of the choreographer, the number of channels of DJ 200 transduction, the specific brand or type of DJ 200, or other information. The database would then return a listing of the various dance file that met the criteria requested. The remote user would then choose one or more of the files to download to the remote computer, and then the database would retrieve the dance files from storage and then transmits the dance file over the wide area network. On the remote computer or unit 100, the dance file would become associated with the corresponding song file through means such as naming the dance file appropriately or making an association between the song file and the dance file in a database or indexing file. Alternatively, the dance file can be integrated into the song file as mentioned elsewhere within this specification.

It can be useful to preview a dance file for its desirability or suitability. Since the dance files can be retrieved from a wide area network such as the Internet, it is convenient for such an emulator to operate on a computer that may not be portable or have the proper transmitter that allows communications with a DJ 200. In such case, it is preferable to have an emulator which places an image or drawing of a DJ 200 on the screen, and which is provided the name of a song file and a dance file, and which then plays the song file through the audio of the computer and displays appropriate images or drawings of transducers being activated within the emulator image or drawing. The characteristics of the DJ 200 being

emulated (e.g. colors of lights, frequency responses, levels of illumination, arrangement of lights, response to amplitude, etc.) can be simulated by a number of means. For example, the user can move slider controls, set checkboxes and radio boxes, enter numerical values, click-and-drag icons and use other standard user interface controls to make the DJ 200 operate as desired. Alternatively, manufacturers of DJ 200s can create configuration files (including, for example, bitmaps of photos of the actual DJ 200) that can be downloaded for this purpose (and which can also be used by prospective purchasers to view the "virtual" operation of the DJ 200 prior to purchase, for example, through an Internet merchant). The configuration files would contain the information necessary for the emulator to properly display the operation of the specific DJ.

Alternatively, as described above, the dance file information can be stored within the song file as, for example, another channel in place of an audio channel, or alternatively within MP3 header or other file information. In such case, the step 1307 would have the alternative function of looking through song files to find the song file with the particular desired embedded dance file within.

In addition to sending dance files from computers to units 100 or between units 100, the dance files can be streamed from unit 100 to unit 100 through the normal unit-to-unit communications, in the manners described above for audio communications. This is particularly convenient given that DJ 200 displays can be used to show group identification, and such displays can be more effective if the DJs for each user are nearly identical (which might not be the case if the users were using, for example, different dance files). The dance file control signal information can be transmitted in a variety of ways, including multiplexing the control signals into the same packets as the audio information as if it were a different audio channel, alternating packets of control signals with packets of the audio information, or broadcasting control signals on a different UDP socket as the audio. Alternatively, if the receiving unit has a copy of the dance file corresponding to the song file being transferred by unit-to-unit communication, the receiving unit can determine the current time being played, and to extract from the local dance file the control signals for the receiving unit DJ 200.

It should be known that most streaming protocols have relatively small data packets that are communicated, due to the fact that reception at the source is not guaranteed and it is not desirable to lose a large amount of information in any one stream. Thus, it is possible with smaller transmission buffers and higher data rates to send a single DJ control signal in each transmission. For example, with a buffer size of 600 bytes, and an audio rate of 22,050 Hz with two single byte channels, each transmission covers only about 12 milliseconds, and

any signal would therefore be at most 13 milliseconds from its correct time. Alternatively, each control signal can be accompanied by an offset in time from the beginning of the transmitted audio signal. Also, the time or packet number of each transmission buffer can be sent, as well as the time or packet number of the DJ audio signals, so that the audio unit 100
5 can compute the proper offset.

Stationary Transducers

DJs 200 that have been previously described are portable devices, usually associated with a particular user and unit 100. Figures 5A and 5B indicate the ways in which DJs 200 associated with multiple users can be controlled by a single unit 100.

10 It is also convenient for transducers to be non-portable and stationary. Consider, for example, a user who is at home listening to music. Instead of a DJ 200 worn by the user, the user can alternatively have a bank of lights or other transducers in fixed locations through the room that operate under the same or similar control signals as to which DJs respond. Such fixed transducers can operate at far higher power than portable DJs 200, and can each
15 incorporate a large number of separate transducers.

Furthermore, in a party, concert or other large social gathering, the effects of portable DJs worn by guests can be supplemented by large transducers that are generally perceptible by most guests. For example, such transducers can include spark or smoke generators, strobe lights, laser painters, arrays of lights similar to Christmas light strings, or mechanical devices
20 with visible (e.g. a flag waving device) or tactile effects (e.g. a machine that pounds the floor). In general, transducers for large gatherings will not communicate with a unit 100, but will be directed by a wide-area broadcast unit 360, as in Figure 5B.

Because of the large area over which such stationary transducers can operate, the communications between the unit 100 and the stationary transducers can be through wired
25 rather than wireless transmission. Furthermore, there can be mixed communication, such as wireless transmission of control signals from a portable unit 100 to a stationary receiver, and thence wired transmission to one or multiple transducers.

Modular Configurations

In the embodiments above, the audio player 130 is directly integrated with the inter-
30 unit and unit-to-DJ communications. This requires both a re-engineering of existing audio players (e.g. CD, MP3, MO and cassette players), and furthermore does not allow the communications functionality to be reused between players.

An alternative embodiment of the present invention is to place the communications functions external to the audio playing functions, and to adjustably connect the two via the audio output port of the audio player. Figure 12A is a schematic diagram of a modular audio unit 132. Audio player 131 is a conventional audio player (e.g. CD or MP3 player) without the functionality of the present invention. Analog audio output is sent via audio output port 136 through the cable 134 to the audio input port 138 of the modular audio unit 132. The modular audio unit 132 comprises the inter-unit transmitter/receiver 110 and the DJ transmitter 120, which can send and receive inter-unit and unit-to-DJ communications in a manner similar to an audio unit 100. A switch 144 chooses between audio signals from the audio player 131 and from the inter-unit transmitter/receiver 110 for output to the output audio port 142 to the earphone 901 via cable 146 (the earphone 901 can also be a wireless earphone, wherein the output port 142 can be a wireless transmitter, which can also be a DJ transmitter 120). A convenient configuration for the switch 144 is a three way switch. In an intermediate position, the unit 132 acts simply as a pass-through, in which output from the audio player 131 is conveyed directly to the earphone 901, and the transmitter/receiver functions of the unit 132 do not operate. In another position, the unit 132 operates as a receiver, and audio from the inter-unit transmitter/receiver 110 is conveyed to the earphone 901.

When the combined system operates as a broadcast unit 710, audio input from the audio unit 131 is directed to the inter-unit transmitter/receiver 110 for transmission to receive units 730, as well as for output to the earphone 901 (which can be direct to the earphone 901 through the switch, or indirectly through the inter-unit transmitter/receiver 110).

When the combined system operates as a conventional audio player, the switch directs audio signals from the input port 138 directly through to the output port 142. In this mode of operation, it can be arranged for the audio output to traverse the modular audio unit 132 without the unit being powered up. In case there is a transmission delay to the receive unit 730 such that audio played locally through the earphone 901 and audio played remotely on the receive unit 730 are not in synchrony, the system can incorporate a time delay in the output port 142 such that the local and remote audio output play with a common time delay, and are thus in synchrony.

When the combined system operates as a receiver unit 730, audio input from the input port 138 is ignored, and signals to the audio output port are delivered solely through the inter-unit transmitter/receiver 110.

It is convenient for the modular audio unit 132 to be able to operate independently of the associated audio player 131. In such a case, the unit 132 must have an independent energy store, such as one or more batteries, which can be rechargeable. In that case, the unit 132 has no audio signals locally to listen to through the earphone 901 or to transmit over the transmitter/receiver 110. However, the unit 132 can in that case receive external audio signals sent by other units 132 or units 100 for listening.

The audio player 131 can be placed in a backpack, purse, or other relatively inaccessible storage location, while the modular audio unit is, like a "remote control", accessible for interaction with other users.

Video

While the units 100 described above have comprised audio players 130, within the spirit of the present invention, such units can also comprise video or audio/visual players (both of which are referred to below as video players). Such video players would be used generally for different entertainment and educational purposes, not limited to films, television, industrial training and music videos. Such video enabled units can operate similarly to audio units, including the capability of sharing video signals, synchronously played, with nearby units through inter-unit communication, as well as the use of DJ's that can produce human-perceptible signals (such as light transduction for accompaniment of audio signals in music videos). It should be noted, however, that there is a larger bandwidth requirement for the inter-unit transmitter/receiver 110 for the communication of video signals as compared with audio signals. In the case of shared video, wire connections (e.g. FireWire) between two units can allow simultaneous viewing of a single video signal.

In addition, text, including language-selectable closed caption and video subtitling, can accompany such video, as well as chat or dubbing to allow the superposition of audio over the audio normally accompanying such video.

Music Distribution Using Audio Units

The music industry is suffering from reduced sales due to the advent of Internet-based music file sharing; in addition, the manufacturers of personal audio devices are bringing to market audio devices that can wirelessly transfer music files between the devices. Such sharing-enabled devices could significantly reduce the sales of music. Audio units of the present invention, however, can be used to provide new means of music distribution and thereby increase the sales of music.

Figure 25 is a schematic flow diagram indicating music sharing using audio devices, providing new means of distributing music to customers. Three entities are involved in the transactions – the DJ (operating a broadcast unit 710), the cluster member (operating a receive unit 730), and the music distributor, and their actions are tracked in separate columns.

5 In this case, the term DJ is used to indicate the person operating a broadcast unit 710, and has no meaning with respect to a DJ unit 200. Indeed, the DJ unit 200 is a part of the system only inasmuch as it provides for heightened pleasure of the DJ and the member in enhancing their experience of the music. For the rest of this section, DJ will refer specifically to the person operating the broadcast unit 710.

10 In a first step 1340, the DJ registers with the distributor, who places information about the DJ into a database in a step 1342. Part of this information is a DJ identifier (the DJ ID), which is unique to the DJ, and which DJ ID is provided to the DJ as part of the registration process. This ID is stored in the unit 100 for later retrieval. The DJ at some later time broadcasts music of the type distributed by the distributor, in a step 1344. The broadcast of
15 the music by the DJ can be adventitious (that is, without respect to the prior registration of the DJ with the distributor), or the distributor can provide the music to the DJ either free of charge, at a reduced charge, or free of charge for a limited period of time.

In a step 1346, the member becomes a part of the cluster 700 of which the DJ is the broadcaster broadcasting the distributor's music, and has thereby an opportunity to listen to
20 the music. Along with the transfer of the audio signal of the music, in a step 1348, the DJ can send information about the song, which can include a numerical identifier of the music or album from which the music is derived. Furthermore, the DJ ID is provided to the member, and is associated with the music ID and stored in a database on the member unit 100 in a step 1350. In order to prevent this database from becoming too sizable, music IDs and DJ IDs can
25 be purged from it on a regular basis (for example, IDs which are older than 60 or 120 days can be removed).

If the member requests purchase of the music from the distributor in a step 1352, in a step 1354, the distributor stores the member information, the music ID, and the DJ ID associated with the music (i.e. the person who introduced the member to the music). The
30 distributor then completes the transaction with the member, providing a copy of the music in exchange for money, in a step 1356. As the member receives the music copy, he also becomes registered as a DJ as well in a step 1358. Thus, if the member now becomes the DJ of his own cluster, and introduces people to this music, he will also be known to the distributor as an introducer of the music.

In a step 1360, the distributor provides points to the DJ who introduced the member to the music and facilitated the sale of the music. In a step 1362, the DJ accumulates points related to the sale of the music to the member, as well as points related to the sale of other music to other members. These points can at that point or later be redeemed for money,
5 discounted music, free music, gifts, access to restricted activities (e.g. seats at a concert) or other such real or virtual objects of value to the DJ.

In a step 1364, the DJ is optionally further linked to the music and member for whom he has received points. If this member introduces the music to yet other members, who are induced to buy the music from the distributor, the DJ is further awarded points in a step 1366,
10 given that the "chain" of members introduced directly or indirectly to the music includes the original DJ.

This set of interactions does not decrease music sales as does file sharing, but rather increases sales of music, as the DJ has incentives to encourage others to buy the music, and the offering of the music by the DJ through his broadcasts introduces music to people who
15 may not have already had the opportunity to hear the music.

Figure 31 contains tables of DJ, song and transaction information according to the methods of Figure 25. A USER table 1810 comprises information about the USER, which can include the name of the person (Alfred Newman), their nickname/handle ("WhatMeWorry"), their email address (AEN@mad.com), and the machine ID of their unit
20 100 (B1B25C0). This information is permanently stored in the audio unit 100. A second set of information relates to music that the USER has heard while in other clusters 700 that the USER liked, and which is indicated as the USER's "wish list". This set of information includes a unique ID associated with the song (or other music or audio signal), which is transmitted by the broadcast unit 710 of the cluster 700. This information can alternatively or
25 additionally include other information about the music, such as an album name, an artist name, a track number, or other such information that can uniquely identify the music of interest.

Along with each song ID is a DJ identifier, indicating the unique ID associated with the DJ who introduced the desired music to the USER. Additionally or alternatively, the
30 information can comprise the DJ's email address, personal nickname/handle, name, or other uniquely identifying information.

The Wish List can either be permanent, or it can be that each song entry is dated, and that after a predetermined amount of time, which can be set by the user, the songs that are still on the Wish List are removed. It is also convenient that songs that are purchased

according to the methods of the present invention, such as Figure 25, are also removed from the list automatically.

A DISTRIBUTOR table 1812 comprises information about purchases made by USERS with the DISTRIBUTOR. The table 1812 has numerous records keyed according to unique USER identifiers, which in this case is the MAC ID of the unit 100. A single record from the table is provided, of which there can be hundreds of thousands or millions of such records stored.

The record can include contact information about the USER, including name, email address, or other business related information such as credit card number. In addition, each record comprises a list of all of the songs known to have been purchased through the DISTRIBUTOR, as identified by a unique song ID. In addition, the DJ associated with the purchase of the given song by the USER is also noted. This information was previously transmitted from the USER table 1810, which includes the associated DJ identifier along with the song identifier, at the time of purchase of the song. This association allows the DISTRIBUTOR to compensate the DJ for his part in introducing the USER to the song.

It should also be noted that such an arrangement of information allows the compensation, if desired, of the individual who introduced the DJ to the song, prior to the DJ introducing the USER to the song. For example, when the user purchased the song with song ID 230871C40, points were credited with the DJ whose ID is 42897DD. Looking in the record for the DJ 42897DD, one can determine whether there is another individual (DJ) associated with the purchase of the song 230871C40 by the DJ. If so, that individual can also receive compensation for the purchase of the song by the USER.

Use of Internet Connections

It is within the teachings of the present invention to allow normal Internet connections of the audio unit 100 with non-mobile devices connected with the Internet. Figure 29A is a schematic block diagram of the connection of an Internet-enabled audio unit 100 with an Internet device through the Internet cloud 1708, using an Internet access point 1704. An Internet-enabled audio unit 1700, unit A, is wirelessly connected to an audio unit 100, denoted unit B, as members in a cluster 700. The dashed line connecting the two units A and B indicates that the connection is wireless, whereas the solid connecting lines indicate wired connections. The unit A is connected to a wireless access point 1704, such as an 802.11 access point, which is connected to an Internet device 1706 via wired connections through the Internet cloud 1708.

Figure 29B is a schematic block diagram of the connection of an Internet-enabled audio unit 1702 with an Internet device through the Internet cloud, with an audio unit 1702 directly connected to the Internet cloud 1708. In this case the audio unit 1702 is capable of directly connecting to the Internet cloud 1708, and thence to the Internet device 1706, through a wired connection. This could be through a high speed connection (such as a twisted wire Ethernet connection) or through a lower speed connection (e.g. a serial port connection, or a dial-up modem).

The connection of the unit 1700 or unit 1702 is illustrated in Figure 30, tables of ratings of audio unit 100 users. As described above, members of a cluster can decide whether or not to admit a new member to the cluster using a variety of automatic or manual methods. One method of determining the suitability of a user to become a member of the cluster 700 is to determine the user's ratings by members of other clusters to which the user has previously been a member. In this case, the Internet device 1706 is a computer hosting a database, which can be queried and to which information can be supplied by the unit A (either 1700 or 1702). On the Internet device 1706 are stored ratings of units 100, as indicated by the table 1802. The left hand column is the primary key of the database, and is a unique identifier associated with each unit 100. This ID can be a numerical MAC ID, associated with the hardware and software of each unit 100, a unique nickname or word handle (e.g. "Jen412smash") associated with each audio unit user, or other such unique identifier.

The second and third columns, indicated as numbers with dollar signs, are the total summed positive ratings (column two) and the negative ratings (column three) registered with each user by another member of a cluster 700 with which the user has been associated, and in which the user was operating the broadcast unit 710. This rating can, for example, reflect the perceived quality of music provided by the user. The fourth and fifth columns are the total, summed ratings of the user by other members of clusters 700 with which the user has been associated, in which the user was the operator of a receive unit 730. This rating can, for example, indicate the good spirits, friendliness, dress or other characteristics of the user as perceived by other members of the cluster. The sixth column indicates the largest cluster 700 for which the user has been the broadcaster. This is a good indicator of a broadcaster's popularity, since a poor or unpopular broadcaster would not be able to attract a large group of members for a cluster.

There are many other characteristics that can be stored in such a database, and can also include IDs of other members of groups with which the user has been associated (so that members can accept new members who have been associated with friends of those in the

cluster), specific music that the user has played (in order to determine musical compatibility), information on the individuals making each rating (in order to determine rating reliability), and gradations of ratings (rather than simply a positive or negative response).

5 The cluster members can access the ratings of the user requesting membership in the cluster 700 in order to determine their desirability and suitability. This would require a connection with the Internet device 1706 at the time that the user was requesting to join, and would preferably involve a wireless connection through an access point, as in Figure 29A. The information from the database on the device 1706 can either be displayed to the members of the cluster 700, or can be used by an automatic algorithm to determine whether the person
10 can join.

The table 1800 represents the ratings of a cluster 700 of 5 total members (comprising a broadcaster with ID 12089AD, and four additional members with IDs E1239AC, F105AA3, B1B25C0, and ED5491B). The ratings are supplied by ED5491B (whose ID is preceded by a zero), and then specific ratings of each member are made. The DJ is indicated by a dollar
15 sign preceding his ID. These ratings can be made by putting the nicknames/handles of the cluster members on a screen, and allowing the member to indicate positive or negative ratings by pressing one of two buttons. A plus in the first column indicates a positive response, and a minus sign indicates a negative response. These ratings can then be sent during either wired communications directly to the Internet device 1706 or via the access point 1704. It should
20 be noted that the ratings, once made, can be stored on the unit 1700 or 1702 indefinitely, until connection with the Internet cloud 1708 can be made. As indicated by the arrow, the information for B1B25C0 can be added to the table 1802 – in this case, by incrementing the value in the fourth column (a positive rating for a user who is not the broadcaster).

Other applications of connections to Internet devices 1706 include exchanging (via
25 uploading and downloading) dance files with distant individuals, and obtaining music via downloading, which can include transactions with distributors similar to that seen in Figure 25. Such connections also allow the integration of other connectivity, such as telephone and messaging capabilities, expanding the usefulness and attractiveness of audio units 100.

Many Embodiments Within the Spirit of the Present Invention

30 It should be apparent to one skilled in the art that the above-mentioned embodiments are merely illustrations of a few of the many possible specific embodiments of the present invention. For example, the elements of a unit 100, including the inter-unit transmitter/receiver 110 protocol and hardware, the DJ transmitter 120 and the audio player

130 can be chosen from a range of available technologies, and can be combined with user interface elements (keyboards, keypads, touch screens, and cursor buttons, without significantly affecting the operation of the unit 100. Furthermore, many different transducers can be combined into DJs 200, which can further comprise many decorative and functional pieces (e.g. belt clasps, functional watches, microphones, or wedding rings) within the spirit of the present invention. Indeed, the unit 100, itself, can comprise transducers 240, 250 or 260.

It should also be appreciated that communications protocols provide a nearly uncountable number of arrangements of communications links between units in a cluster, that the links can be of mixed software protocols (e.g. comprising both TCP and UDP protocols, and even non-IP protocols) over a variety of hardware formats, including DECT, Bluetooth, 802.11 a, b, and g, Ultra-Wideband, 3G/GPRS, and i-Beans, and that communications can include not only digital but also analog communications modes. Furthermore, communications between audio units and digital jewelry can further comprise analog and digital communications, and a variety of protocols (both customized as well as well-established IP protocols).

It is important, as well, to note that the inter-unit communication and the unit-to-DJ communication can operate and provide significant benefits independently of one another. For example, members listening to music together gain the benefits of music sharing, even without the use of DJs 200. Alternatively, an individual's appreciation of music and personal expression can be augmented through use of a DJ 200, even in the absence of music sharing. However, the combination of music sharing along with enhanced personal expression through a DJ 200 provides a synergistic benefit to all members sharing the music.

Numerous and varied other arrangements can be readily devised by those skilled in the art without departing from the spirit and scope of the invention. Moreover, all statements herein reciting principles, aspects and embodiments of the present invention, as well as specific examples thereof, are intended to encompass both structural and functional equivalents thereof. Additionally, it is intended that such equivalents include both currently known equivalents as well as equivalents developed in the future, i.e. any elements developed that perform the same function, regardless of structure.

In the specification hereof any element expressed as a means for performing a specified function is intended to encompass any way of performing that function. The invention as defined by such specification resides in the fact that the functionalities provided by the various recited means are combined and brought together in the manner which the

specification calls for. Applicant thus regards any means which can provide those functionalities as equivalent as those shown herein.

Claims

What is claimed is:

1. A method for sharing music from stored musical signals between a first user with a first music player device and at least one second user with at least one second music player device, comprising:
 - playing the musical signals for the first user on the first music player device while essentially simultaneously wirelessly transmitting the musical signals from the first music player device to the at least one second music player device; and
 - receiving the musical signals by the at least one second player device, wherein the musical signals can be played on the at least one second player device essentially simultaneously with the playing of the musical signals on the first music player device;
 - wherein the first and the at least one second users are mobile and maintain less than a predetermined distance.
2. The method of claim 1 wherein the predetermined distance is 40 feet.
3. The method of claim 1 wherein the predetermined distance is 100 feet.
4. The method of claim 1 wherein the predetermined distance is 500 feet.
5. The method of claim 1 wherein the transmitting and receiving involve the use of an Internet protocol.
6. The method of claim 5 wherein the Internet protocol uses a connection-less transport layer.
7. The method of claim 5 wherein the Internet protocol uses a link layer involving a protocol chosen from the group consisting of 802.11, Bluetooth, ultrawideband, HiperLAN and WDCT.
8. The method of claim 1 wherein the transmitting and receiving use analog radio frequency transmission.
9. The method of claim 1 wherein the transmitted music signals are digital.
10. The method of claim 9 wherein the transmitted digital music signals are compressed.
11. The method of claim 1 wherein the first music player device is carried by the first user.
12. The method of claim 1, additionally comprising displaying by the first user a humanly-perceptible light display that is roughly synchronized with the beat of the musical signals.

13. The method of claim 1, additionally comprising communicating an identifier of the at least one second user to the first music player device, and indicating the identifier to the first user on a display located on the first music player device.

14. The method of claim 1, additionally comprising transferring a substantially
5 unique music identifier of the musical signal wherein the personal identifier is stored on the at least one second music player device in association with the musical identifier.

15. The method of claim 14 wherein the user of the at least one second music player device retrieves the musical signal from a remote system by using the music identifier.

16. The method of claim 15 wherein the remote system is located on the Internet.

10 17. The method of claim 15, additionally comprising a substantially unique personal identifier of the first user wherein the personal identifier is transferred to the at least one second music player device and stored in association with the music identifier of the music transmitted by the first music player device to the at least one second music player device.

15 18. The method of claim 1, additionally comprising monitoring the strength of the musical signal received by the at least one second music player device and communicating with the at least one second user the strength of the musical signal.

19. A system of music sharing for a plurality of users, comprising:

20 a first sharing device and at least one second sharing device, each comprising a musical signal store, a musical signal transmitter, a musical signal receiver, and a musical signal player;

a broadcast user operating the first sharing device; and

25 at least one member user operating the at least one second sharing device, wherein the broadcast user plays the musical signal for his own enjoyment on the first sharing device and simultaneously transmits the musical signal to the receiver of the at least one second sharing device of the at least one member user, on which the musical signal is played for the at least one member user, wherein the broadcast user and the at least one member user hear the musical signal substantially simultaneously.

30 20. The system of claim 19, additionally comprising a manual switch on at least one second sharing device, wherein actuation of the switch causes the first sharing device to cease transmitting the musical signal and for the second sharing device to begin transmitting musical signal from the musical signal store of the second sharing device, wherein the broadcast user and the at least one member user hear the musical signal from the second sharing device substantially simultaneously.

21. The system of claim 19 wherein the first sharing device and the at least one second sharing device additionally comprise a substantially unique identifier, wherein the first sharing device communicates its substantially unique identifier to the at least one second sharing device and the at least one second sharing device communicates its substantially
5 unique identifier to the first sharing device.

22. The system of claim 21, wherein the identifier includes a least one piece of information about the user of the sharing device taken from the group consisting of the name of the user, the nickname of the user, a numerical identifier, an alphanumerical identifier, the musical preferences of the user, the age of the user, and the educational institution that the
10 user attends.

23. The system of claim 22 wherein the first sharing device and the at least one second sharing device additionally comprise a visual display on which the substantially unique identifiers which have been communicated to the respective device are displayed.

24. The system of claim 19 additionally comprising at least one requesting
15 member user operating an at least one requesting sharing device, wherein the at least one requesting member user requests to receive transmission of the musical signal from the first sharing device.

25. The system of claim 24 wherein the at least one requesting sharing device is associated with a substantially unique identifier which is communicated by the requesting
20 sharing device to the first sharing device and wherein the first sharing device additionally comprises a display on which the identifier is displayed.

26. The system of claim 25 wherein the broadcast user elects whether to transmit the musical signal from the first sharing device to the requesting sharing device.

27. The system of claim 25 wherein the broadcast user and the at least one
25 member user elect through voting whether to transmit the musical signal from the first sharing device, wherein a majority vote is determinative.

28. The system of claim 25 wherein the broadcast user and the at least one member user elect through voting whether to transmit the musical signal from the first sharing device, wherein a unanimous vote is required for transmitting.

30 29. The system of claim 25 wherein the identifier of the requesting sharing device is compared with identifiers stored in the first device, such that the match of the identifier of the requesting sharing device with a stored identifier allows the first sharing device to transmit the musical signal to the requesting sharing device.

30. The system of claim 29 wherein the stored identifiers are retrieved from the Internet.

31. The system of claim 19 wherein the first sharing device additionally comprises a visual display.

5 32. The system of claim 31 wherein the visual display indicates the number of member users receiving the musical signal.

33. The system of claim 31 wherein the visual display can display whether the musical being played by the sharing device comes from the musical signal store of the same device, or from the musical signal store another sharing device.

10 34. The system of claim 19 wherein the first sharing device is mobile.

35. The system of claim 19 wherein the first sharing device is carried by the user.

36. The system of claim 19 wherein the first sharing device and at least one sharing device maintain less than a predetermined distance.

37. The system of claim 36 wherein the predetermined distance is 40 feet.

15 38. The system of claim 36 wherein the predetermined distance is 100 feet.

39. The system of claim 36 wherein the predetermined distance is 500 feet.

40. The system of claim 19, additionally comprising a substantially unique user identifier associated with the broadcast user and a substantially unique musical signal identifier associated with each musical signal, wherein the user identifier and the musical signal identifier are transmitted from the first sharing device to the at least one second sharing device and stored in association with one another on the at least one second sharing device.

41. A wireless communications system for sharing audio entertainment between a first mobile device and a second mobile device in the presence of a non-participating third mobile device, comprising:

25 an announcement signal transmitted by the first mobile device for which the second mobile device and the third mobile device are receptive;

a response signal transmitted by the second mobile device in response to the announcement signal for which the first mobile device is receptive and for which the third mobile device is not receptive;

30 an identifier signal transmitted by the first mobile device to the second mobile device in response to the response signal, and which is not receptive to the third mobile device; and

a broadcast signal comprising audio entertainment that is transmitted by the first mobile device, and which is receptive by the second mobile device on the basis of the reception of the identifier signal.

42. The system of claim 41 wherein the announcement signal is transmitted via an Internet protocol.

43. The system of claim 42 wherein the Internet protocol is a connection-oriented transport protocol.

5 44. The system of claim 43 wherein the connection-oriented protocol is TCP.

45. The system of claim 42 wherein the Internet protocol further comprises a link layer protocol selected from the group consisting of 802.11, Bluetooth, ultrawideband, HiperLAN and WDCT.

10 46. The system of claim 41 wherein the broadcast signal is transmitted via an Internet protocol.

47. The system of claim 46 wherein the Internet protocol is a connectionless transport protocol.

48. The system of claim 47 wherein the connectionless transport protocol is UDP.

15 49. The system of claim 46 wherein the Internet protocol further comprises a link layer protocol selected from the group consisting of 802.11, Bluetooth, ultrawideband, HiperLAN and WDCT.

50. The system of claim 41 wherein the broadcast signal is an analog radio frequency signal.

20 51. The system of claim 41 wherein the broadcast signal is transmitted via telephony.

52. The system of claim 41 wherein the broadcast signal is compressed.

53. The system of claim 52 wherein the compression is MP3.

54. The system of claim 41 wherein the broadcast signal is encrypted.

55. The system of claim 41 wherein the identifier signal is an IP socket address.

25 56. The system of claim 41 wherein the identifier signal is a decryption code.

57. The system of claim 41 wherein the identifier signal is a digital password.

58. The system of claim 41 wherein the announcement signal and the broadcast signal use different IP transport protocols.

30 59. The system of claim 41 additionally comprising a transfer signal that is transmitted by the first mobile device upon halting the transmission of its broadcast signal, and which transfer signal is receptive by the second mobile device, such that reception by the second mobile device causes the device to transmit a second broadcast signal comprising audio entertainment for which the first unit is receptive.

60 The system of claim 41 wherein the broadcast signal is transmitted via peer-to-peer communications.

61 The system of claim 60, additionally comprising a switch, wherein actuation of the switch causes the device to toggle between peer-to-peer communications and
5 infrastructure-based communications.

62 The system of claim 41 wherein the broadcast signal is transmitted via infrastructure-based communications.

63. An audio entertainment device, comprising:
a signal store that stores an audio entertainment signal;
10 a transmitter that can transmit the stored audio entertainment signal;
a receiver that can receive the transmitted audio entertainment signal from a transmitter of another such device; and
a player that can play audio entertainment from a member selected from the group of stored audio entertainment signals or audio entertainment signals transmitted from the
15 transmitter of another such device.

64. The device of claim 63 wherein the transmitter transmits to the receiver of the same device in order to play through the player of that device.

65. The device of claim 63 wherein the player can play audio entertainment from signals obtained from either the receiver or from the audio store.

20 66. The device of claim 63 additionally comprising a visual display that is perceptive to the user of the device, indicating information about the audio entertainment signal.

67. The device of claim 66 wherein a third device can receive the transmitted audio signal, wherein all such devices transmitting and receiving the same audio
25 entertainment signal constitute a sharing group.

68. The device of claim 67 additionally comprising a substantially unique identifier that identifies the user of the device, wherein at least one device of the sharing group displays the identifier of the user of another device within the sharing group.

69. The device of claim 68 wherein the identifier includes a least one piece of
30 information about the user taken from the group consisting of the name of the user, the nickname of the user, the musical preferences of the user, a numerical identifier, an alphanumerical identifier, the age of the user, or the educational institution that the user attends.

70. The device of claim 68 wherein the identifier is interspersed within the transmitted audio entertainment signal.

71. The device of claim 66 wherein the visual display can display the number of devices in the sharing group.

5 72. The device of claim 66 wherein the visual display can display whether the audio entertainment being played by a device comes from the audio store of the device of which the display is a part, or from another device.

73. The device of claim 63 wherein the second device provides information to the second user about the strength of the audio signal received by the receiver.

10 74. The device of claim 73 additionally comprising a visual display, wherein the signal strength information is provided in the visual display

75. The device of claim 73 wherein the signal strength information is provided secondary audio signals played substantially simultaneously with the audio entertainment signal through the player of the device.

15 76. The device of claim 63 wherein the signal store stores a compressed audio signal.

77. The device of claim 63 wherein the compressed audio signal is transmitted through the transmitter.

20 78. The device of claim 63 wherein the compressed audio signal is decompressed prior to being transmitted through the transmitter.

79. The device of claim 63 wherein the identifier is interspersed within the transmitted audio signal.

80. The device of claim 63 additionally comprising a delay circuit.

25 81. The device of claim 80 wherein the playing of the audio entertainment signal is delayed in each device so that the audio entertainment signal in all devices playing from the same stored audio entertainment signal is played substantially simultaneously.

30 82. The device of claim 63 additionally comprising a manual control with at least three states, wherein in the first state the player plays a stored audio entertainment signal and does not transmit the stored entertainment signal, wherein in the second state the player plays a stored audio entertainment signal and substantially simultaneously transmits the same audio entertainment signal, and wherein in the third state the player plays an audio entertainment signal received by the receiver.

83. The device of claim 63 wherein the ambient sound admitted by the earphone can be manually controlled.

84. The device of claim 63 additionally comprising a microphone wherein the device can transmit vocal sounds from the user of the device.

85. The device of claim 63 additionally comprising a manual control such that actuation transmits a message.

5 86. The device of claim 63 wherein the device can connect with the telephone network.

87. The device of claim 63 wherein the device is mobile and can be carried by a user.

88. The device of claim 63 additionally comprising a control signal transmitter
10 that transmits control signals that controls a humanly-perceptible display in a wearable accessory.

89. The device of claim 88 wherein the control signal transmitter is the same as the transmitter that transmits stored audio entertainment signals.

90. The device of claim 88, additionally comprising a store of control signals.

15 91. The device of claim 88, additionally comprising a generator wherein the generator converts stored audio entertainment signals into control signals.

92. An audio entertainment device comprising:

a wireless transmitter for the transmission of audio entertainment signals;

20 a wireless receiver for the reception of the transmitted audio entertainment signals from a transmitter of audio entertainment signals;

a first manually-separable connector for electrically connecting with an audio player, wherein the connector allows transfer of audio entertainment signals from the player to the device;

a second connector for connecting with a speaker; and

25 a control to manually switch between at least three states, wherein in the first state the speaker plays audio entertainment signals from the audio player and the transmitter does not transmit the audio entertainment signals, wherein in the second state the speaker plays audio entertainment signals from the audio player and the transmitter essentially simultaneously transmits the audio entertainment signals, and wherein in the third state the speaker plays
30 audio entertainment signals received by the receiver.

93. The device of claim 92 wherein the first connector connects to the audio player through the audio output jack.

94. The device of claim 92 wherein the speaker is chosen from the group consisting of earphone and headphone.

95. The device of claim 92 wherein the device additionally comprises a control signal generator that converts audio entertainment signals into control signals that control a humanly-perceptible display in a wearable accessory.

96. The device of claim 92 additionally comprising a second transmitter that
5 transmits control signals that control a humanly-perceptible display in a wearable accessory.

97. The device of claim 92 additionally comprising a display that indicates the state of the device.

98. The device of claim 92 wherein the receiver is receptive of the audio entertainment signals transmitted by the transmitter.

10 99. The device of claim 92 wherein the receiver is operational when the first connector is manually separated from the audio player.

100. A system for the sharing of stored music between a first user and a second user, comprising:

a first device for playing music to the first user, comprising a store of musical signals,
15 a first controller that prepares musical signals from the first store for transmission and playing, a first player that takes musical signals from the first controller and plays the signals for the first user, and a transmitter, that is capable of taking the musical signals from the controller and transmitting the musical signals via wireless broadcast; and

a second device for playing music to the second user, comprising a receiver receptive
20 of the transmissions from the transmitter of the first device, a second controller that prepares musical signals from the receiver for playing, and a second player that takes musical signal from the second controller and plays the signals for the second user;

wherein the first user and the second user hear the musical signals at substantially the same time.

25 101. The system of claim 100 wherein the store of musical signals comprises compressed digital audio files.

102. The system of claim 100 wherein the compressed digital audio files are in formats taken from the group consisting of mp3, wav, cda, wma, and au.

103. The system of claim 100 wherein the preparation of the musical signals by the
30 first controller comprises decompression of compressed binary files.

104. The system of claim 100 wherein the transmitter transmits via an Internet protocol.

105. The system of claim 104 wherein the Internet protocol has a link level selected from the group consisting of 802.11, Bluetooth, ultrawideband, HiperLAN and WDCT.

106. The system of claim 100 wherein the first device is a telephone.

107. The system of claim 100 wherein the first device transmits compressed musical signals and the second controller decompresses the compressed musical signals received by the receiver.

5 108. The system of claim 100 wherein the transmitter additionally transmits an annunciator signal indicating the presence of a transmitter of musical signals.

109. The system of claim 100 wherein the second device receiver becomes receptive of transmissions from the first device by obtaining an identifier from the first device.

10 110. The system of claim 109 wherein the first device additionally comprises an associated visual transducer and the second device additionally comprises a visual receiver, wherein the visual transducer outputs a visual identifier signal that is received by the visual receiver.

111. The system of claim 109 wherein the first device additionally comprises a first
15 electrical connector and the second device additionally comprises a second electrical connector, wherein the identifier is communicated via an electrical signal passed from the first electrical connector to the second electrical connector.

112. The system of claim 111 wherein the communication of the identifier signal occurs via an inductive coupling between the first connector and the second connector.

20 113. The system of claim 109 wherein the second device additionally comprises a microphone, wherein the identifier is communicated in an audio signal that is produced by the first player and which is received by the microphone.

114. The system of claim 109 wherein the identifier is transmitted by the transmitter and received by the receiver.

25 115. The system of claim 100 wherein the second device receiver becomes receptive of transmissions from the first device by transmitting an identifier to the first device.

30 116. The system of claim 115 wherein the second device additionally comprises a visual transducer and the first device additionally comprises a visual receiver, wherein the visual transducer outputs a visual identifier signal that is received by the visual receiver.

117. The system of claim 115 wherein the first device additionally comprises a first electrical connector and the second device additionally comprises a second electrical connector, wherein the identifier is communicated via an electrical signal passed from the second electrical connector to the first electrical connector.

118. The system of claim 117 wherein the communication of the identifier signal occurs via an inductive coupling between the first connector and the second connector.

119. The system of claim 115 wherein the first device additionally comprises a microphone, wherein the identifier is communicated in an audio signal that is produced by the second player and which is received by the microphone.

120. The system of claim 115 wherein the identifier is transmitted by the transmitter and received by the receiver.

121. The system of claim 100 wherein the first controller additionally generates control signals related to the musical signals.

122. The system of claim 121 wherein the store additionally comprises stored control signals that are used by the first controller.

123. The system of claim 121 wherein the controller generates control signals from the musical signals.

124. The system of claim 123 wherein the controller performs frequency analysis of the musical signals in order to generate the control signals.

125. The system of claim 121 wherein the first device additionally transmits the control signals to the second device using the first transmitter.

126. The system of claim 125 wherein the control signals and the musical signals transmitted by the first device are interspersed within a single digital stream.

127. The system of claim 125 wherein the control signals and the musical signals are transmitted in two digital streams.

128. The system of claim 121 wherein the first device additionally comprises a second transmitter that transmits control signals.

129. The system of claim 128 wherein the second transmitter uses an IP protocol.

130. The system of claim 129 wherein the Internet protocol uses a link layer selected from the group consisting of 802.11, Bluetooth, and ultrawideband.

131. The system of claim 121 wherein the second transmitter uses analog radio frequency transmissions.

132. The system of claim 100, additionally comprising a wearable accessory, comprising a control signal receiver that is receptive of control signals transmitted by the first device, an accessory controller that generates transducer signals from the received control signals, and a transducer that generates humanly-perceptive music accompaniment signals from the transducer signals.

133. The system of claim 132 wherein the music accompaniment signals are visual signals.

134. The system of claim 133 wherein the visual signals are generated by a light emitting diode

5 135. The system of claim 132 wherein the music accompaniment signals are tactile signals.

136. The system of claim 132 wherein the accessory controller performs frequency analysis on the received control signals.

137. The system of claim 132 wherein the wearable accessory additionally
10 comprises a switch, wherein actuation of the switch causes the accessory to be receptive to an entraining signal from the first device, such that the accessory will accept signals from the first device.

138. An earphone for listening to audio entertainment allowing for the controlled reception of ambient sound by a user, comprising

15 a speaker that is oriented towards the user's ear;
an enclosure that reduces the amount of ambient noise perceptible to the user; and
a manually-adjustable characteristic of the enclosure that adjusts the amount of ambient sound perceptible to the user.

139. The earphone of claim 138 wherein the manually-adjustable characteristic is
20 the size of apertures in the enclosure that admit ambient sound.

140. The earphone of claim 138 wherein the manually-adjustable characteristic is the number of apertures in the enclosure that admit ambient sound.

141. The earphone of claim 138 wherein the manually-adjustable characteristic is the surface area of the enclosure.

25 142. The earphone of claim 138 wherein the manually-adjustable characteristic is the thickness of the enclosure.

143. The earphone of claim 138, additionally comprising a spacer that attaches to the earphone and which increases the distance between the speaker and the user's ear.

144. The earphone of claim 138, additionally comprising a secondary enclosure
30 that encloses the enclosure and the speaker, and which is detachable from the earphone.

145. A mobile device for the transmission of audio entertainment signals comprising

an audio signal store for the storage of the audio entertainment signals;
an audio signal player for the playing of the audio entertainment signals;

an wireless transmitter for the transmission of the audio entertainment signals; and
a transmitter control to manually switch between two states consisting of the
operation and the non-operation of the audio transmitter.

146. The device of claim 145 wherein the transmitter is connected to the audio jack
5 of the audio signal player.

147. The device of claim 146 wherein the audio signal that is transmitted via the
audio jack is subsequently divided between the transmitter and the headphone.

148. The device of claim 146 wherein the transmitter additionally comprises a
second jack to which the earphone plug can connect, such that the insertion of the earphone
10 plug into the second jack allows the audio signal to be heard in the earphone

149. A mobile device for the reception of digital audio entertainment signals
comprising:

an audio signal store for the storage of the digital audio entertainment signals;

an audio receiver for the reception of external digital audio entertainment signals from
15 a mobile audio signal transmitter located within a predetermined distance of the audio
receiver;

a receiver control with at least a first state and a second state; and

an audio signal player that plays digital audio entertainment signals from the audio
signal store when the receiver control is in the first state, and that plays digital audio
20 entertainment signals from the audio receiver when the receiver control is in the second state.

150. The device of claim 149, additionally comprising a jack to which a listening
device chosen from the group consisting of earphone and headphone can be connected.

151. A method for the shared enjoyment of music from stored musical signals
between a first user with a first music player device and at least one second user with at least
25 one second music player device, comprising:

playing the musical signals for the first user on the first music player device while
essentially simultaneously wirelessly transmitting synchronization signals from the first
music player device to the at least one second music player device; and

receiving the synchronization signals by the at least one second player device,
30 wherein the synchronization signals allow the musical signals on the at least one second
player device to be played essentially simultaneously with the playing of the musical signals
on the first music player device;

wherein the first and the at least one second users are mobile.

152. The method of claim 151 wherein the same musical signals are stored on the first music player device and the at least one second music player device.

153. The method of claim 152 wherein the musical signals are stored as compressed audio files.

5 154. The method of claim 151 wherein the synchronization signals comprise the current sample number being played on the first music player device.

155. The method of claim 154 wherein the synchronization signals further comprise the sample rate of the musical signal being played on the first music player device.

10 156. The method of claim 151 wherein the synchronization signals comprise the current time relative to the beginning of the musical signal being played on the first music player device.

157. The method of claim 151, additionally comprising transferring the musical signals from the first music player device to the at least one second music player device.

15 158. A wireless communications system for sharing audio entertainment between a first mobile device and a second mobile device, comprising:

a broadcast identifier transmitted by the first mobile device to the second mobile device;

a personal identifier transmitted by the second mobile device to the first mobile device; and

20 a broadcast signal comprising audio entertainment that is transmitted by the first mobile device of which the second device is receptive;

wherein the first mobile device and the second mobile device have displays which can display the identifier signal that they receive and the second mobile device can play the audio entertainment from the broadcast signal that it receives.

25 159. The system of claim 158 wherein the identifier signals are transmitted via an Internet protocol.

160. The system of claim 159 wherein the Internet protocol is a connection-oriented transport protocol.

30 161. The system of claim 159 wherein the Internet protocol further comprises a link layer protocol selected from the group consisting of 802.11, Bluetooth, ultrawideband, HiperLAN and WDCT.

162. The system of claim 158 wherein the broadcast signal is transmitted via an Internet protocol.

163. The system of claim 162 wherein the Internet protocol is a connectionless transport protocol.

164. The system of claim 162 wherein the Internet protocol further comprises a link layer protocol selected from the group consisting of 802.11, Bluetooth, ultrawideband, 5 HiperLAN and WDCT.

165. The system of claim 158 wherein the broadcast signal is an analog radio frequency signal.

166. The system of claim 158 wherein the broadcast signal is transmitted via telephony.

10 167. The system of claim 158 wherein the broadcast signal is compressed.

168. The system of claim 158 wherein the broadcast signal is encrypted.

169. The system of claim 158, additionally comprising a transfer signal that is transmitted by the first mobile device upon halting the transmission of its broadcast signal, and which transfer signal is receptive by the second mobile device, such that reception by the 15 second mobile device causes the device to transmit a second broadcast signal comprising audio entertainment for which the first unit is receptive.

170. The system of claim 158 wherein the broadcast signal is transmitted via peer-to-peer communications.

171. The system of claim 170 additionally comprising a switch, wherein actuation 20 of the switch causes the device to toggle between peer-to-peer communications and infrastructure-based communications.

172. The system of claim 158 wherein the broadcast signal is transmitted via infrastructure-based communications.

173. The system of claim 158 wherein the personal identifier includes a least one 25 piece of information about the user of the second mobile device taken from the group consisting of the name of the user, the nickname of the user, a numerical identifier, an alphanumerical identifier, the musical preferences of the user, the age of the user, and the educational institution that the user attends.

174. The system of claim 158 additionally comprising a third mobile device whose 30 user requests to receive the broadcast signal and additionally comprising a requester identifier signal transmitted by the third mobile device to the first mobile device.

175. The system of claim 174 wherein the requester identifier signal is additionally transmitted by the third mobile device to the second mobile device.

176. The system of claim 174 wherein the requester identifier signal is additionally transmitted by the first mobile device to the second mobile device.

177. The system of claim 174 wherein the requester identifier is displayed on the display of the first mobile device.

5 178. The system of claim 174 wherein the third mobile device receives the broadcast signal without the affirmative action of the first mobile device.

179. The system of claim 174 wherein the third mobile device must receive the affirmative action of the first mobile device in order to receive the broadcast signal.

10 180. The system of claim 174 wherein the third mobile device must receive the affirmative action of the first mobile device and all other mobile devices receptive of the broadcast signal.

181. The system of claim 180 wherein the affirmative action is a unanimous vote.

182. The system of claim 180 wherein the affirmative action is a majority vote.

15 183. The system of claim 158, additionally comprising a substantially unique musical identifier related to the audio entertainment, wherein the musical identifier is transmitted by the first mobile device to the second mobile device in association with the broadcast signal, and the musical identifier is stored on the second mobile device in association with the broadcast identifier.

20 184. A system for identifying a first device that introduces a music selection to a second device, comprising:

a mobile music transmitter operated by the first device;

a mobile music receiver operated by the second device;

a music signal comprising the music selection transmitted by the transmitter and received by the receiver;

25 an individual musical identifier that is associated with the music selection; and

an individual transmitter identifier that identifies the transmitter, wherein the transmitter identifier and the individual music identifier are stored in association with each other in the receiver.

185. A method for enhancing enjoyment of a musical selection, comprising:

30 obtaining control signals related to the musical selection;

transmitting the control signals wirelessly;

receiving the control signals; and

converting the control signals to a humanly-perceptible form.

186. The method of claim 185 wherein the distance over which the control signals are transmitted has a minimum range of at least 1 foot.

187. The method of claim 185 wherein the distance over which the control signals are transmitted has a maximum range of less than 500 feet.

5 188. The method of claim 185 wherein the humanly-perceptible form is a visually-perceptible form.

189. The method of claim 185 wherein the humanly-perceptible form is a tactilely-perceptible form.

10 190. The method of claim 185 wherein the transmitting and receiving use radio frequency communications.

191. The method of claim 185 wherein the radio frequency communications use an Internet protocol.

15 192. The method of claim 185 wherein the Internet protocol involves a link layer chosen from the group consisting of 802.11, Bluetooth, ultrawideband, HiperLAN and WDCT.

193. The method of claim 185 wherein obtaining additionally comprises extracting the musical beat from the musical selection.

194. The method of claim 185 wherein obtaining additionally comprises retrieving the control signals from a store of control signals.

20 195. The method of claim 185 wherein the control signals are divided into multiple channels.

196. The method of claim 195 wherein the step of converting comprises a step of demultiplexing the control signals.

25 197. The method of claim 185 wherein the steps of transmitting and receiving are performed using means conveyed by a single person.

198. The method of claim 197 wherein the user is mobile.

199. The method of claim 185 wherein the step of transmitting is performed by a stationary user and the step of receiving is performed by a mobile user.

30 200. The method of claim 185 wherein the musical selection is provided by musicians in concert playing essentially simultaneously with obtaining control signals.

201. The method of claim 185 wherein the control signals are transmitted with a substantially unique identifier.

202. The method of claim 185 wherein the control signals control the appearance of a wearable device.

203. A method for generating and storing control signals corresponding to musical signals, comprising:

playing musical signals for a user;

5 receiving manual input signals from the user that are produced substantially in synchrony with the music;

generating control signals from the input signals; and

storing the control signals so that they can be retrieved with the musical signals.

204. The method of claim 203 wherein receiving comprises transducing tactile
10 input from the user.

205. The method of claim 204 wherein the tactile input is produced using a user appendage selected from the group consisting of hand, finger and foot.

206. The method of claim 203 wherein receiving comprises transducing sound output produced by the user.

15 207. The method of claim 203 wherein storing comprises interleaving control signals within the musical signals.

208. The method of claim 203 wherein storing additionally comprises creating a digital file.

209. The method of claim 203 wherein storing additionally comprises associating
20 the digital file with the musical signals.

210. A wearable personal accessory, comprising:

an input transducer taken from the group consisting of a microphone and an accelerometer, said transducer generating a time-varying input transduction signal;

25 a controller which accepts the input transduction signal, and which generates an output transducer signal whose signal varies in amplitude with time;

an output transducer receptive of the output transducer signal, and which provides a humanly-perceptible signal; and

an energy source for powering the input transducer, controller and output transducer.

211. The accessory of claim 210 wherein the accelerometer responds to the
30 movements of the user in one axis of movement.

212. The accessory of claim 210 wherein the accelerometer responds to the movements of the user in more than one axis of movement.

213. The accessory of claim 210 wherein the controller performs automatic gain control on the signal from the input transducer.

214. The accessory of claim 210 wherein the output transducer is a light transducer.

215. The accessory of claim 214 wherein the output transducer comprises a light emitting diode.

216. The accessory of claim 214 wherein the output transducer comprises a liquid
5 crystal display.

217. The accessory of claim 214 wherein the output transducer comprises an organic light emitting diode.

218. The accessory of claim 210 wherein the output transducer is a tactile transducer.

10 219. The accessory of claim 210 wherein the output transducer comprises a first and a second transduction element.

220. The accessory of claim 219 wherein the first transduction element responds preferentially to a first band of frequencies of the input transduction signal and the second transduction element responds preferentially to a second band of frequencies of the input
15 transduction signal.

221. The accessory of claim 219 wherein the first transduction element responds preferentially to a first band of amplitudes of the input transduction signal and the second transduction element responds preferentially to a second band of amplitudes of the input transduction signal.

20 222. A wearable personal accessory controlled via wireless communications, comprising:

a wireless communications receiver, receptive of an external control signal;

a controller that accepts the external control signal and that generates a time-varying visual output transducer signal;

25 a visual output transducer that is receptive of the output transducer signal, and which provides a humanly-perceptible visual signal; and

an energy store for powering the receiver, controller and output transducer;

wherein the visual output transducer generates visually-perceptive output.

30 223. The accessory of claim 222 wherein the external control signal is a time-varying amplitude signal.

224. The accessory of claim 223 wherein the amplitude signal is analog.

225. The accessory of claim 223 wherein the amplitude signal is digital.

226. The accessory of claim 223 wherein the amplitude signal is substantially a musical signal.

227. The accessory of claim 226 wherein the controller performs frequency filtering of the external control signal.

228. The accessory of claim 226 wherein the controller determines the primary beat of the music.

5 229. The accessory of claim 228 wherein the primary beat of the music is determined according to the positions in the music where the amplitude rises at least a predetermined factor over less than a predetermined time.

230. The accessory of claim 222 wherein the external control signal is a digital signal of pairs comprising positions within the music and the amplitudes to which the output
10 transducers are set at such times.

231. The accessory of claim 230 wherein the positions are chosen from the group of characteristics consisting of samples and times.

232. The accessory of claim 222 wherein the receiver is an analog radio frequency receiver.

15 233. The accessory of claim 232 wherein the analog radio frequency communications are at a frequency set by a crystal oscillator, wherein the crystal oscillator can be manually exchanged in order to change the frequency of operation.

234. The accessory of claim 222 wherein the receiver receives Internet protocol signals.

20 235. The accessory of claim 234 wherein the Internet protocol further comprises a link layer selected from the group consisting of Bluetooth, 802.11 and ultrawideband.

236. The accessory of claim 222 wherein the visual output transducer comprises a first light transducer and a second light transducer.

237. The accessory of claim 236 wherein the external control signal comprises a
25 multiplexed first signal and second signal, and wherein the first signal relates to the first light transducer and the second signal relates to a second light transducer.

238. The accessory of claim 237 wherein the controller comprises at least two ports, wherein the first port relates to the first light generating element and the second port related to the second light generating element.

30 239. The accessory of claim 236 wherein the first light transducer responds preferentially to a first range of frequency in the musical signal, and the second light transducer responds preferentially to a second range of frequency in the musical signal.

240. The accessory of claim 236 wherein the first light transducer responds preferentially to a first range of amplitude in the musical signal, and the second light transducer responds preferentially to a second range of amplitude in the musical signal.

241. The accessory of claim 222 wherein the visual output transducer comprises a
5 light emitting diode.

242. The accessory of claim 222 wherein the visual output transducer comprises an organic light emitting diode.

243. The accessory of claim 222 wherein the visual output transducer comprises a liquid crystal display.

10 244. The accessory of claim 222 additionally comprising a tactile transducer responsive to the external control signal.

245. The accessory of claim 222 additionally comprising a sound transducer responsive to the external control signal.

15 246. The accessory of claim 222 wherein the external control signal additionally comprises a digital identifier.

247. The accessory of claim 246 wherein the visual output transducer additionally provides an identifier signal that corresponds to the digital identifier, wherein such identifier signal is higher frequency than is humanly perceptible, and wherein the identifier signal is output substantially simultaneously with the humanly-perceptible visual signal.

20 248. The accessory of claim 222 additionally comprising a manually-separable electrical connection, wherein the connection connects the receiver to the controller.

249. The accessory of claim 222 additionally comprising a manually-separable electrical connection, wherein the connection connects the controller to the output transducer.

25 250. The accessory of claim 222 wherein the accessory functions as an accoutrement selected from the group consisting of a ring, a pendant, a hair clip, an earring, a necklace, a brooch, a belt buckle, watches, and a bracelet.

251. The accessory of claim 222 wherein the accessory is incorporated into an article of clothing selected from the group consisting of a backpack, a wallet, a purse, a hat and a shoe.

30 252. The accessory of claim 222 additionally comprising a manual control whose operation sets either a first state and a second state and wherein in the first state the accessory responds exclusively to external control signals comprising a substantially unique identifier.

253. The accessory of claim 252 wherein the identifier to which the accessory responds in the first state is received by the receiver when the accessory is set to the second state.

254. The accessory of claim 253 additionally comprising a third state which is manually set, wherein in the third state, the accessory responds to external control signals in the absence of a substantially unique identifier.

255. A device for converting user tactile responses to stored music into a stored control signal, comprising:

a player that plays stored music audible to the user;

a manually-operated transducer that outputs an electrical signal, wherein the transducer is actuated by the user in response to the music;

a controller that receives the electrical signal and outputs a control signal; and

a store that receives the control signal and stores it.

256. The device of claim 255 wherein the control signal comprises a list of positions, wherein the positions are chosen from the set of characteristics consisting of samples and time relative to the beginning of the stored music.

257. The device of claim 255 wherein the control signal further comprises a list of amplitudes corresponding to the positions.

258. The device of claim 257 additionally comprising a second manually-operated transducer that outputs a second electrical signal, wherein the controller additionally receives the second electrical signal.

259. The device of claim 255 wherein the transducer comprises a musical drum operated by the user whose audible output is received by a microphone that generates the electrical signal received by the controller.

260. The device of claim 255 wherein the controller performs frequency filtering on the electrical signal.

261. The device of claim 255 wherein the transducer is a tactile transducer that is operated by an appendage of the user selected from the group consisting of finger, hand, and foot.

262. The device of claim 255 wherein the store is a file system and the control signal is stored as control file and the stored music is stored as a music file, wherein the control file and the music file are stored in association with each other.

263. The device of claim 255 wherein the control signals are stored in an interleaved fashion with the stored music.

264. The device of claim 71 additionally comprising a visual display on which a visual representation of the control signals is displayed.

265. A music player that wirelessly transmits control signals related to the music, wherein the control signals control a wearable electronic accessory, comprising:

- 5 a store of music signal files;
- a controller that reads a musical signal file from the store and generates audio signals and that further generates the control signals;
- a transducer that converts the audio signals into sound audible to the user; and
- a wireless transmitter that transmits the control signal to the wearable electronic

10 accessory.

266. The player of claim 265 wherein the music signal files comprise compressed audio files.

267. The player of claim 265 wherein the controller performs frequency filtering of the external control signal.

15 268. The player of claim 265 wherein the controller determines the primary beat of the music.

269. The player of claim 268 wherein the primary beat of the music is determined according to the positions in the music where the amplitude rises at least a predetermined factor over less than a predetermined time.

20 270. The player of claim 265 wherein the control signals are pairs comprising positions within the music and the amplitudes to which the output transducers are set at such times.

271. The player of claim 265 wherein the positions are chosen from the set of characteristics consisting of samples and times.

25 272. The player of claim 265 wherein the transmitter is an analog radio frequency transmitter.

273. The player of claim 265 wherein the transmitter transmits according to an Internet protocol.

30 274. The player of claim 273 wherein the Internet protocol has a link layer selected from the group consisting of 802.11, Bluetooth, ultrawideband, HiperLAN and WDCT.

275. The player of claim 265 additionally comprising a visual transducer that generates visually-preceptive output in response to the control signal, wherein the visual transducer receives the control signal from the controller.

276. The player of claim 275 wherein the visual transducer is a light emitting diode.

277. The player of claim 275 wherein the visual transducer is a liquid crystal display.

278. The player of claim 265 additionally comprising a store of configuration files wherein the configuration files include parameters read by the controller that affect the generation of control signals from the musical signals.

279. A music player that wirelessly transmits control signals related to the music, wherein the control signals control a wearable electronic accessory, comprising:

a store of music signal files;

a second store of control signal files associated with the music signal files;

a controller that reads a musical signal file from the store and generates audio signals and that further reads an associated control signal file;

a transducer that converts the audio signals into sound audible to the user, and

a wireless transmitter that transmits the control signals from the associated control signal file to the wearable electronic accessory.

280. The player of claim 279 wherein the music signal files comprise compressed audio files.

281. The player of claim 279 wherein the transmitter is an analog radio frequency transmitter.

282. The player of claim 279 wherein the transmitter transmits according to an Internet protocol.

283. The player of claim 282 wherein the Internet protocol has a link layer selected from the group consisting of 802.11, Bluetooth, ultrawideband, HiperLAN and WDCT.

284. The player of claim 279 wherein the store of control information comprises at least one binary data file that relates to a single music signal file.

285. The player of claim 279 wherein the store of control information is interleaved with a related music signal file in binary data format.

286. The player of claim 279 additionally comprising a visual transducer that generates visually-preceptive output in response to the electrical control signal, wherein the visual transducer receives the electrical control signal from the controller.

287. A system for exhibition of music enjoyment, comprising:

a source of music signals;

a controller that generates control signals from the music signals;

a transmitter of the control signals, wherein the transmission of the control signals is synchronized with the playing of the music signals;

a receiver of the control signals; and
a transducer that responds to the control signals.

288. The system of claim 287 wherein the source of music signals are stored binary music files.

5 289. The system of claim 288 wherein the stored binary data files are compressed audio files.

290. The system of claim 288 wherein the music signals are stored on a medium selected from the group consisting of compact disk, digital video disk, hard disk drive, removable disks, flash memory, magneto-optical disk, and audio tape.

10 291. The system of claim 287 wherein the source of music signals are musicians playing live music.

292. The system of claim 287 wherein the receiver and transducer are worn by a user.

15 293. The system of claim 287 wherein the receiver and the transducer are physically separated from the transmitter by more than 5 feet.

294. The system of claim 287 wherein the transducer is a visual transducer that generates humanly-perceptible visual signals.

295. The system of claim 294 wherein the visual transducer is a light emitting diode

20 296. The system of claim 294 wherein the visual transducer is a liquid crystal display

297. The system of claim 287 wherein the transducer is a tactile transducer.

298. The system of claim 287 wherein the minimum range of the transmitter for which the receiver is receptive is at least 1 foot.

25 299. The system of claim 287 wherein the maximum range of the transmitter for which the receiver is receptive is less than 500 feet.

300. The system of claim 287 wherein the controller determines the primary beat of the music.

301. The system of claim 287 wherein the controller performs frequency filtering.

30 302. The system of claim 287, additionally comprising a second controller that is electrically connected to the receiver, and which generates transducer control signals from the control signals received by the receiver, wherein the transducer control signals directly control the transducer.

303. The system of claim 302 wherein the control signals are time-varying amplitude signals and wherein the second controller performs frequency filtering.

304. The system of claim 302 additionally comprising a switch that is electrically connected to the second controller such that actuation of the switch places the second controller into entraining mode, wherein the receiver becomes receptive of entraining signals from the transmitter.

305. The system of claim 304 wherein a receiver that is entrained will receive control signals other than from the transmitter in the absence of signals from the transmitter.

306. The system of claim 304 wherein additional actuation of the switch will allow reception of control signals by the receiver from a transmitter to which it is not entrained.

307. The system of claim 287 wherein the transmitter and receiver communicate via an Internet protocol.

308. The system of claim 287 wherein the transmitter and receiver communicate via digital radio frequency communications.

309. The system of claim 287 wherein the transmitter and receiver communicate via analog radio frequency communications.

310. The system of claim 309 wherein the analog radio frequency communications are at a frequency set by a crystal oscillator and wherein the crystal oscillator can be manually replaced.

311. The system of claim 287 additionally comprising a transferror for the transmission of music signals.

312. The system of claim 311 wherein the transmitter and the transferror are the same.

313. The system of claim 311 wherein the transmitter and the transferror are different, and communicate via different communications protocols.

314. The system of claim 287 additionally comprising a second receiver for the reception of music signals, wherein the music signals that are received are played by the music player.

315. A method for transferring a wearable-accessory control file stored on a first device to a second device in which an associated music file is stored, comprising:

storing on the first device the name of the music file in conjunction with the control file with which it is associated;

requesting by the second device of the first device for a control file stored in conjunction with the name of the music file; and

transferring the control file from the first device to the second device, wherein the control file is stored on the second device in conjunction with the name of the associated music file.

5 316. The method of claim 315 additionally comprising the step of extracting, wherein the control file is stored on the first device interspersed with the music file as a combined file, and the control file is extracted from the music file prior to being transferred to the second device.

317. The method of claim 315 wherein the requesting is performed over a wide area network.

10 318. The method of claim 315 wherein the name of the control file comprises the root of the name of the associated music file appended with a predetermined extension that is different from that of the music file.

319. The method of claim 315 wherein the control file and the associated music file both comprise the same substantially unique identifier.

15 320. A device for transmitting control signals to a wearable accessory receptive of such control signals, comprising:

a manually-separable input connector for connecting to an output port of an audio player, wherein audio signals are conveyed from the audio player to the device across the connector;

20 a controller for generating control signals from the audio signals; and
a transmitter for transmitting the control signals.

321. The device of claim 320 additionally comprising a second connector, wherein a portion of the audio signal conveyed to the device is conveyed to the second connector, and wherein an earphone can connect to the second connector.

25 322. The player of claim 320 wherein the controller performs frequency filtering of the audio signal.

323. The player of claim 320 wherein the controller determines the primary beat of the audio signal.

30 324. The player of claim 320 additionally comprising a visual transducer that generates visually-preceptive output in response to the control signals, wherein the visual transducer receives the control signals from the controller.

325. The system of claim 320 wherein the transmitter transmits via an Internet protocol.

326. The system of claim 320 wherein the transmitter transmits via digital radio frequency communications.

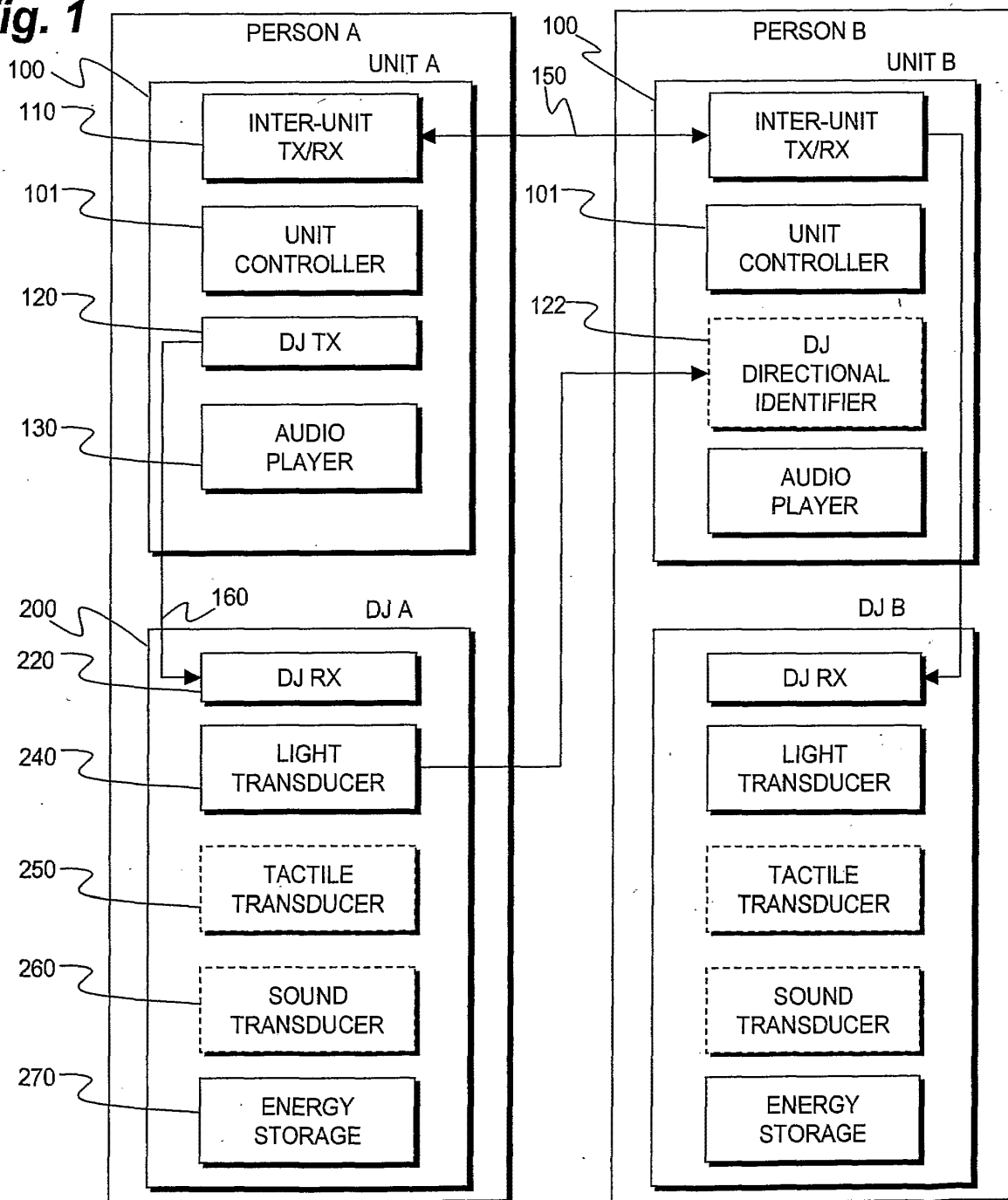
327. The system of claim 320 wherein the transmitter transmits via analog radio frequency communications.

5 328. The system of claim 327 wherein the analog radio frequency communications are at a frequency set by a crystal oscillator and wherein the crystal oscillator can be manually replaced.

329. The system of claim 320 wherein the transmitter additionally transmits the audio signal to wireless earphones.

10

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Fig. 1

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Fig. 2A

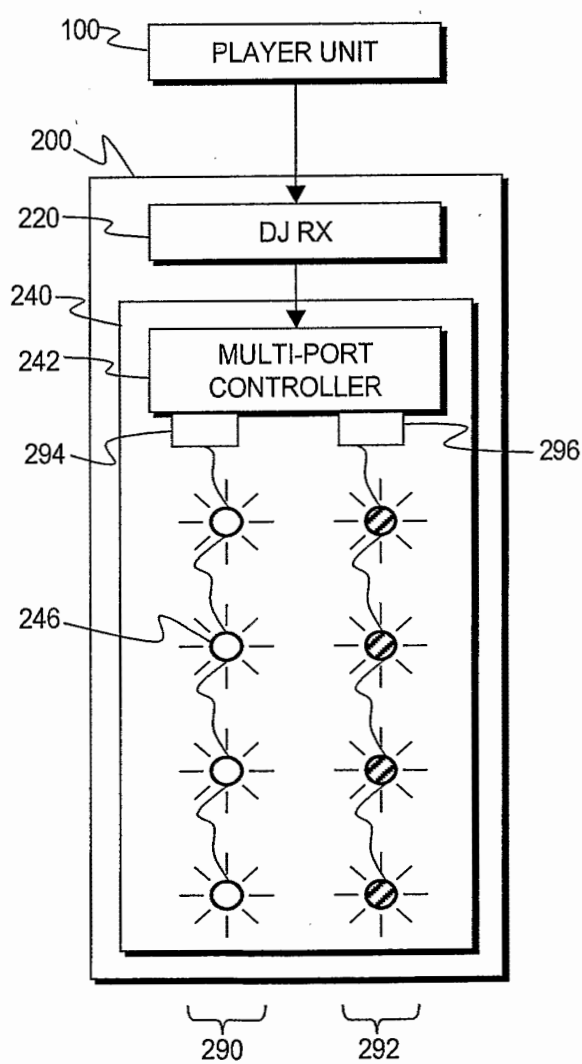
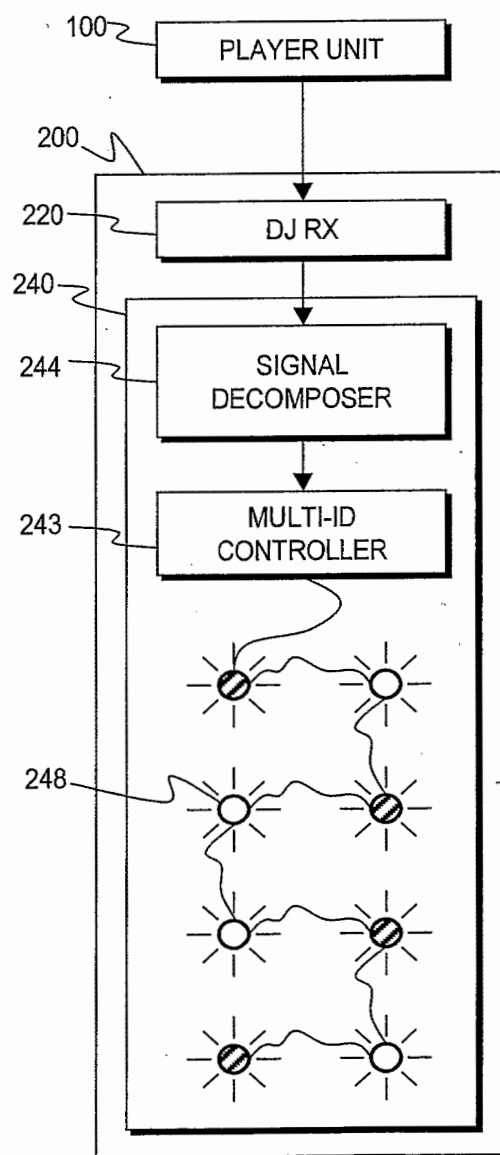
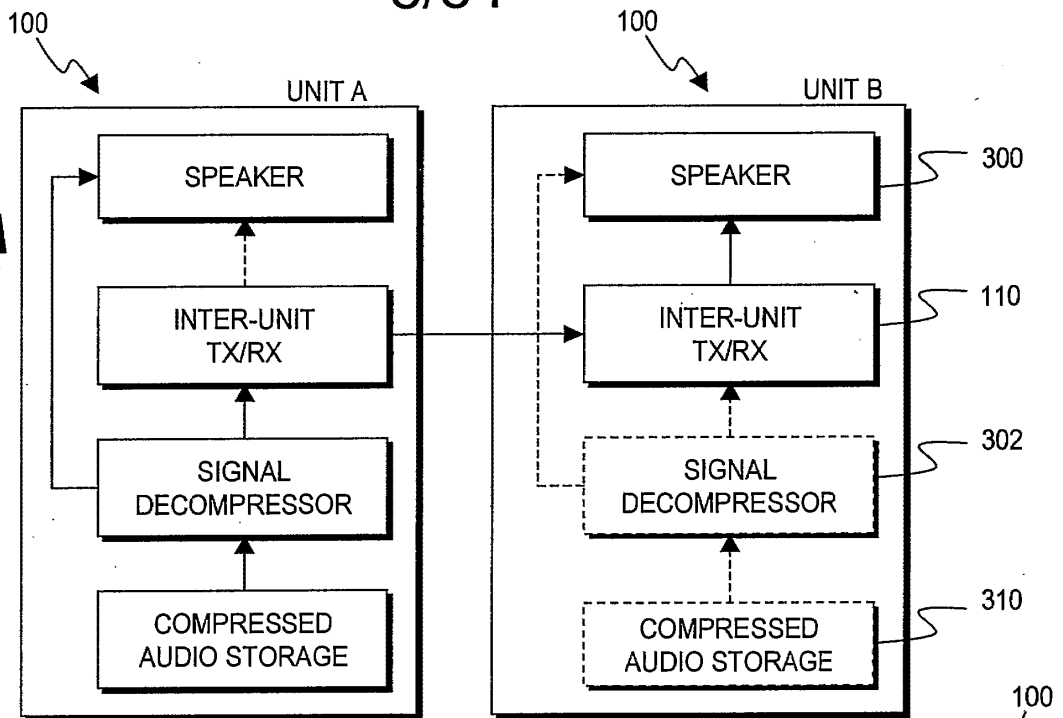
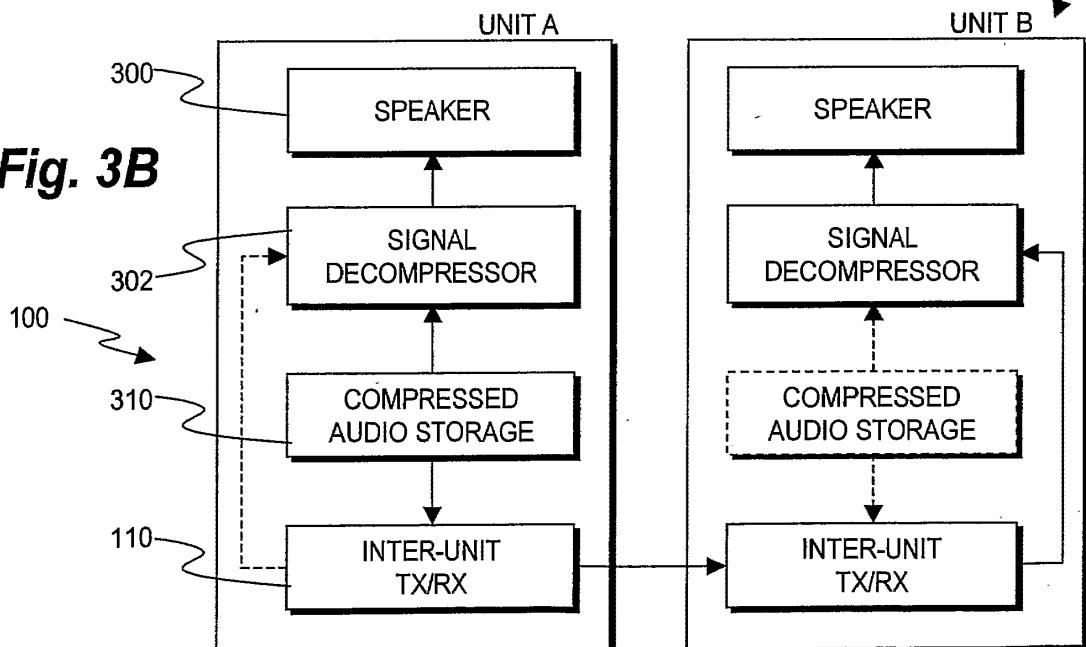


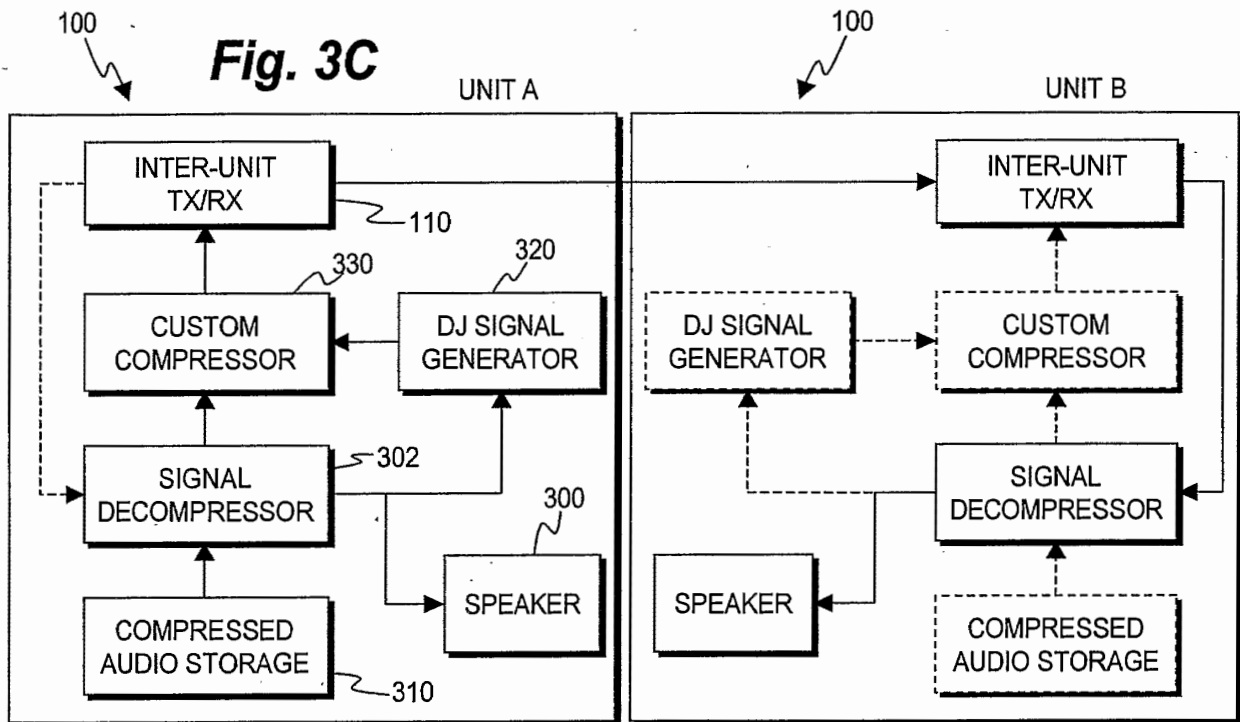
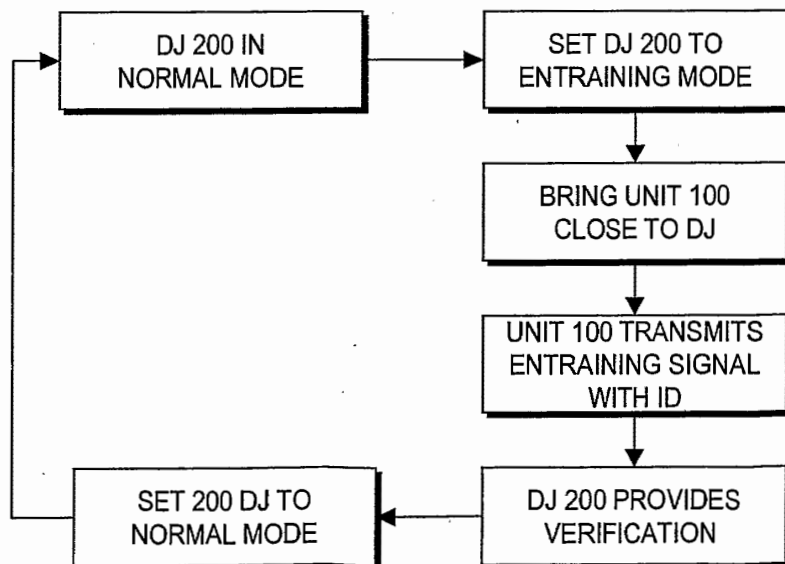
Fig. 2B



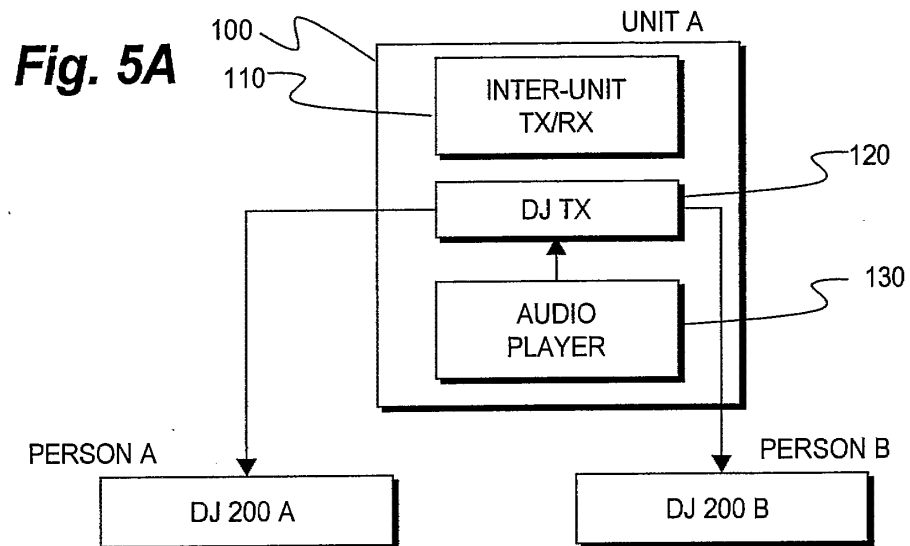
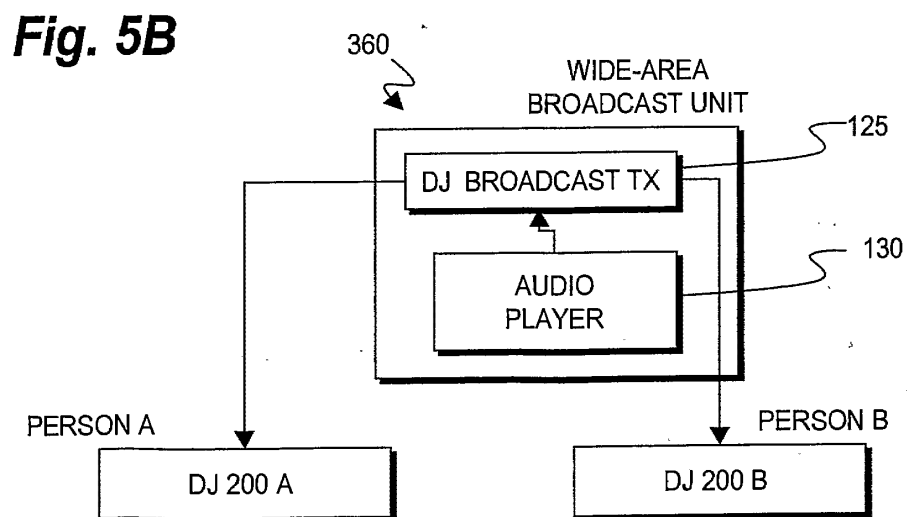
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Fig. 3A**Fig. 3B**

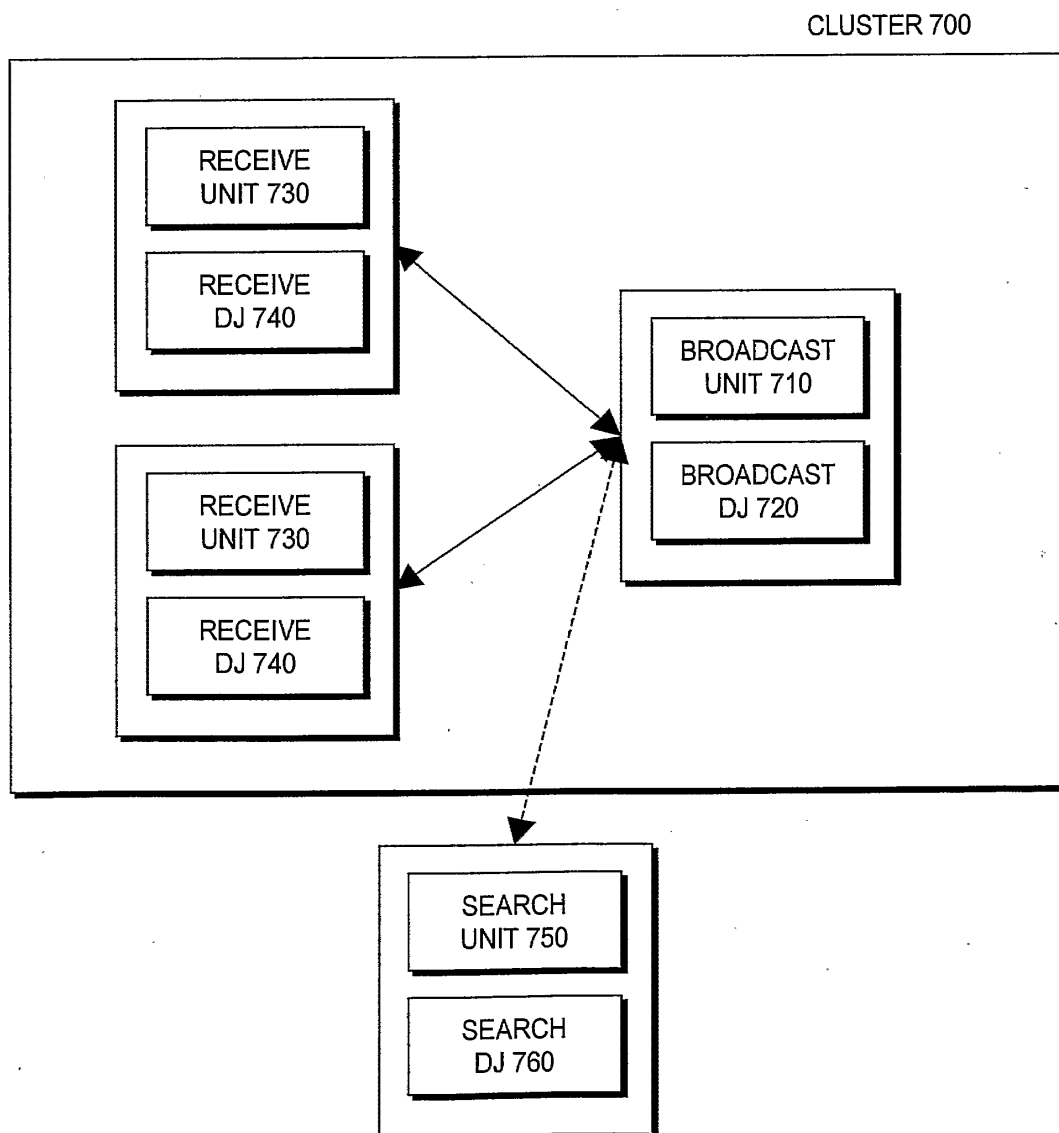
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**Fig. 4**

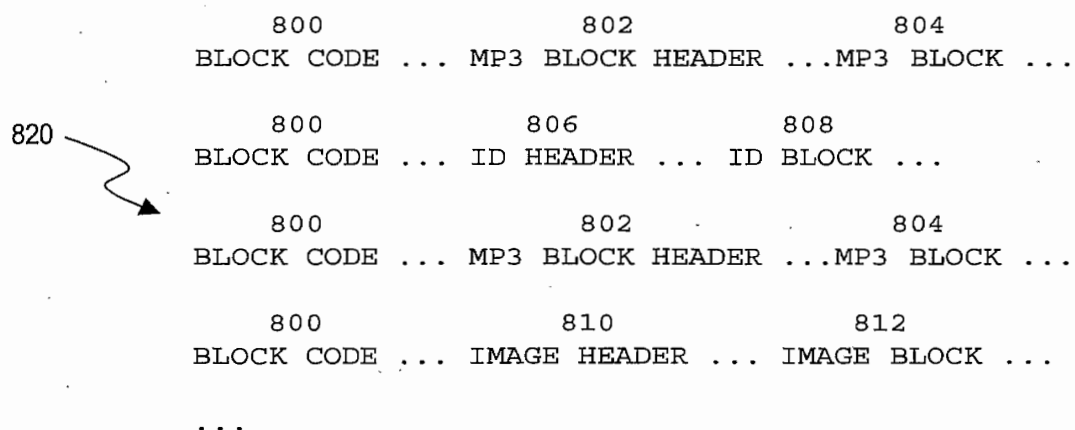
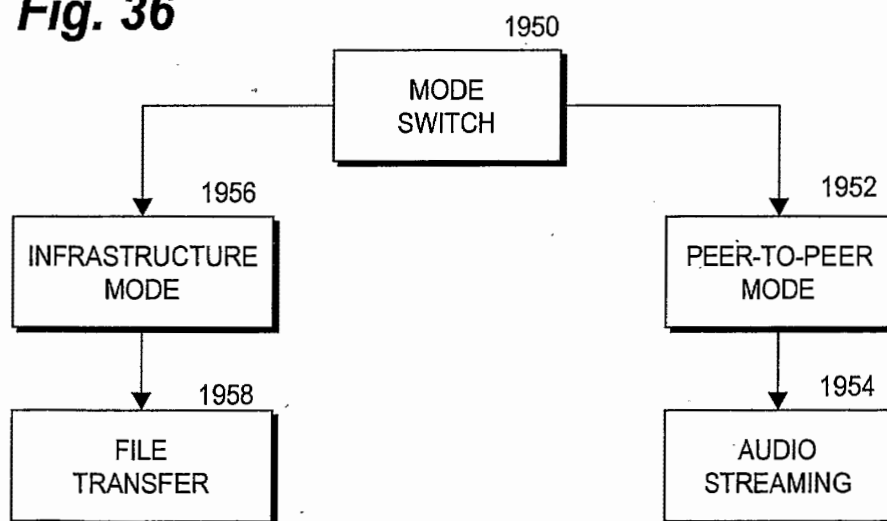
5/34



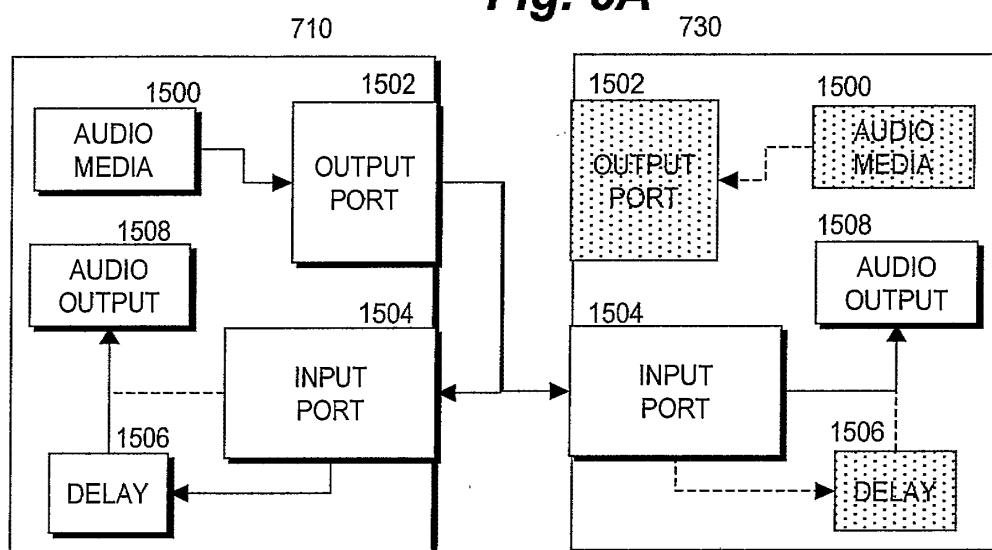
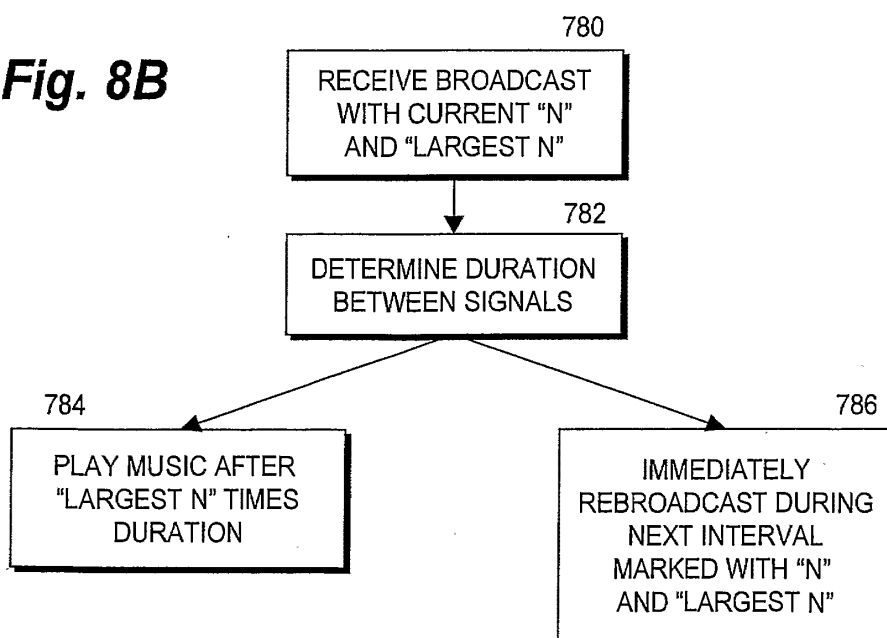
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Fig. 6

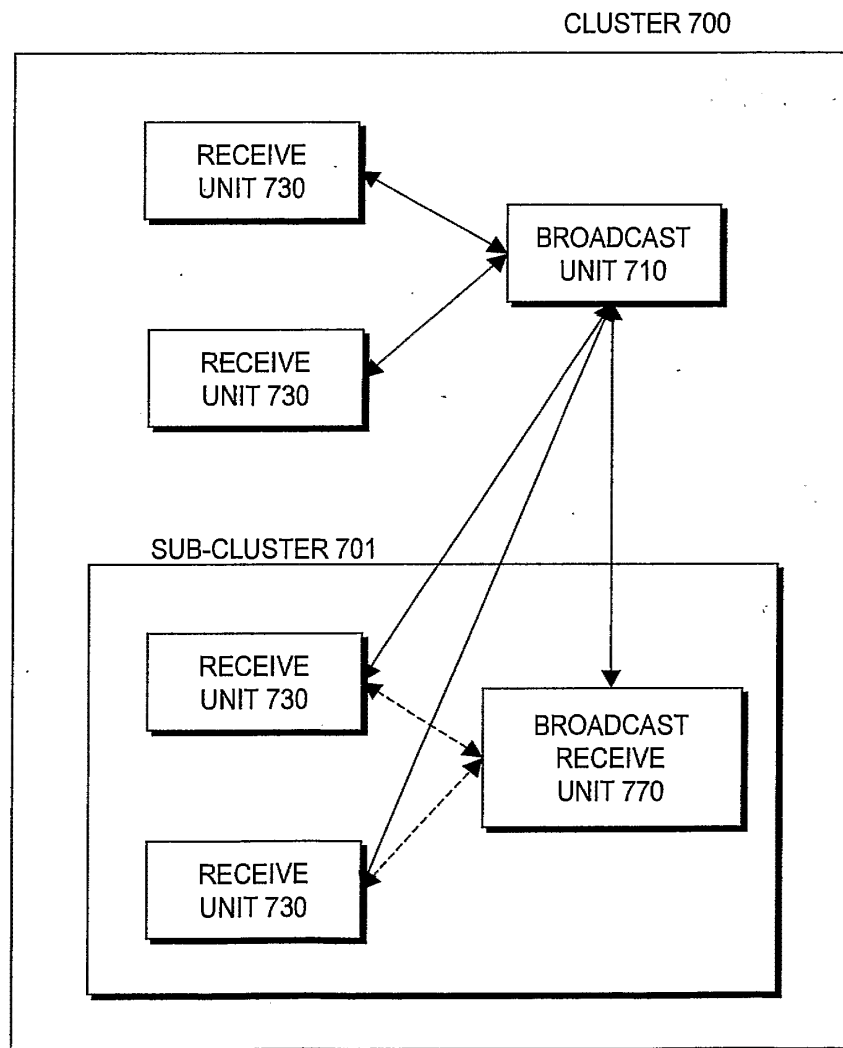
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Fig. 7**Fig. 36**

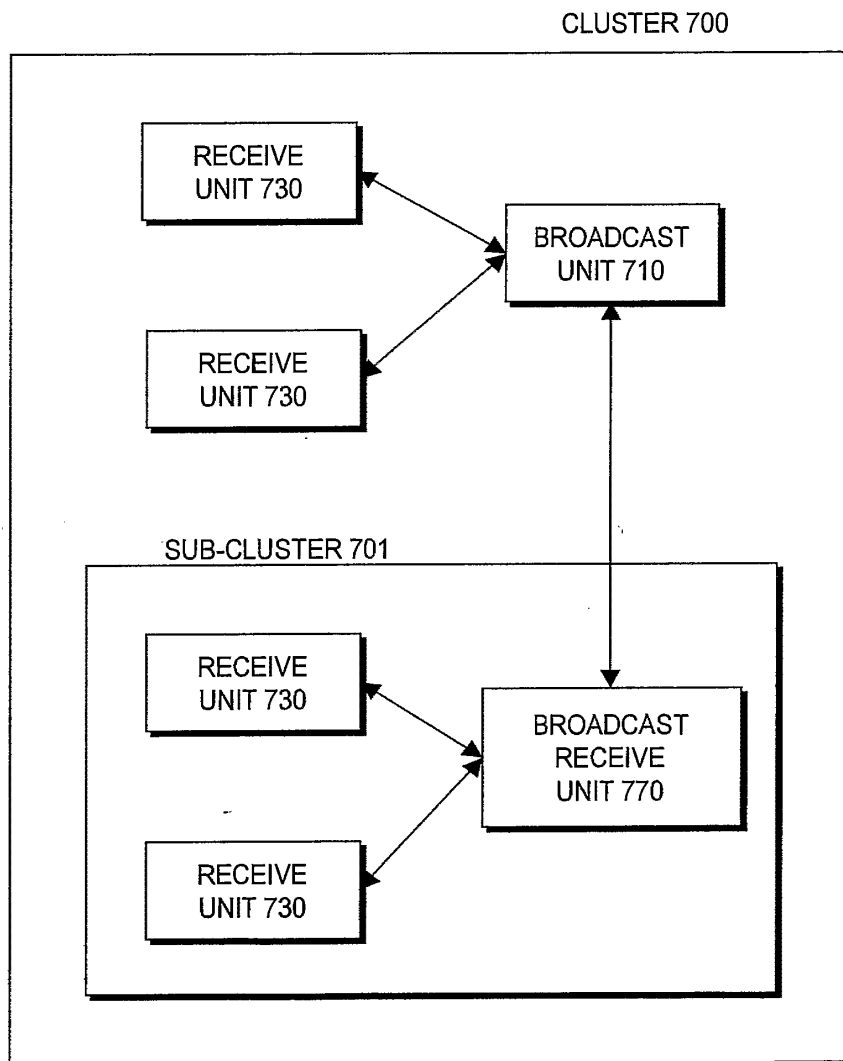
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Fig. 8A**Fig. 8B**

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Fig. 9A

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Fig. 9B

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Fig. 10

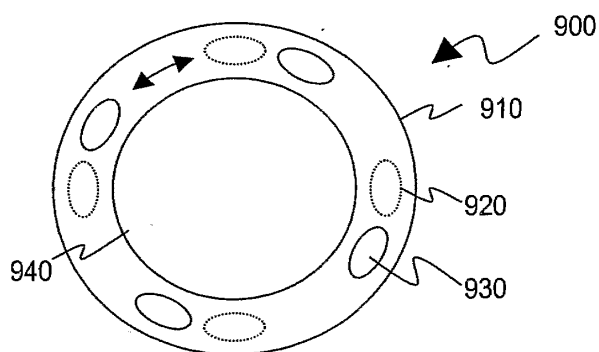


Fig. 11A

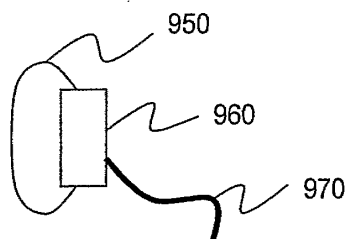
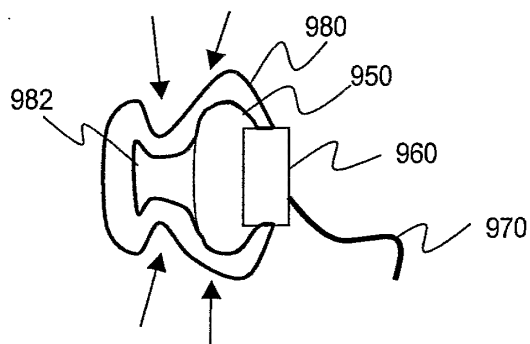


Fig. 11B



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Fig. 12A

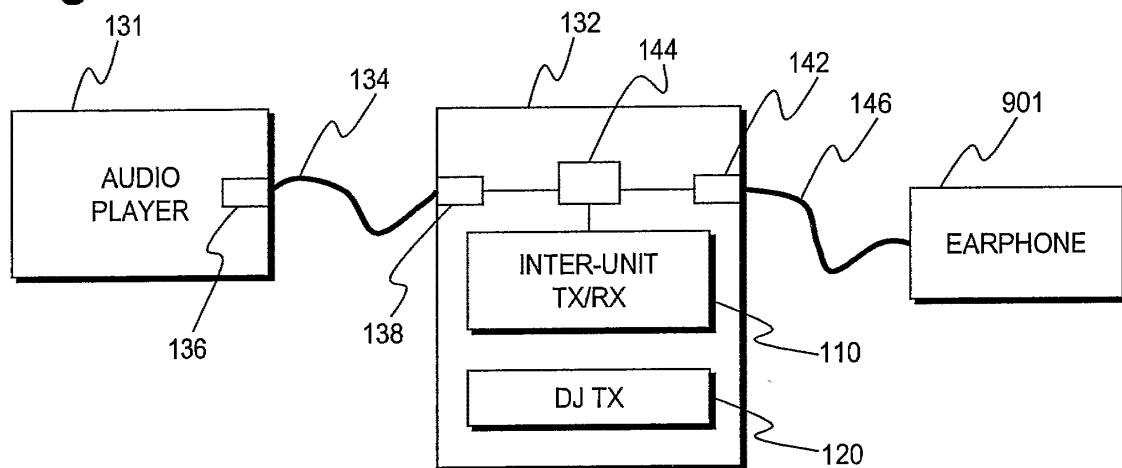
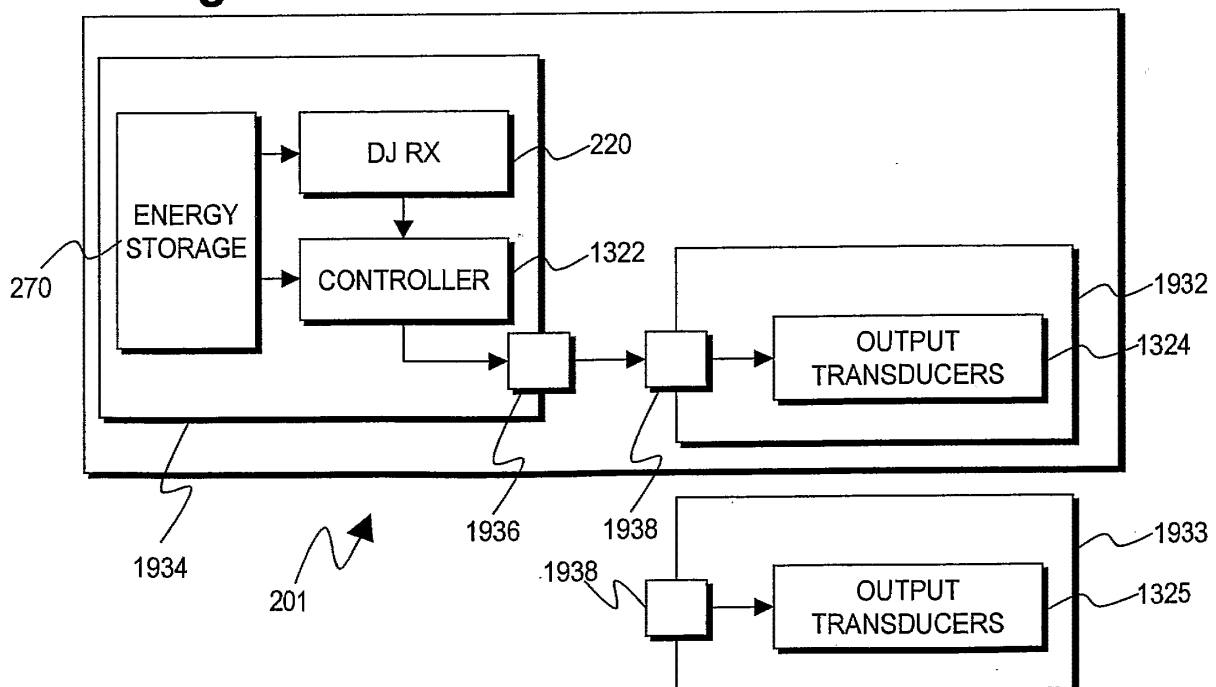
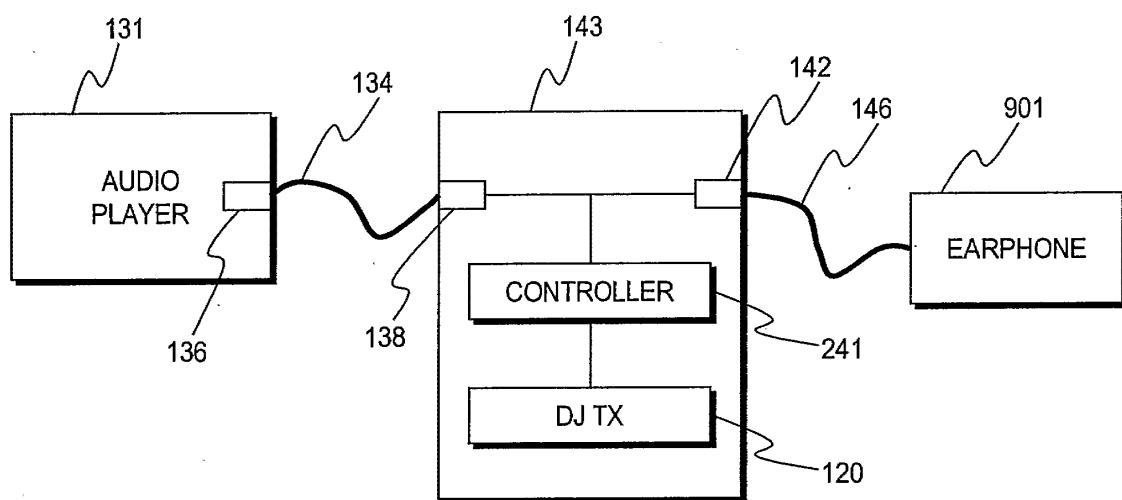


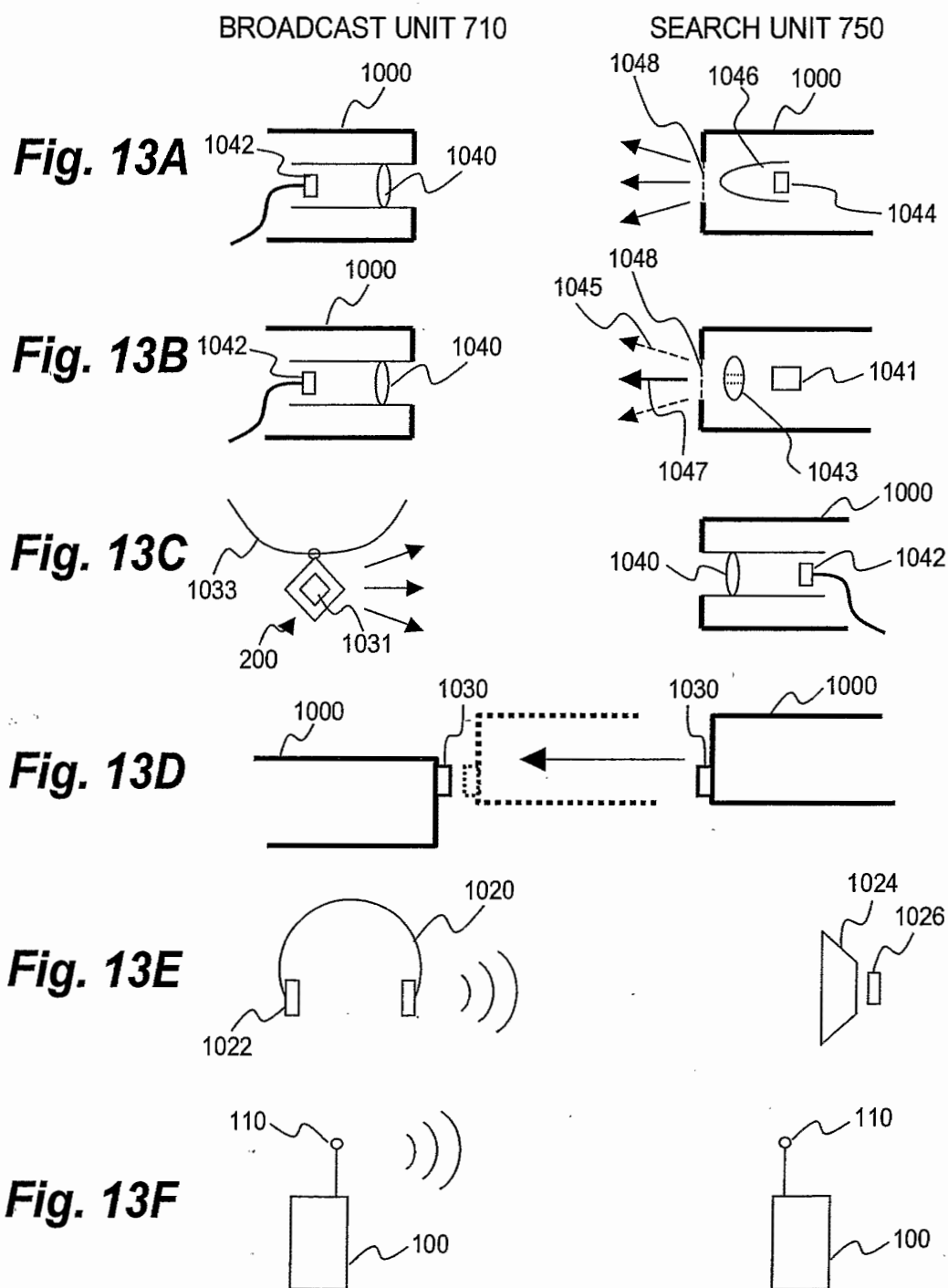
Fig. 12B



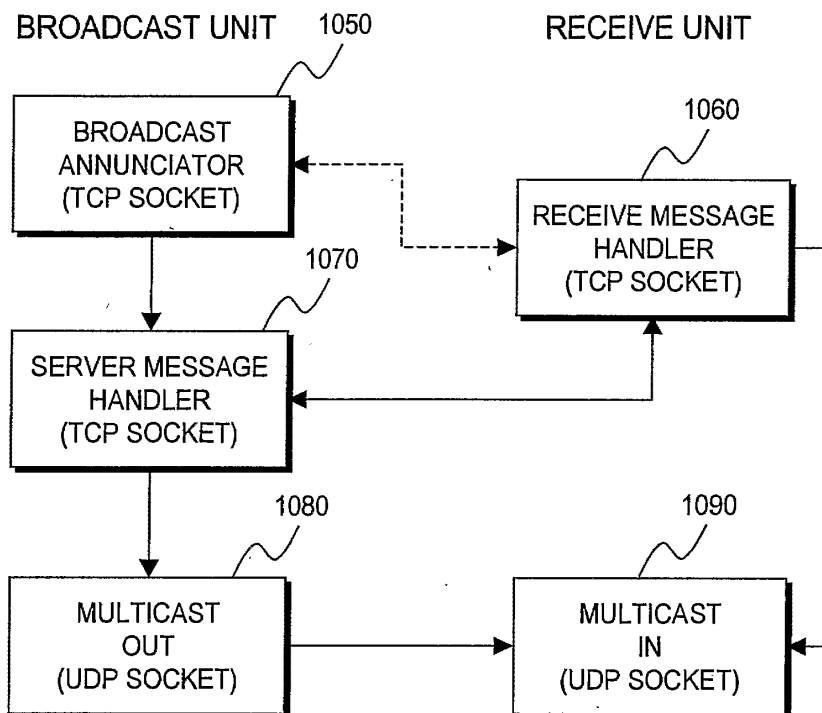
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Fig. 12C

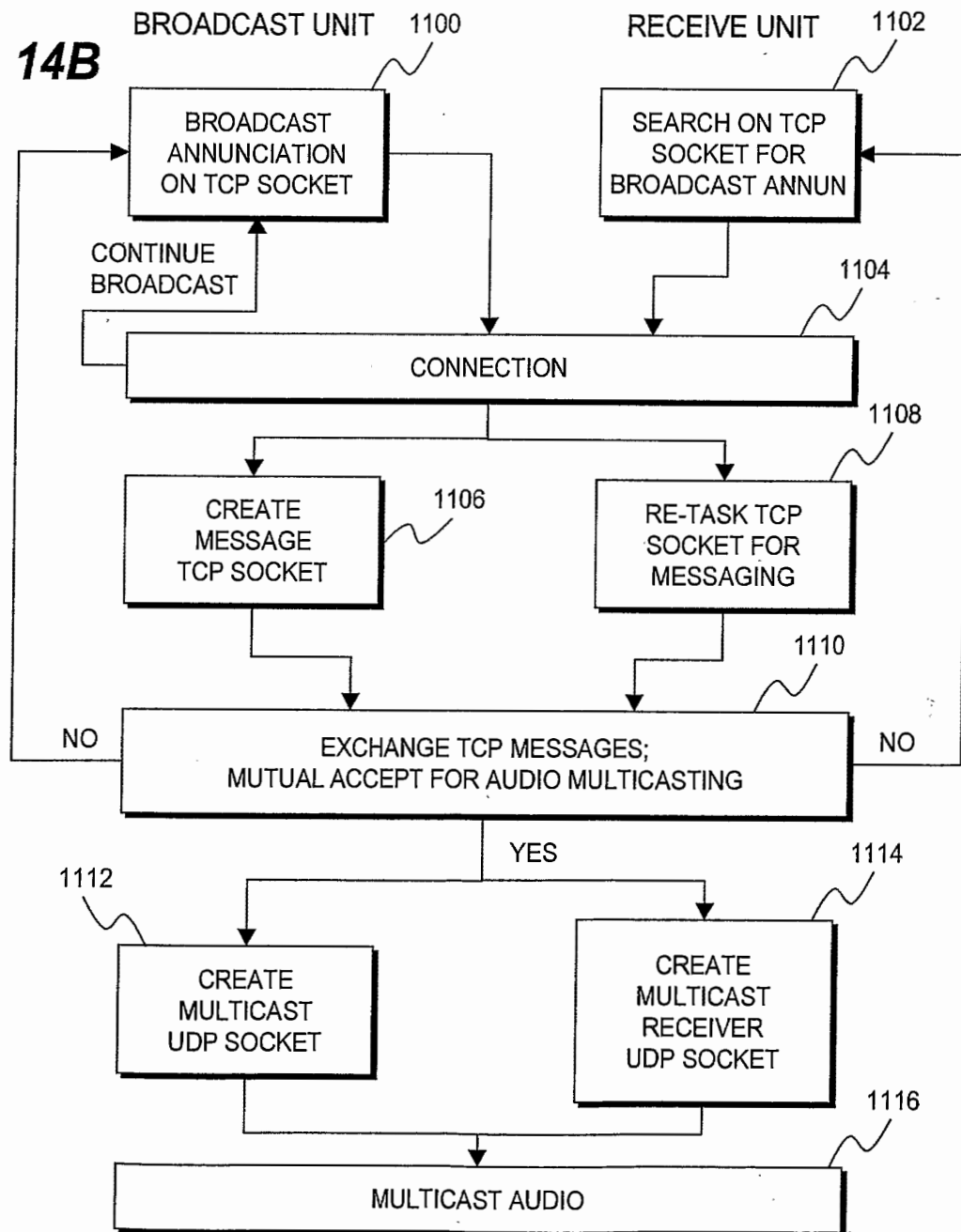
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Fig. 14A

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Fig. 14B

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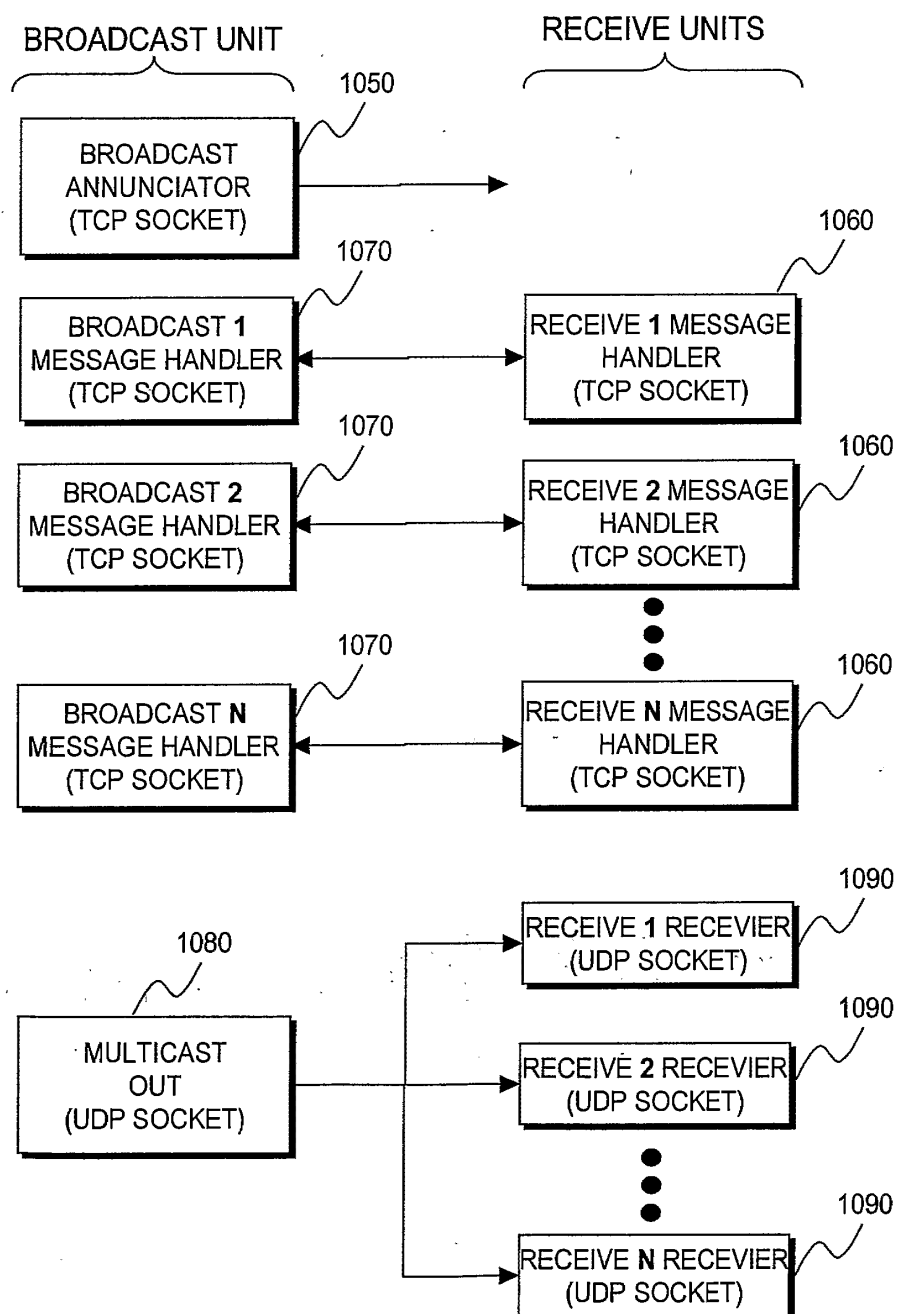
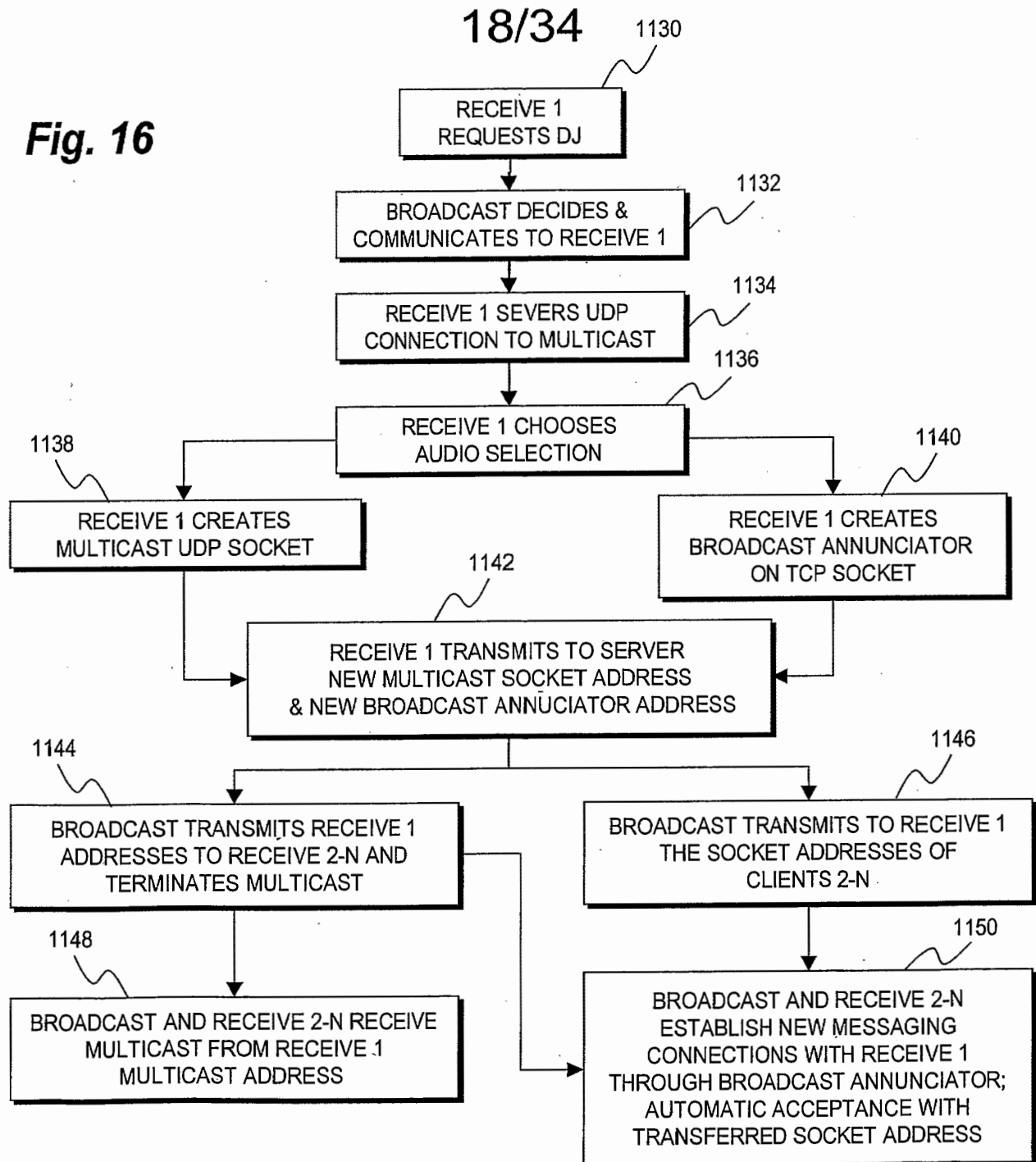
Fig. 15

Fig. 16

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Fig. 17

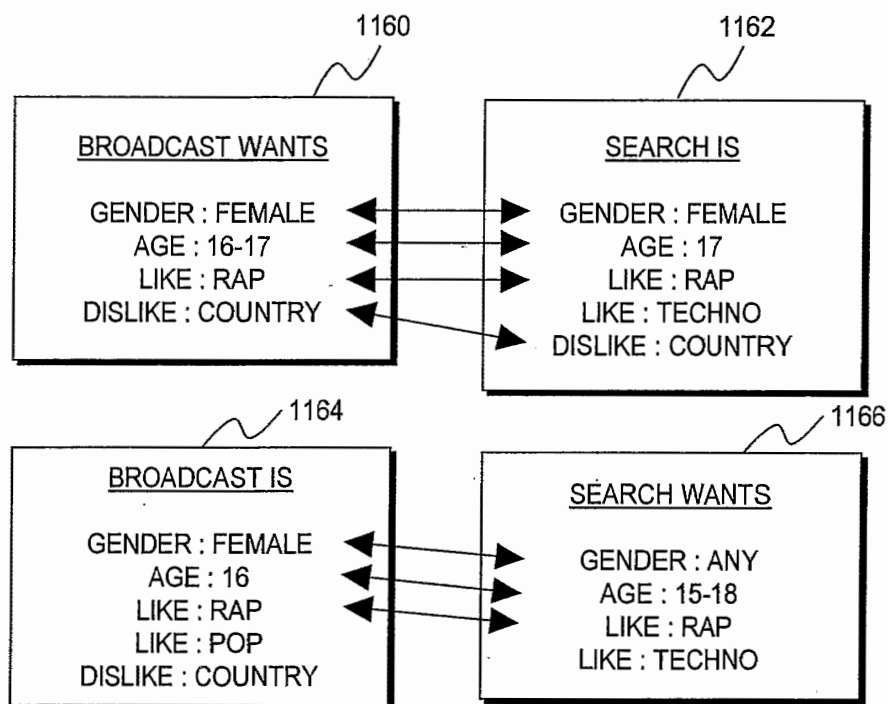


Fig. 18A

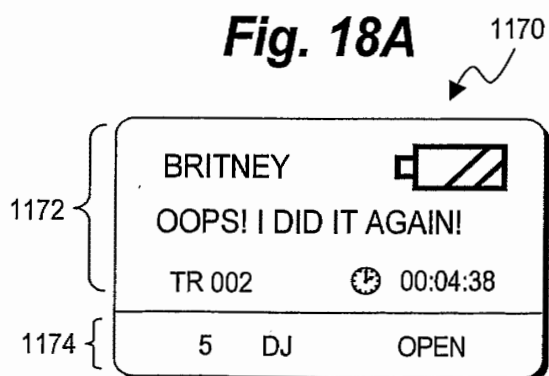
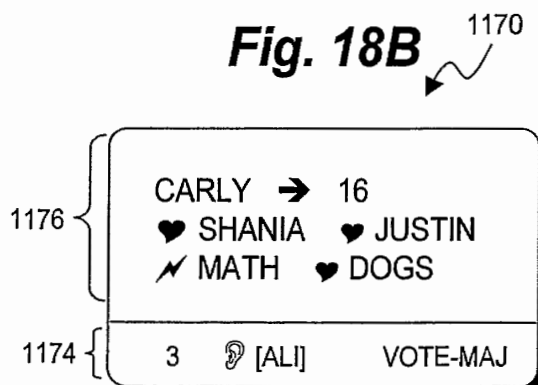


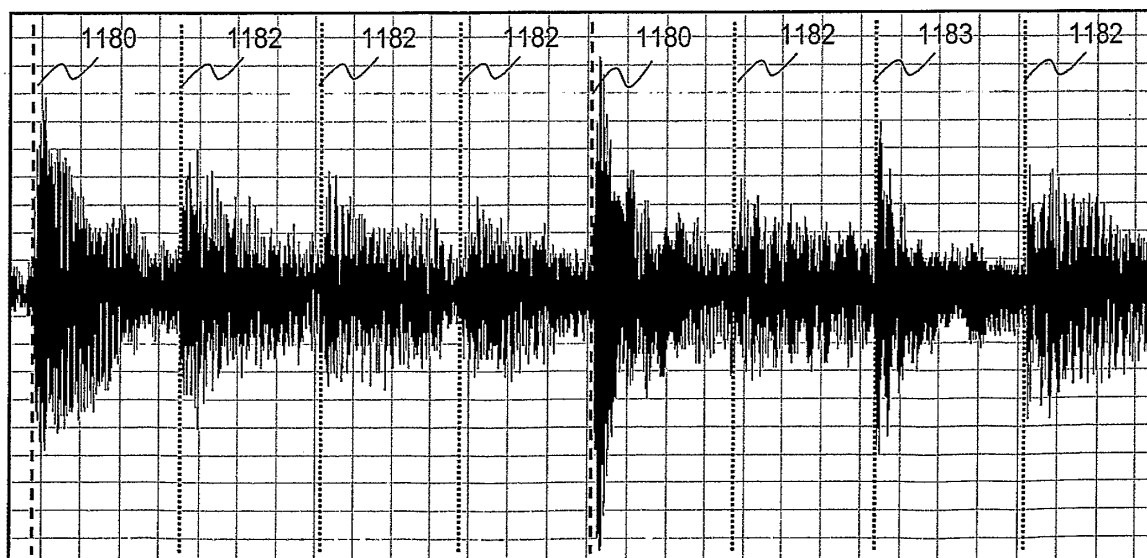
Fig. 18B



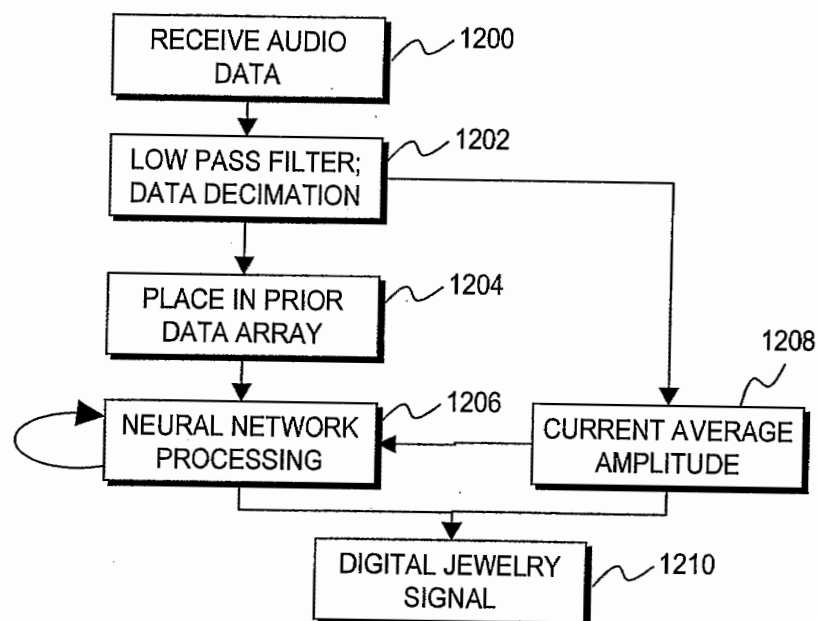
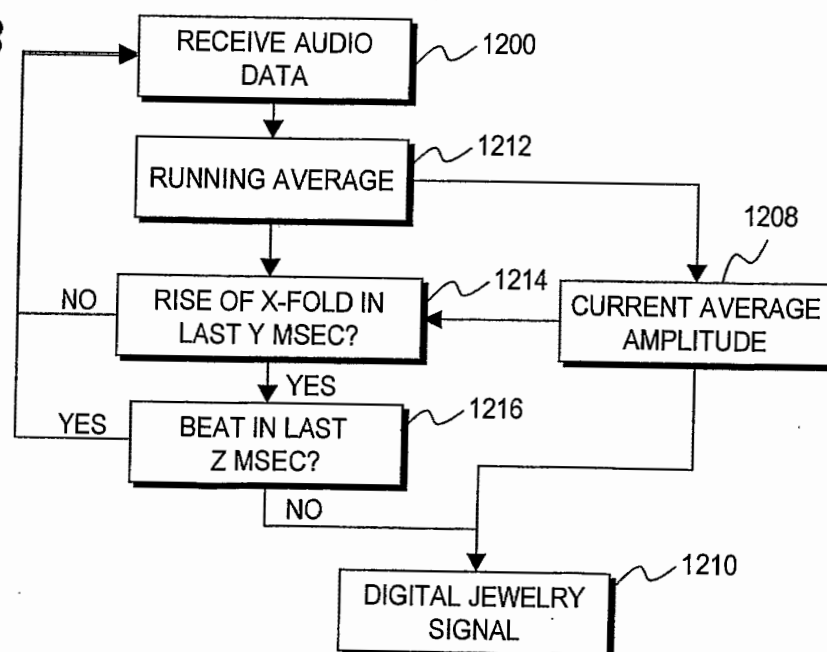
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Fig. 19

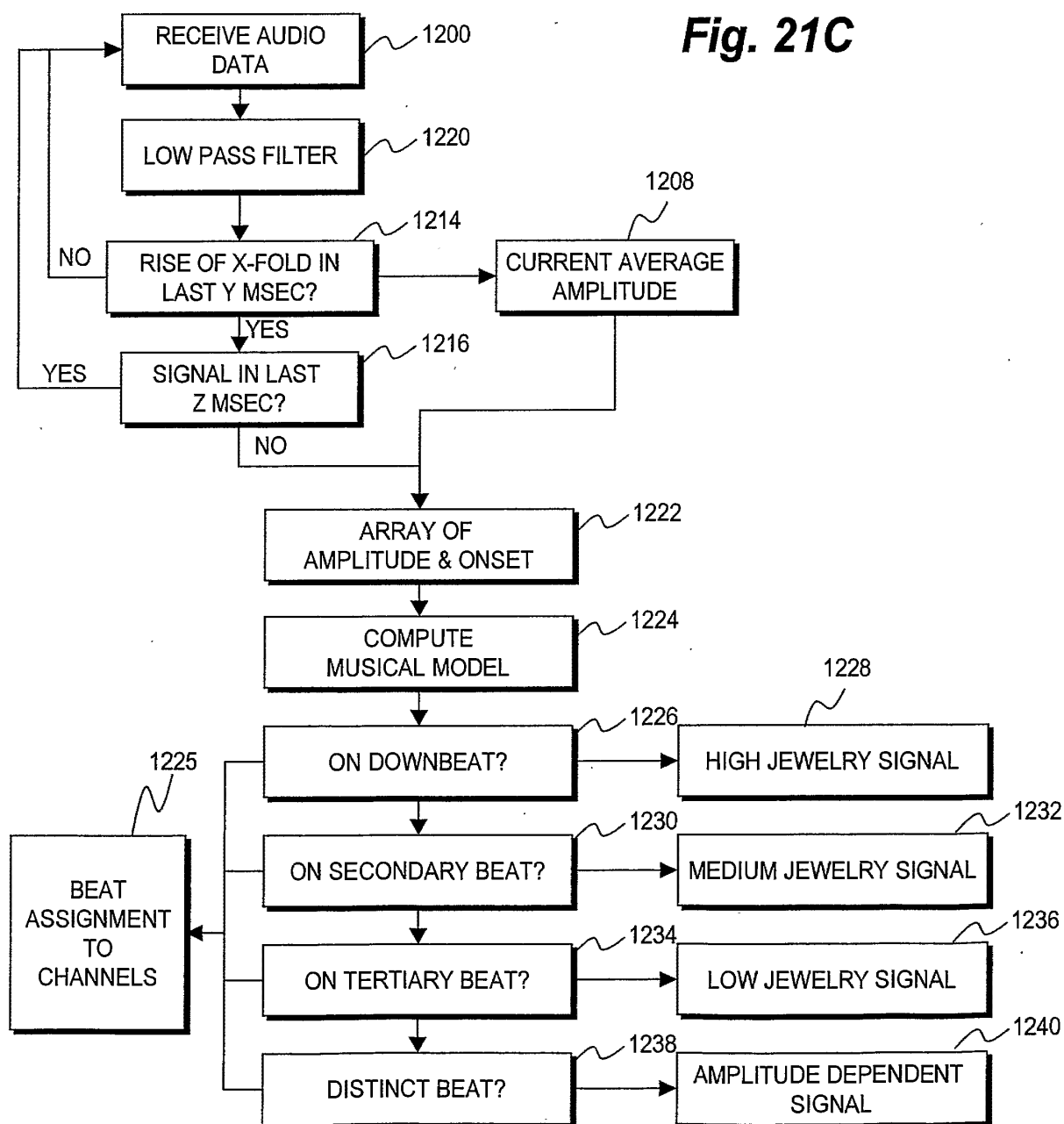
RULE	METHOD
BROADCASTER	ACCEPTED WHEN BROADCASTER INDICATES "YES". REJECTED WHEN BROADCASTER INDICATES "NO".
MAJORITY	ACCEPTED WHEN $(N/2)+1$ INDICATE "YES". REJECTED WHEN $(N/2)$ INDICATE "NO".
UNANIMOUS	ACCEPTED WHEN N MEMBERS INDICATE "YES". REJECTED WHEN 1 MEMBER INDICATES "NO".
TIMED MAJORITY	MACHINE INDICATES ONSET OF VOTING, AND COUNTS DOWN TO VOTING TERMINATION. ACCEPTED IF $(Q/2)+1$ INDICATE "YES" BEFORE TERMINATION; OTHERWISE "NO".
SYNCHRONIZED MAJORITY	MACHINE COUNTS DOWN TO VOTE. ACCEPTED IF $(Q/2)+1$ INDICATE "YES" WITHIN PREDETERMINED TIME OF COUNTDOWN, OTHERWISE "NO".

Fig. 20

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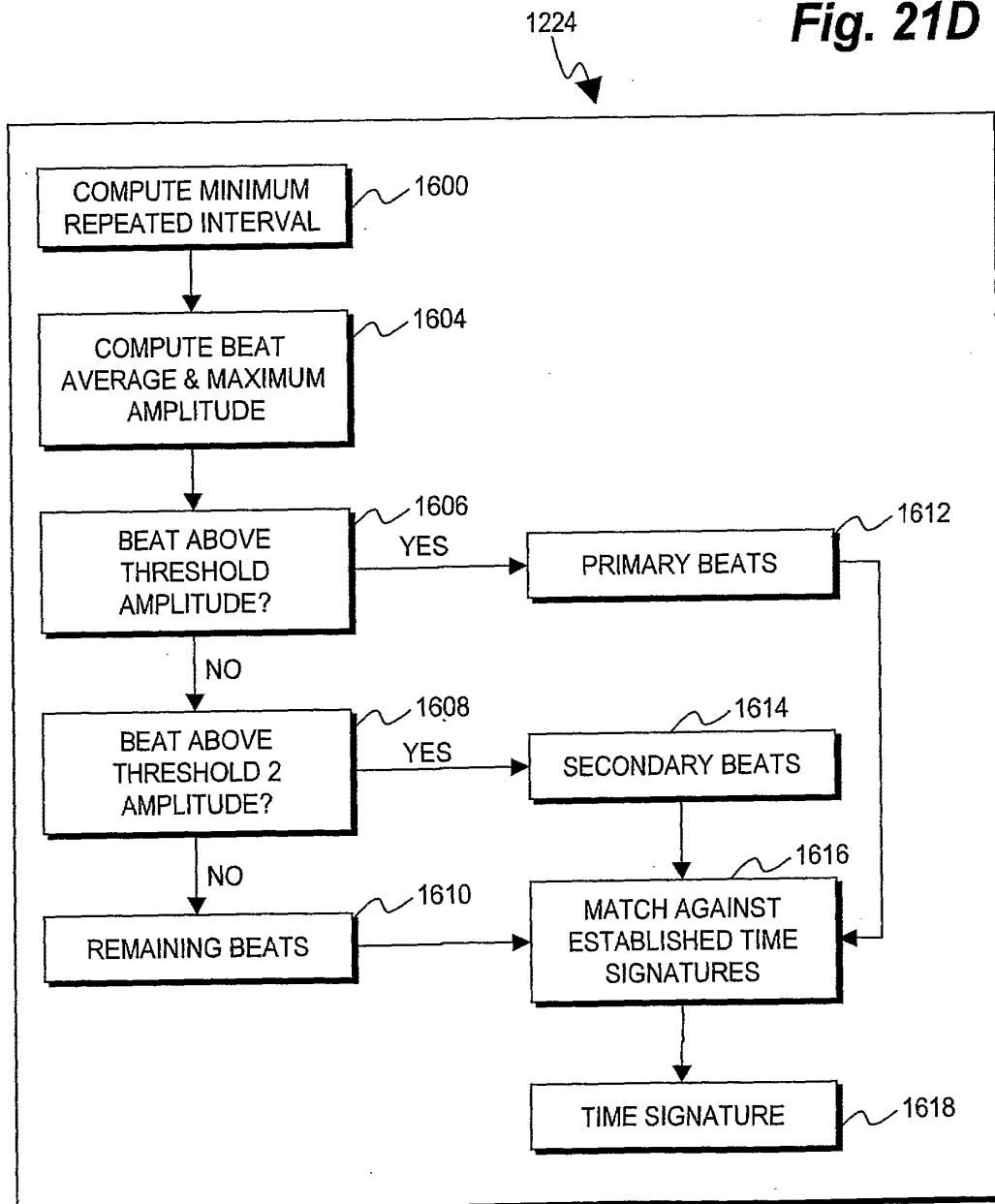
Fig. 21A**Fig. 21B**

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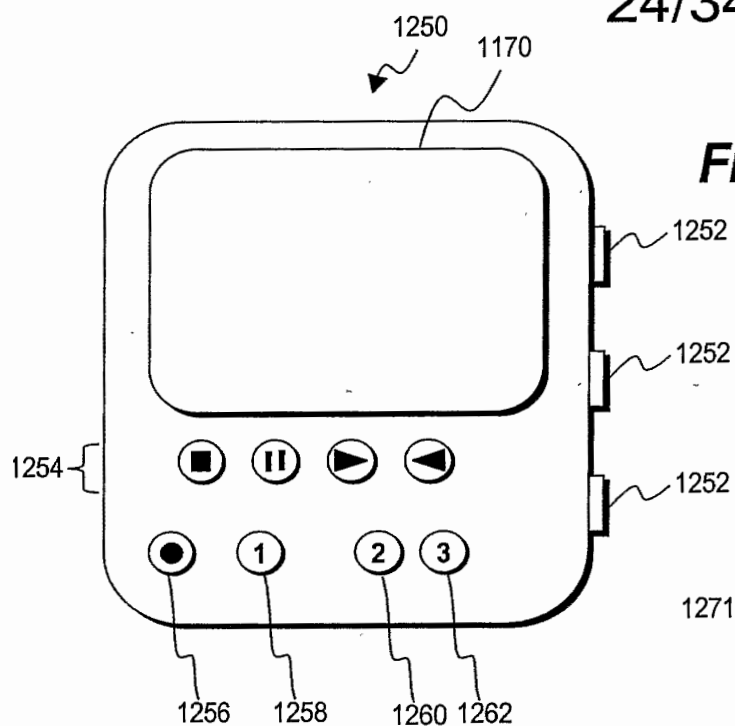
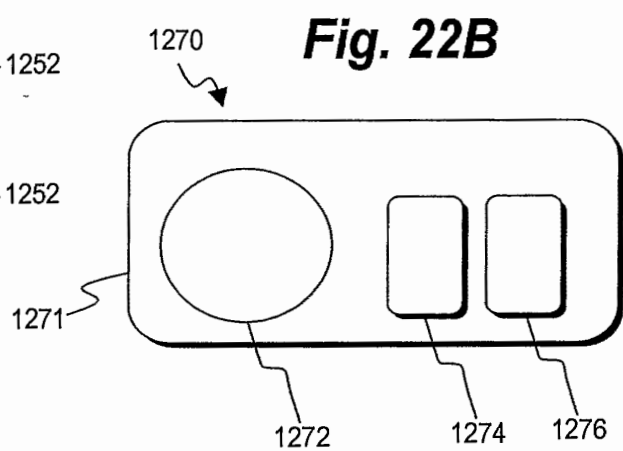
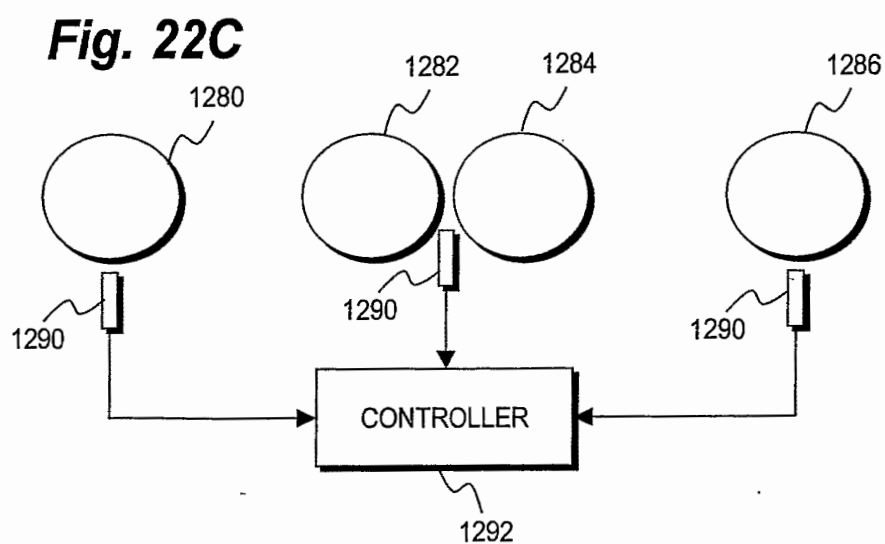
Fig. 21C

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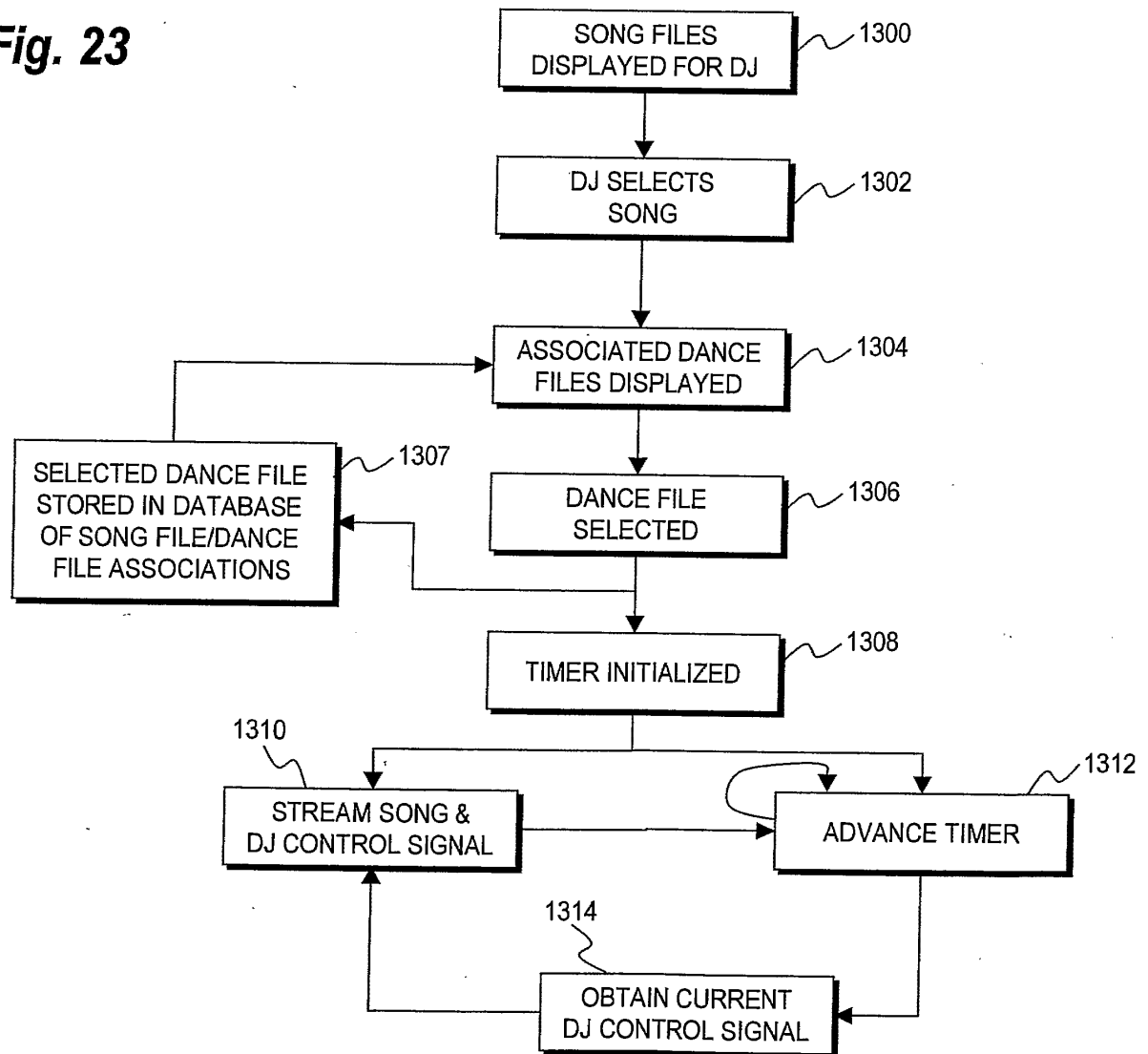
Fig. 21D



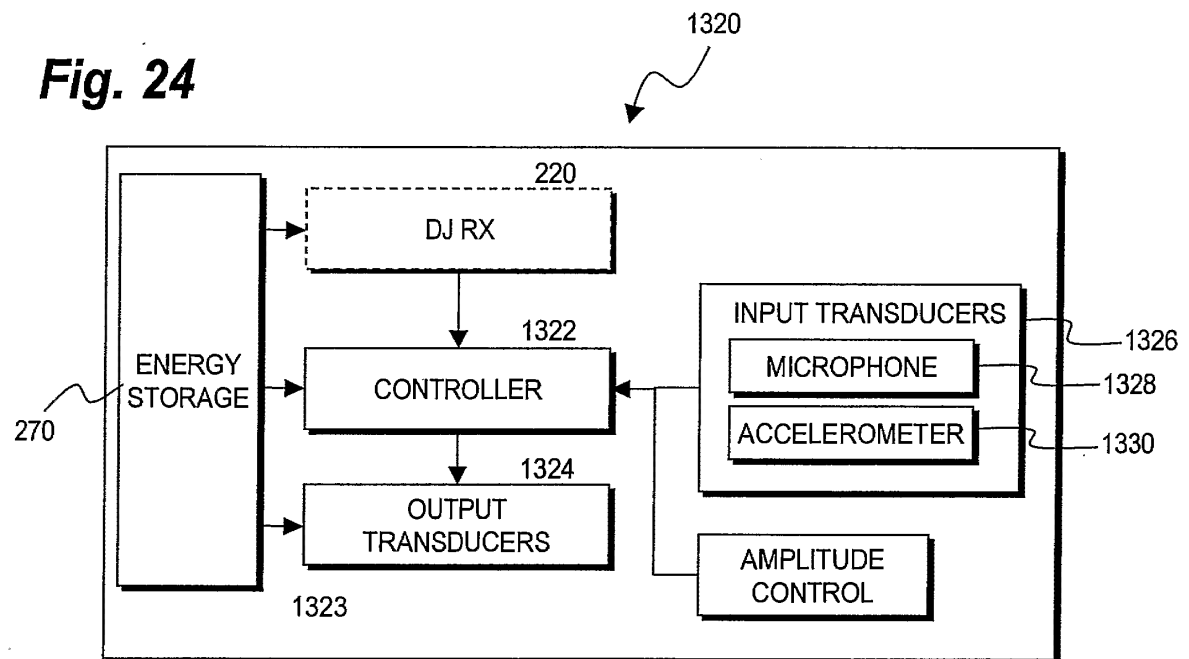
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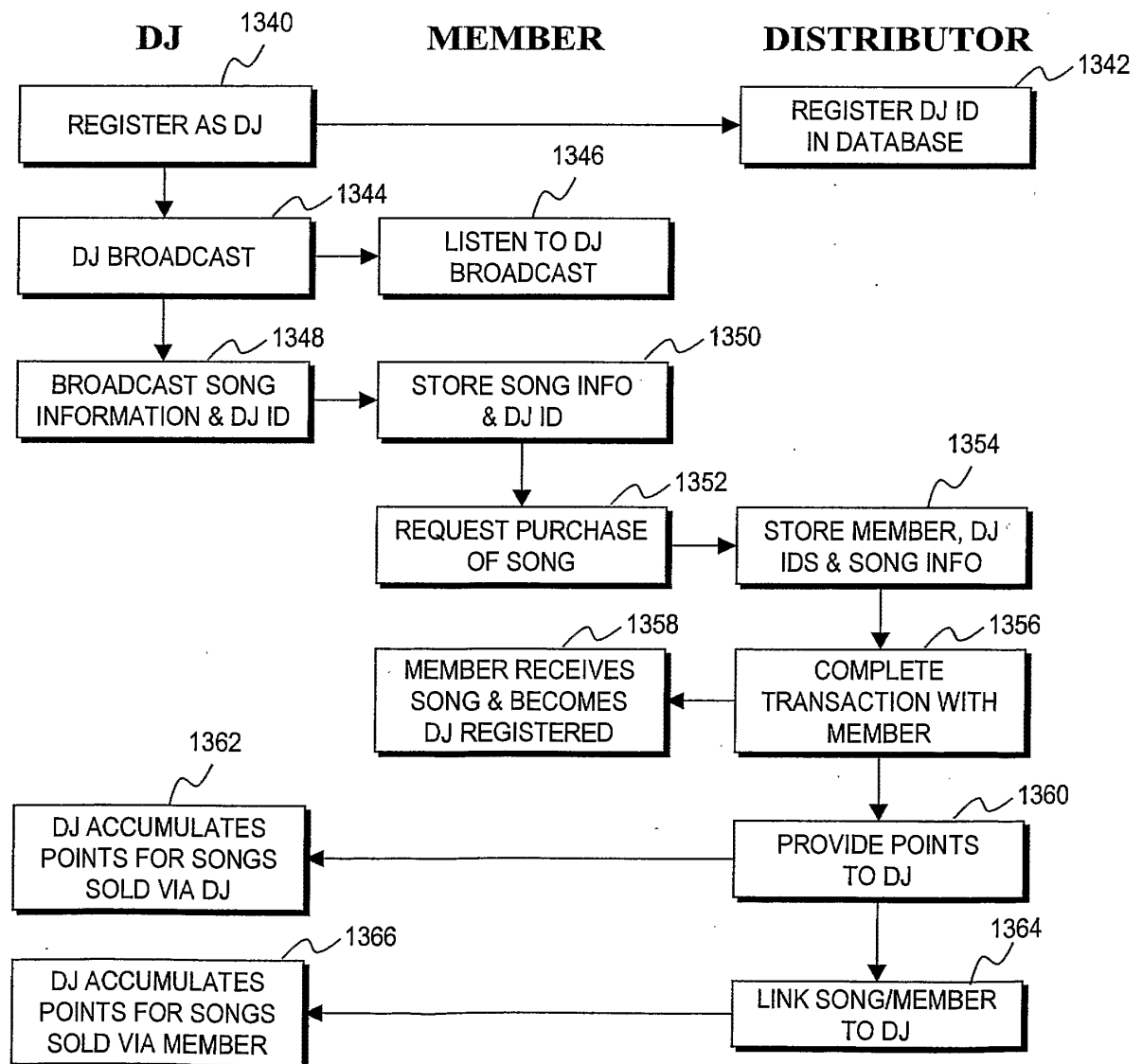
25/34

Fig. 23

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Fig. 24

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Fig. 25

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Fig. 26

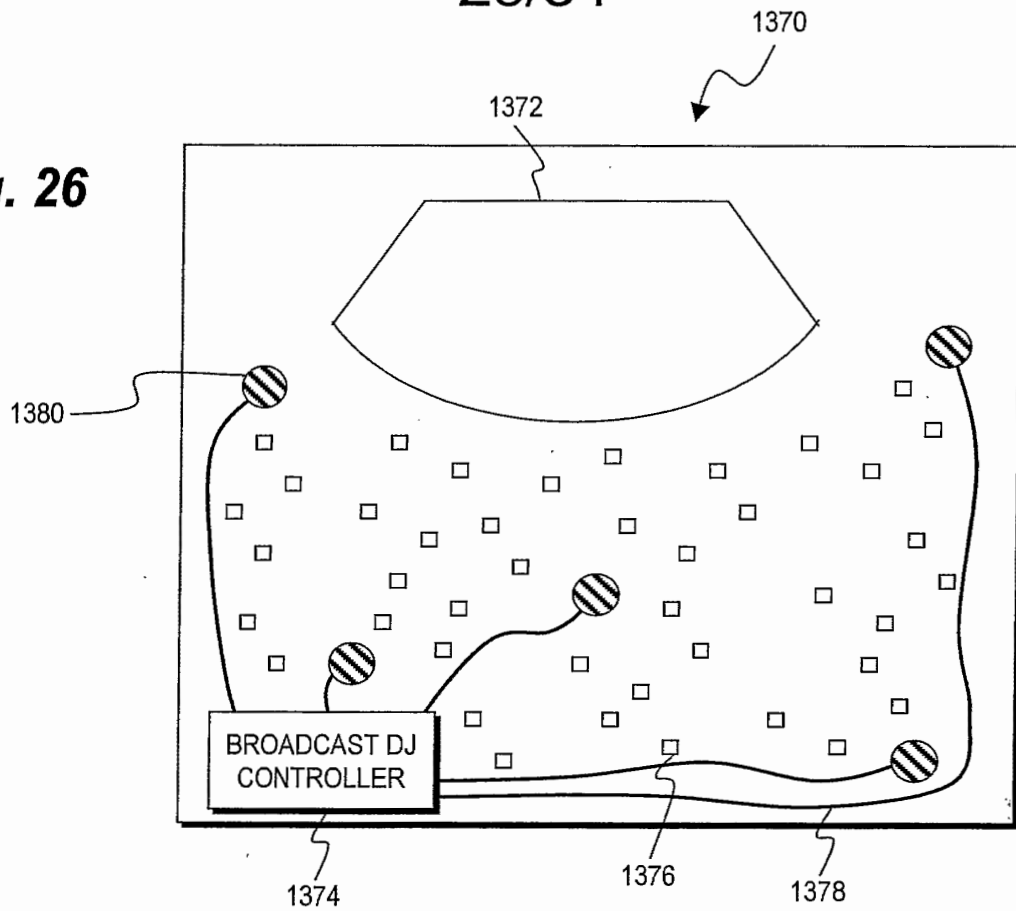
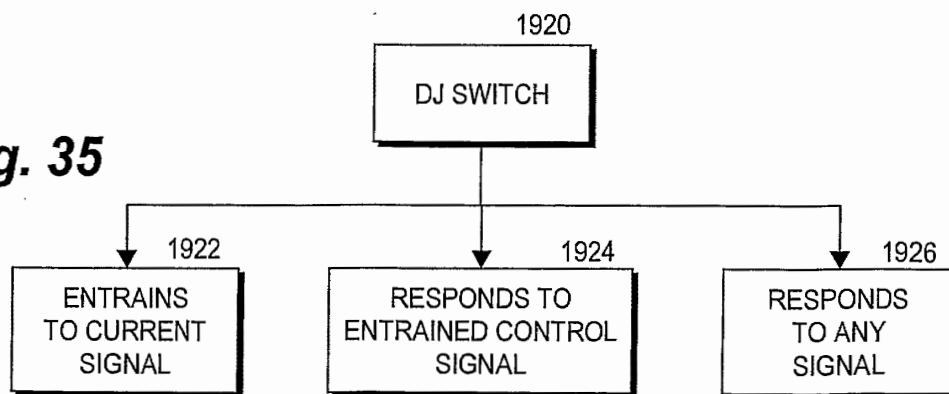
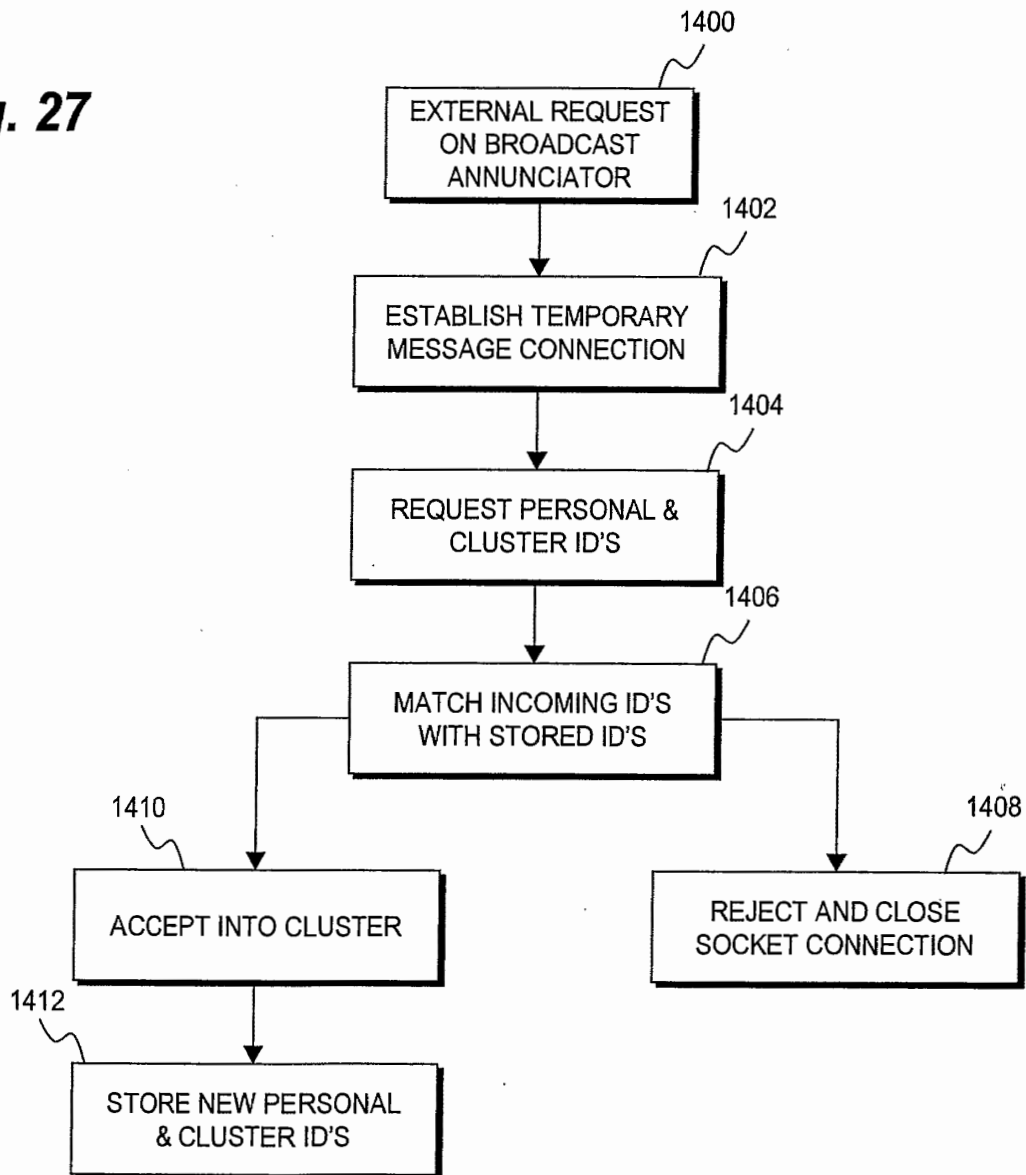


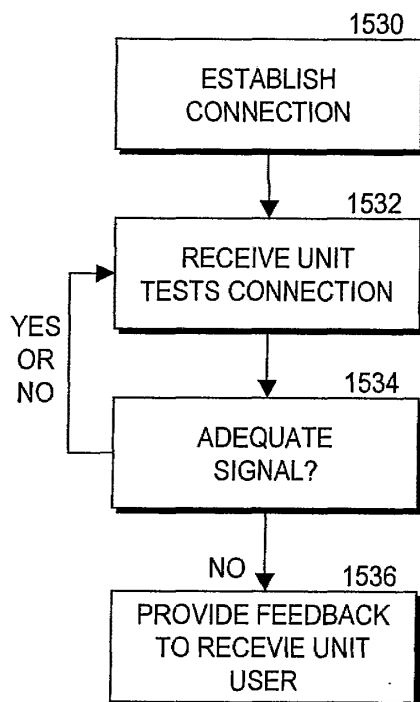
Fig. 35



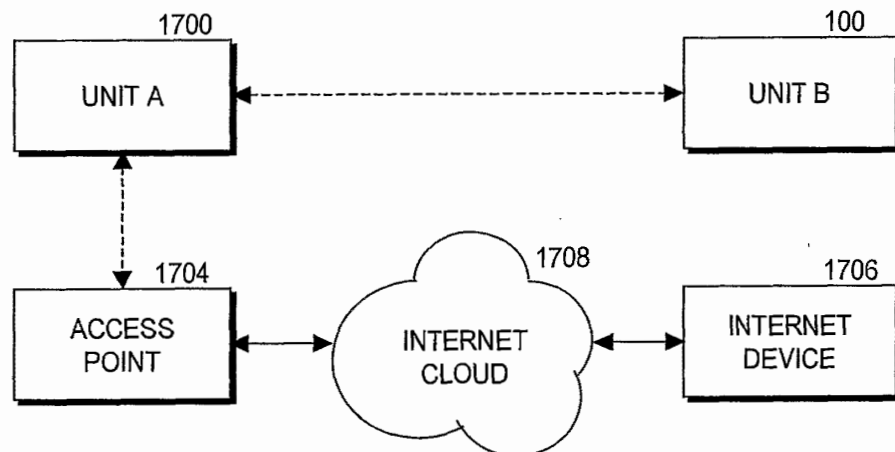
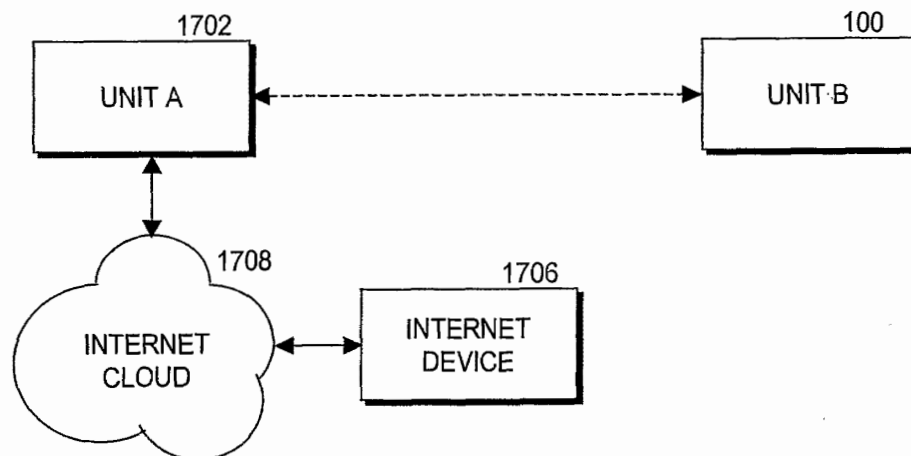
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Fig. 27

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Fig. 28

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Fig. 29A**Fig. 29B**

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Fig. 30

Fig. 30

1800

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1802

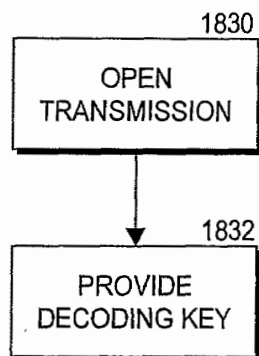
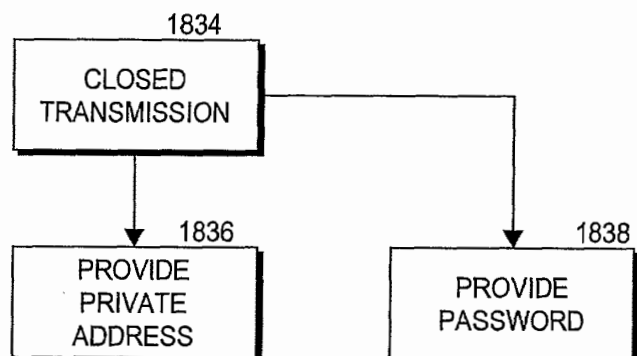
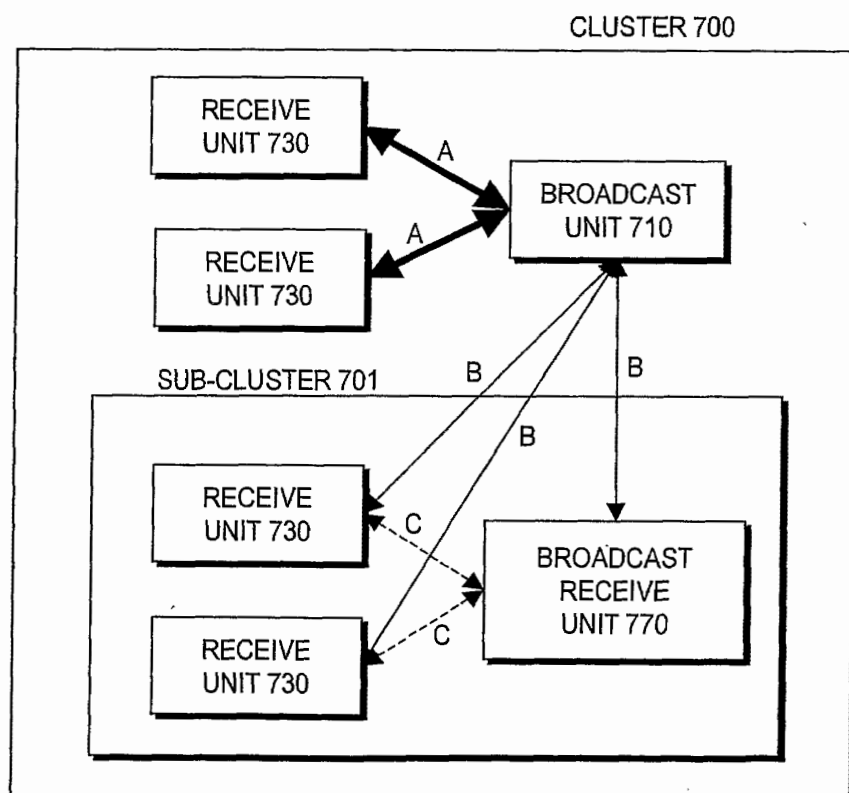
Fig. 31

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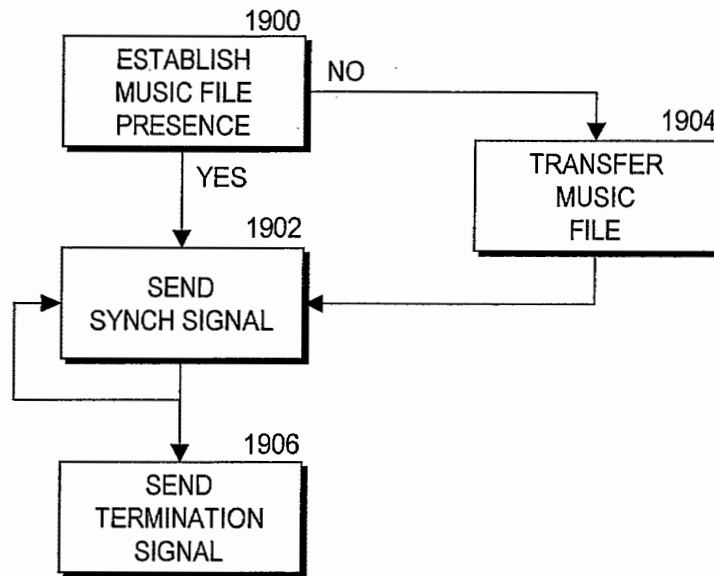
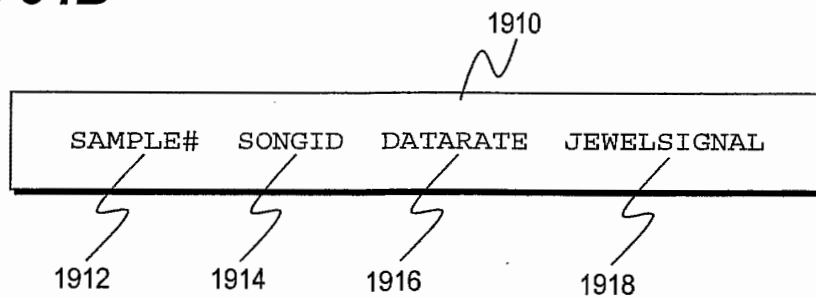
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Fig. 32A**Fig. 32B****Fig. 33**

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Fig. 34A**Fig. 34B**

Electronic Patent Application Fee Transmittal				
Application Number:		13864249		
Filing Date:		17-Apr-2013		
Title of Invention:		METHOD AND APPARATUS FOR PLAYING AND PROVIDING AUDIO CONTENT IN A SYNCHRONY GROUP		
First Named Inventor/Applicant Name:		Nicholas A. J. Millington		
Filer:		Benjamin Michael Urban		
Attorney Docket Number:		13-0402 (MBHB 15-518-CON)		
Filed as Large Entity				
Filing Fees for Utility under 35 USC 111(a)				
Description	Fee Code	Quantity	Amount	Sub-Total in USD(\$)
Basic Filing:				
Pages:				
Claims:				
Miscellaneous-Filing:				
Petition:				
Patent-Appeals-and-Interference:				
Post-Allowance-and-Post-Issuance:				
Extension-of-Time:				

Description	Fee Code	Quantity	Amount	Sub-Total in USD(\$)
Miscellaneous:				
Submission- Information Disclosure Stmt	1806	1	180	180
Total in USD (\$)				180

Electronic Acknowledgement Receipt	
EFS ID:	28346991
Application Number:	13864249
International Application Number:	
Confirmation Number:	9806
Title of Invention:	METHOD AND APPARATUS FOR PLAYING AND PROVIDING AUDIO CONTENT IN A SYNCHRONY GROUP
First Named Inventor/Applicant Name:	Nicholas A. J. Millington
Customer Number:	107361
Filer:	Benjamin Michael Urban
Filer Authorized By:	
Attorney Docket Number:	13-0402 (MBHB 15-518-CON)
Receipt Date:	14-FEB-2017
Filing Date:	17-APR-2013
Time Stamp:	15:49:55
Application Type:	Utility under 35 USC 111(a)

Payment information:

Submitted with Payment	yes
Payment Type	DA
Payment was successfully received in RAM	\$ 180
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Authorized User	
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Document Number	Document Description	File Name	File Size(Bytes)/ Message Digest	Multi Part /.zip	Pages (if appl.)
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Information:					
58	Non Patent Literature	DEFENDANTS_BRIEF_IN_SUPPORT_10-12-16-.pdf	262459	no	24
			3a022330e79f6dec7f0854059b69b8c079e1e40		
Warnings:					
Information:					
59	Non Patent Literature	Sonos_No14-1330-RGA_DI_206-1_28pages.pdf	299059	no	28
			9f2e542b1e6cf27946890f1b973e4007410a5651		
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60	Fee Worksheet (SB06)	fee-info.pdf	31100	no	2
			d66f5e29a07ec2df67705e4b0dd8fc755b0f87e1		
Warnings:					
Information:					
Total Files Size (in bytes):			134908601		

This Acknowledgement Receipt evidences receipt on the noted date by the USPTO of the indicated documents, characterized by the applicant, and including page counts, where applicable. It serves as evidence of receipt similar to a Post Card, as described in MPEP 503.

New Applications Under 35 U.S.C. 111

If a new application is being filed and the application includes the necessary components for a filing date (see 37 CFR 1.53(b)-(d) and MPEP 506), a Filing Receipt (37 CFR 1.54) will be issued in due course and the date shown on this Acknowledgement Receipt will establish the filing date of the application.

National Stage of an International Application under 35 U.S.C. 371

If a timely submission to enter the national stage of an international application is compliant with the conditions of 35 U.S.C. 371 and other applicable requirements a Form PCT/DO/EO/903 indicating acceptance of the application as a national stage submission under 35 U.S.C. 371 will be issued in addition to the Filing Receipt, in due course.

New International Application Filed with the USPTO as a Receiving Office

If a new international application is being filed and the international application includes the necessary components for an international filing date (see PCT Article 11 and MPEP 1810), a Notification of the International Application Number and of the International Filing Date (Form PCT/RO/105) will be issued in due course, subject to prescriptions concerning national security, and the date shown on this Acknowledgement Receipt will establish the international filing date of the application.

Electronic Acknowledgement Receipt	
EFS ID:	28346999
Application Number:	13864249
International Application Number:	
Confirmation Number:	9806
Title of Invention:	METHOD AND APPARATUS FOR PLAYING AND PROVIDING AUDIO CONTENT IN A SYNCHRONY GROUP
First Named Inventor/Applicant Name:	Nicholas A. J. Millington
Customer Number:	107361
Filer:	Benjamin Michael Urban
Filer Authorized By:	
Attorney Docket Number:	13-0402 (MBHB 15-518-CON)
Receipt Date:	14-FEB-2017
Filing Date:	17-APR-2013
Time Stamp:	15:50:47
Application Type:	Utility under 35 USC 111(a)

Payment information:

Submitted with Payment	no
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File Listing:

Document Number	Document Description	File Name	File Size(Bytes)/ Message Digest	Multi Part /.zip	Pages (if appl.)
1	Non Patent Literature	Sonos_No14-1330-RGA_DI_207_88pages.pdf	716877 7d274830ede80d5fb9a3605ba8a096cd59484053	no	88

Warnings:

Information:					
2	Non Patent Literature	Sonos_No14-1330-RGA_Markman_Transcript_69pages.pdf	370786	no	69
			7694a1fe17066580b44b12e7e7b9b78215fccde7		
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Information:					
3	Non Patent Literature	Sonos_No14-1330-RGA_DI_214_13pages.pdf	352204	no	13
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Warnings:					
Information:					
4	Non Patent Literature	Sonos_No14-1330-RGA_DI_215_15pages.pdf	567248	no	15
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Warnings:					
Information:					
5	Non Patent Literature	Sonos_No14-1330-RGA_DI_219_24pages.pdf	2776630	no	24
			6e9ad0abcc5d8aa7a33ade6c190ab8385284007e		
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Information:					
6	Non Patent Literature	Sonos_No14-1330-RGA_DI_221_2pages.pdf	239928	no	2
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Warnings:					
Information:					
7	Non Patent Literature	Universal_Plug_and_Play_DM_0298151-72_22pages.pdf	3308815	no	22
			ab6cb4c8625458677ced57470ae5704786bd56055		
Warnings:					
Information:					
8	Non Patent Literature	UPnP_AV_Architechure_SONDM000115483_22pages.pdf	2988946	no	22
			20d7e8d15e1716ba753271e6613e1f05a37b3111		
Warnings:					
Information:					

9	Non Patent Literature	RenderingControl_SONDM000 115187-249_63pages.pdf	3127850	no	63
			407cd629500ac62246e0727a55a4650605c c945f		
Warnings:					
Information:					
10	Non Patent Literature	MediaServer_2002_12pages. pdf	325151	no	12
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Warnings:					
Information:					
11	Non Patent Literature	Universal_Plug_and_Play_Devi ce_UPnP_2000_V1_54pages. pdf	3033111	no	54
			4fc9201973ba3a7a92f039585f4854f1d7e1 4207		
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Information:					
12	Non Patent Literature	MediaRenderer_1_2002_12pag es.pdf	314359	no	12
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Information:					
13	Non Patent Literature	AVTransport_V1_2002_66page s.pdf	438810	no	66
			5b3625001968f78015af1f092e04310895b1 92b3		
Warnings:					
Information:					
14	Non Patent Literature	ConnectionManager_V1_2002_ 25pages.pdf	565998	no	25
			39ffedbddabfa39251516eae6e162e192c62 8c29		
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Information:					
15	Non Patent Literature	ContentDirectory_V1_2002_89 pages.pdf	605416	no	89
			a72140a728367bfc74f14a61790fddcf37cc8 f0a		
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16	Non Patent Literature	WANCommonInterface_DM04 01820-43_24pages.pdf	23867914	no	24
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Information:					
17	Non Patent Literature	WANIPConnection_DM_04018 44-917_74pages.pdf	3357113	no	74
			b015922a376780e63007b30444e0469108876152		
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18	Non Patent Literature	WANPPPConnection_DM_0401 918-2006_89pages.pdf	3959025	no	89
			e6caeab4f8ecf922ed6e2d5d3862bb85ea60123df		
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Information:					
19	Non Patent Literature	IntelMediaRendererDevice_200 2_62pages.pdf	230873	no	62
			93af567dc27e60f1dc2bd2053b9b4b32d02c4f7		
Warnings:					
Information:					
20	Non Patent Literature	IntelDesigning_SONDM000115 117-62_46pages.pdf	21487917	no	46
			3818acfd58a1844f3bdcf875b8fabeb056bead49c		
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Information:					
21	Non Patent Literature	Windows_Media_Connect_200 4_16pages.pdf	612040	no	16
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Warnings:					
Information:					
22	Non Patent Literature	Universal_Plug_and_Play_Plugfest_VendorGuide_2000_7pages.pdf	94631	no	7
			6d33af37a20b364ce9eaaeb13d1b04fc2cad6796		
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23	Non Patent Literature	Intel_SDK_UPnP_Programming_Guide_2002_30pages.pdf	287348	no	30
			9f944b5fb8ff224578ed477d527f670591bcb16e		
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24	Non Patent Literature	Linux_SDK_for_UPnP_2002_101pages.pdf	330243	no	101
			4b9250fccdc7f98de95e3ab422d6b6b393b99145		
Warnings:					
Information:					
25	Non Patent Literature	Designing_a_UPnP_AV_Media_Server_Kidd_SONDM000115062-116_55pages.pdf	23701179	no	55
			3abc025375056fb3500bae8ee39ff77373c9c19d		
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Information:					
26	Non Patent Literature	Simple_Service_Discovery_Protocol_1999_24pages.pdf	58504	no	24
			ac0377e0f99322881679643fdd6b6802aa4d6d4		
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27	Non Patent Literature	General_Event_Notification_Architecture_Base_2000_23pages.pdf	51980	no	23
			906c2b6278a8d07fdeb95cde5d9e83306e6ce288		
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Information:					
28	Non Patent Literature	Microsoft_Universal_Plug_and_Play_DM_0402007-24_18pages.pdf	8990496	no	18
			19c9666824691f55156ab31872552738400d79cf		
Warnings:					
Information:					
29	Non Patent Literature	Home_Networking_DM_0402025-40_16pages.pdf	9816717	no	16
			d16acbc922c0684bfa987a1b5be62ed16365caec		
Warnings:					
Information:					

30	Non Patent Literature	Universal_Plug_and_Play_DM_0402041-73_33pages.pdf	9144769	no	33
			ee301763531b3ea9e00ca230963d5f3cc0d5c2d8		
Warnings:					
Information:					
31	Non Patent Literature	Understanding_Universal_Plug_and_Play_DM_0402074-118_45pages.pdf	9005889	no	45
			431cbe90f9a08914e3cb93fa280a23b2fcd28814		
Warnings:					
Information:					
32	Non Patent Literature	Microsoft_Windows_XP_Reviewers_Guide_DM_0402225-85_61pages.pdf	14113380	no	61
			6de6c34d7ddfe24047ef54d2a2bad98b731a9df4		
Warnings:					
Information:					
33	Non Patent Literature	Simple_Network_Time_Protocol_SNTPI_DM_0397537-46_10pages.pdf	1304386	no	10
			e1bb0ef63b2a1e32160157535e94dde433da0b36		
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Information:					
34	Non Patent Literature	Simple_Network_Time_Protocol_SNTPII_DM_0397663-76_14pages.pdf	1912668	no	14
			71a466552a1f5f7ec535c943864340aad46469a5		
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35	Non Patent Literature	Realtime_Transport_DM_0398235-323_89pages.pdf	18673787	no	89
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36	Non Patent Literature	RealTime_Control_Protocol_DM_0397810-84_75pages.pdf	10808098	no	75
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Information:					

37	Non Patent Literature	Realtime_Streaming_Protocol_DM_0397945-8036_92pages.pdf	11501792	no	92
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38	Non Patent Literature	Network_Time_Protocol_DM_0397417-536_120pages_Part1.pdf	12964521	no	60
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Information:					
39	Non Patent Literature	Network_Time_Protocol_DM_0397417-536_120pages_Part2.pdf	12684304	no	60
			34c64ea88c1b1967f97c23c716a2e7d16654235f		
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40	Non Patent Literature	UPnP_Design_By_Example_DM_0401307-818_511pages.pdf	24678896	no	511
			3b4285e06a903b1990f603e94bbdfa3367356446		
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41	Non Patent Literature	DI_226_Opinion_re_Order_Denying_Inequitable_Conduct_06Feb2017.pdf	602768	no	5
			8e6e6c88bf24bc089ba08eeb4583f08fd404bec		
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42	Non Patent Literature	DI_227_Order_Denying_Inequitable_Conduct_06Feb2017.pdf	81657	no	1
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43	Non Patent Literature	DEFENDANTS_OPPOSITION_TO_SONOS_MOTION_TO_STRIKE_10-3-16-.pdf	241836	no	15
			4a04c692f4446c75e6b4d3fe7df18bd654e99ca5		
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44	Non Patent Literature	Exhibit_A_First_Amended_Answer_to_Third_Amended_Complaint_8-1-16-.pdf	285586	no	26
			077df5a1ace9ed4e15bc140f4cff89a53677a4e1		
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Information:					
45	Non Patent Literature	EXHIBIT_A_DEFENDANTS_SECOND_AMENDED_ANSWER_10-12-2016-.pdf	205207	no	43
			58a1bf313d6cfa29f58406b75452f22e72c918fb		
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46	Non Patent Literature	Exhibit_A_Defendants_Second_Amended_Answer_9-9-16-.pdf	420600	no	88
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47	Non Patent Literature	EXHIBIT_B_DEFENDANTS_SECOND_AMENDED_ANSWER_10-12-2016-.pdf	214180	no	43
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48	Non Patent Literature	OPENING_BRIEF_8-1-16-.pdf	226573	no	11
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49	Non Patent Literature	ORDER_10-7-16-.pdf	74064	no	2
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Information:					
50	Non Patent Literature	OPPOSITION_TO_MOTION_FOR_LEAVE_8-26-16-.pdf	228973	no	25
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51	Non Patent Literature	REDLINED_EXHIBIT_B_FIRST_AMENDED_ANSWER_TO_THIRD_AMENDED_COMPLAINT_8-1-16-.pdf	284998	no	27
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52	Non Patent Literature	REPLY_BRIEF_11-10-2016-.pdf	200331	no	16
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53	Non Patent Literature	REPLY_BRIEF_9-9-2016-.pdf	152851	no	16
			472a37edf435f147843ac9db02deda3a0177c314		
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Information:					
54	Non Patent Literature	SONOS_MOTION_TO_STRIKE_9-15-16-.pdf	164244	no	10
			54ec4c4edde4b0bf036e0e17ebbcfcf3113e7dc		
Warnings:					
Information:					
55	Non Patent Literature	OPPOSITION_TO_MOTION_FOR_LEAVE_10-31-16-.pdf	232927	no	26
			b544490811abd70d4d9ba4527446fe3d8f3dc48d		
Warnings:					
Information:					
Total Files Size (in bytes):			246987394		

This Acknowledgement Receipt evidences receipt on the noted date by the USPTO of the indicated documents, characterized by the applicant, and including page counts, where applicable. It serves as evidence of receipt similar to a Post Card, as described in MPEP 503.

New Applications Under 35 U.S.C. 111

If a new application is being filed and the application includes the necessary components for a filing date (see 37 CFR 1.53(b)-(d) and MPEP 506), a Filing Receipt (37 CFR 1.54) will be issued in due course and the date shown on this Acknowledgement Receipt will establish the filing date of the application.

National Stage of an International Application under 35 U.S.C. 371

If a timely submission to enter the national stage of an international application is compliant with the conditions of 35 U.S.C. 371 and other applicable requirements a Form PCT/DO/EO/903 indicating acceptance of the application as a national stage submission under 35 U.S.C. 371 will be issued in addition to the Filing Receipt, in due course.

New International Application Filed with the USPTO as a Receiving Office

If a new international application is being filed and the international application includes the necessary components for an international filing date (see PCT Article 11 and MPEP 1810), a Notification of the International Application Number and of the International Filing Date (Form PCT/RO/105) will be issued in due course, subject to prescriptions concerning national security, and the date shown on this Acknowledgement Receipt will establish the international filing date of the application.



UNITED STATES PATENT AND TRADEMARK OFFICE

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107361 e 2017-01-24
McDonnell Boehnen Hulbert & Berghoff LLP
Sonos, Inc.
300 South Wacker Drive
Chicago, IL 60606

Paper No.

Application No.:	13/864,249	Date Mailed:	2017-01-24
			PAPER
First Named Inventor:	Nicholas A. J. Millington	Examiner:	SURVILLO, OLEG
Attorney Docket No.:	13-0402 (MBHB 15-518-CON)	Art Unit:	2442
Confirmation No.:	9806	Filing Date:	04/17/2013

Please find attached an Office communication concerning this application or proceeding.

Commissioner for Patents

PTO-90c (Rev.08-06)

Notice of Non-Compliant Amendment (37 CFR 1.121)	Application No. 13/864,249	Applicant(s) MILLINGTON, NICHOLAS A. J.
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-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

The amendment document filed on 30 December, 2016 is considered non-compliant because it has failed to meet the requirements of 37 CFR 1.121 or 1.4. In order for the amendment document to be compliant, correction of the following item(s) is required.

THE FOLLOWING MARKED (X) ITEM(S) CAUSE THE AMENDMENT DOCUMENT TO BE NON-COMPLIANT:

- ☐ 1. Amendments to the specification:
 - ☐ A. Amended paragraph(s) do not include markings.
 - ☐ B. New paragraph(s) should not be underlined.
 - ☐ C. Other _____.
- ☐ 2. Abstract:
 - ☐ A. Not presented on a separate sheet. 37 CFR 1.72.
 - ☐ B. Other _____.
- ☐ 3. Amendments to the drawings:
 - ☐ A. The drawings are not properly identified in the top margin as "Replacement Sheet," "New Sheet," or "Annotated Sheet" as required by 37 CFR 1.121(d).
 - ☐ B. The practice of submitting proposed drawing correction has been eliminated. Replacement drawings showing amended figures, without markings, in compliance with 37 CFR 1.84 are required.
 - ☐ C. Other _____.
- ☒ 4. Amendments to the claims:
 - ☒ A. A complete listing of all of the claims is not present.
 - ☐ B. The listing of claims does not include the text of all pending claims (including withdrawn claims)
 - ☒ C. Each claim has not been provided with the proper status identifier, and as such, the individual status of each claim cannot be identified. Note: the status of every claim must be indicated after its claim number by using one of the following status identifiers: (Original), (Currently amended), (Canceled), (Previously presented), (New), (Not entered), (Withdrawn) and (Withdrawn-currently amended).
 - ☐ D. The claims of this amendment paper have not been presented in ascending numerical order.
 - ☒ E. Other: Missing Claims 1-20.
- ☐ 5. Other (e.g., the amendment is unsigned or not signed in accordance with 37 CFR 1.4): For further explanation of the amendment format required by 37 CFR 1.121, see MPEP § 714.

TIME PERIODS FOR FILING A REPLY TO THIS NOTICE:

1. Applicant is given **no new time period** if the **non-compliant amendment** is an after-final amendment or an amendment filed after allowance, or a drawing submission (only) If applicant wishes to resubmit the non-compliant after-final amendment with corrections, the **entire corrected amendment** must be resubmitted.
2. Applicant is given **two months** from the mail date of this notice to supply the correction, if the non-compliant amendment is one of the following: a preliminary amendment, a non-final amendment (including a submission for a request for continued examination (RCE) under 37 CFR 1.114), a supplemental amendment filed within a suspension period under 37 CFR 1.103(a) or (c), and an amendment filed in response to a Quayle action. If any of above boxes 1 to 4 are checked, the correction required is only the corrected section of the non-compliant amendment in compliance with 37 CFR 1.121.

Extensions of time are available under 37 CFR 1.136(a) only if the non-compliant amendment is a non-final amendment or an amendment filed in response to a *Quayle* action.

Failure to timely respond to this notice will result in:

- Abandonment** of the application if the non-compliant amendment is a non-final amendment or an amendment filed in response to a *Quayle* action; or
- Non-entry** of the amendment if the non-compliant amendment is a preliminary amendment or supplemental amendment.

Legal Instruments Examiner (LIE), if applicable TONI HAKIM Telephone No: (571)272-4353

**IN THE UNITED STATES PATENT AND TRADEMARK OFFICE
(Docket No. 13-0402 (MBHB 15-518-CON))**

In the Application of:	)	
Nicholas A. J. Millington	)	Examiner: SURVILLO, Oleg
	)	
Application No.: 13/864,249	)	Group Art Unit: 2442
	)	
Filed: April 17, 2013	)	Confirmation No. 9806
	)	
Title: System and Method for Synchronizing	)	
Operations Among a Plurality of	)	
Independently Clocked Digital Data	)	
Processing Devices	)	

Mail Stop Amendment
Commissioner for Patents
P.O. Box 1450
Alexandria, VA 22313-1450

**RESPONSE TO THE NOTICE OF NON-COMPLIANT AMENDMENT
MAILED ON JANUARY 24, 2017**

The Notice of Non-Complaint Amendment mailed on January 24, 2017 (the “Notice”), states that Applicant’s Response filed on December 30, 2016 (the “Response”), to the Non-Final Office Action mailed on September 30, 2016, is considered non-compliant because “[a] complete listing of the all the claims is not present.” Notice, p. 1. In particular, the Notice states that the Response filed on December 30, 2016, is missing claims 1-20. Notice, p. 1.

In response to the Notice, Applicant submits herewith a Replacement Response that indicates the status of claims 1-20 as canceled in the “Amendments to the Claims” and in the first paragraph of the “Remarks” section. The Replacement Response submitted herewith is otherwise the same as the Response filed on December 30, 2016.

The Office is hereby authorized to charge any underpayment of fees or credit any overpayment of fees in connection with the prosecution of the above-captioned application to Deposit Account 13-2490.

If further dialog would advance entry and consideration of the Replacement Response submitted herewith, Applicant invites the Office to telephone the undersigned at 312-913-0001.

Respectfully submitted,

**McDONNELL BOEHNEN
HULBERT & BERGHOFF LLP**

Date: January 24, 2017

By: /Jeffrey P. Armstrong/
Jeffrey P. Armstrong
Registration No. 54,967

**IN THE UNITED STATES PATENT AND TRADEMARK OFFICE
(Docket No. 13-0402 (MBHB 15-518-CON))**

In the Application of:	)	
Nicholas A. J. Millington	)	Examiner: SURVILLO, Oleg
	)	
Application No.: 13/864,249	)	Group Art Unit: 2442
	)	
Filed: April 17, 2013	)	Confirmation No. 9806
	)	
Title: System and Method for Synchronizing	)	
Operations Among a Plurality of	)	
Independently Clocked Digital Data	)	
Processing Devices	)	

Mail Stop Amendment
Commissioner for Patents
P.O. Box 1450
Alexandria, VA 22313-1450

**REPLACEMENT RESPONSE TO THE
OFFICE ACTION MAILED SEPTEMBER 30, 2016**

The present communication is a Replacement Response to the Office Action dated September 30, 2016, in the above application, submitted in response to the Notice of Non-Compliant Amendment mailed on January 24, 2016. Amendments to the specification begin on page 2. Amendments to the claims are reflected in the listing of claims beginning on page 3. Remarks begin on page 8.

The Office is hereby authorized to charge any underpayment of fees or credit any overpayment of fees in connection with the prosecution of the above-captioned application to Deposit Account 13-2490.

AMENDMENTS TO THE SPECIFICATION

Please amend the title of the application as follows:

~~System and Method for Synchronizing Operations Among a Plurality of Independently Clocked Digital Data Processing Devices~~Concurrent Transmission and Playback of Audio Information

AMENDMENTS TO THE CLAIMS

Please amend claims 21-40. Following is a completing listing of the claims, as amended.

1-20. (Canceled)

21. (Currently Amended) In a network comprising ~~a plurality of synchrony groups, including at least a first synchrony group and a second synchrony group, wherein a first zone player is a member of the first synchrony group and a second zone player is a member of the second synchrony group~~, a method comprising:

playing first audio information via the first zone player;

transmitting the first audio information to the first zone player via the second zone player over at least one of a local and a wide area network; and

playing second audio information via the second zone player while (i) the first zone player is playing the first audio information and (ii) the second zone player is transmitting the first audio information to the first zone player, ~~by the first zone player while the first zone player is playing a first audio information, a second audio information to the second zone player,~~ wherein the second audio information is different from the first audio information; ~~and~~

playing, by the second synchrony group, the second audio information while the first synchrony group is playing the first audio information.

22. (Currently Amended) The method of claim 21, further comprising~~[[:]]~~ transmitting the first audio information to a third zone player~~[[, by]]~~ while the first-second zone player is:

playing the second audio information; and

transmitting the first audio information to the first zone player, ~~the second audio information to a third zone player, wherein the third zone player is a member of the second synchrony group.~~

23. (Currently Amended) The method of claim ~~[[22]]~~21, further comprising:

transmitting timing information to the first zone player via the second zone player while the second zone player is playing the second audio information, ~~by the first zone player, the~~

~~second audio information to a third zone player, wherein the third zone player is a member of the second synchrony group.~~

24. (Currently Amended) The method of claim ~~[[23,]]~~21 wherein ~~the second audio information comprises playing, by the second synchrony group, the second audio information~~ comprises: utilizing, by the second zone player and the third zone player, the playback timing information to synchronously playback the second audio information.

25. (Currently Amended) The method of claim ~~[[21,]]~~24 wherein the timing information ~~comprises at least one of further comprising: receiving, at the first zone player, the first audio information and playback timing and clock timing information from a fourth zone player; wherein the fourth zone player is a member of the first synchrony group.~~

26. (Currently Amended) The method of claim ~~[[25]]~~ 21, further comprising:
transmitting the first audio information to a third zone player via the second zone player while the second zone player is (i) playing the second audio information and (ii) transmitting the first audio information to the first zone player; and
transmitting timing information associated with the first audio information to the first and third zone players~~utilizing, at the first zone player, the playback timing information received from the fourth zone player to synchronously playback the first audio information with the fourth zone player.~~

27. (Currently Amended) The method of claim ~~[[21,]]~~28 further comprising~~[[:]]~~ receiving the analog audio information, at the first zone player, the second audio information from an audio information source, wherein the from an audio information source that is directly connected to the first zone player.

28. (Currently Amended) The method of claim ~~[[27]]~~ 21, wherein the first audio information comprises digital audio information, and wherein the method further comprises converting analog audio information into the digital audio information via the second zone player ~~second audio information is received at the first zone player as analog audio, and wherein the first zone~~

~~player converts the second audio information from analog to digital prior to transmission of the first audio information to the second-first zone player.~~

29. (Currently Amended) A system comprising a first ~~synchrony group~~zone player and a second ~~synchrony group~~zone player, wherein:

~~[[a]]the first zone player includes at least one processor and tangible non-transitory computer-readable memory comprising instructions that, when executed, cause the first zone player to perform first operations, the first operations comprising~~is a member of the first synchrony group; a second zone player is a member of the second synchrony group; the first synchrony group is arranged to play a playing first audio information; and

~~the first-second zone player includes at least one processor and tangible non-transitory computer-readable memory comprising instructions that, when executed, cause the second zone player to perform second operations, the second operations comprising:~~

~~is arranged to transmit a second~~transmitting the first audio information over at least one of a local and a wide area network to the second-first zone player, and

playing second audio information while (i) the first zone player is playing the first audio information and (ii) the second zone player is transmitting the first audio information to the first zone player, wherein the second audio information is different from the first audio information; and the second synchrony is arranged to play the second audio information while the first synchrony group is playing the first audio information.

30. (Currently Amended) The system of claim 29 wherein the second operations further comprise~~first zone player is further arranged to transmit-transmitting the second-first audio information to a third zone player while the second zone player is:~~

playing the second audio information; and

transmitting the first audio information to the first zone player,~~wherein the third zone player is a member of the second synchrony group.~~

31. (Currently Amended) The system of claim ~~[[30,]]29~~ wherein the second operations further comprise transmitting timing information to the first zone player while the second zone player is playing the second audio information~~is further arranged to transmit playback timing information to the second zone player and the third zone player.~~

32. (Currently Amended) The system of claim ~~[[31,]]~~29 wherein the second audio information comprises timing information~~zone player and the third zone player are arranged to utilize the playback timing information to synchronously playback the second audio information.~~

33. (Currently Amended) The system of claim ~~[[9,]]~~ 32 wherein the timing information comprises at least one of further comprising a fourth zone player, wherein the fourth zone player is a member of the first synchrony group, and wherein the first zone player is further arranged to receive the first audio information and playback timing and clock timing information from the fourth zone player.

34. (Currently Amended) The ~~method~~ system of claim ~~[[25]]~~ 29 wherein ~~the first zone player is arranged to utilize the playback~~ the second operations further comprise:

transmitting the first audio information to a third zone player while the second zone player is (i) playing the second audio information and (ii) transmitting the first audio information to the first zone player; and

transmitting timing information associated with the first audio information to the first and third zone players~~timing information received from the fourth zone player to synchronously playback the first audio information with the fourth zone player.~~

35. (Currently Amended) The system of claim ~~[[29,]]~~ 36 wherein ~~the first zone player is further arranged to receive the second audio information from an audio information source; wherein the~~ second operations further comprise receiving the analog audio information from an audio information source that is directly connected to the ~~first-second~~ zone player.

36. (Currently Amended) The ~~method~~ system of claim 29~~[[27,]]~~ wherein:

the second-first audio information further comprises digital audio information~~is received at the first zone player as analog audio, and wherein the first zone player converts;~~ and

the second audio information from~~operations further comprise converting analog audio information into the digital audio information prior to the transmission of the first audio information to the second-first~~ zone player.

37. (Currently Amended) A tangible, non-transitory computer-readable ~~memory~~ medium having instructions stored thereon that when executed cause a first zone player to:

transmit first audio information to a second zone player over at least one of a local and a wide area network;

playing second audio information while (i) the second zone player is playing the first audio information and (ii) the first zone player is transmitting the first audio information to the second zone player;~~join a first synchrony group with a second zone player; receive a first audio information from the second zone player; play the first audio information in synchrony with the second zone player; and transmit a second audio information to a third zone player while playing the first audio information, wherein the second audio information is different from the first audio information; and, wherein the third zone player is a member of a second synchrony group;~~

transmit the first audio information to a third zone player while the first zone player is (i) playing the second audio information and (ii) transmitting the first audio information to the second zone player.

38. (Currently Amended) The computer-readable medium of claim 37, further comprising instructions that when executed cause the first zone player to:

transmit playback timing information to the second and third zone players while the first zone player is playing the second audio information.

39. (Currently Amended) The computer-readable medium of claim 37, further comprising instructions that when executed cause the first zone player to:

receive the ~~second~~-first audio information from an audio information source, wherein the audio information source is directly connected to the first zone player.

40. (Currently Amended) The computer-readable medium of claim 37, further comprising instructions that when executed cause the first zone player to:

convert the ~~second~~-first audio information from analog to digital prior to transmitting the ~~second~~-first audio information to the first and third zone players.

REMARKS

Claims 21-40 were pending when the present Office Action was mailed. Claims 1-20 were previously canceled. Claims 21-40 have been amended to expedite prosecution, and without conceding the merits of the rejections raised in the Office Action. In view of the above, claims 21-40 are currently pending.

In the Office Action dated September 30, 2016, claim 33 was objected to for informalities and claims 21-40 were rejected. More specifically, claims 29-36 were rejected under 35 U.S.C. § 101 for allegedly being directed to non-statutory subject matter, and claims 21-40 were rejected under 35 U.S.C. § 103 over US 2007/0142944 (Goldberg) in view of US 2005/0159833 (Giaino).

As a preliminary matter, the undersigned attorney thanks the Examiner for engaging in a telephone conference with the undersigned and his colleagues, Chris Butts and John Tolomei, on December 22, 2016, to discuss the present Office Action, the Section 101 and 103 rejections, the Goldberg and Giaino references, and independent claim 29. The following remarks reflect that Applicant understands that the Examiner agreed that (i) Goldberg and Giaino fail to disclose or suggest all the features of independent claim 29, as amended above; and (ii) amended claim 29 is directed to statutory subject matter under Section 101. Applicant requests that this paper constitute the Applicant's Interview Summary. If the Examiner notices any deficiencies with this paper in this regard, he is encouraged to contact the undersigned attorney to correct such deficiencies.

Objection to Claim 33

Claim 33 has been amended to correct the typographical error identified in the Office Action. Accordingly, Applicant requests that the objection to claim 33 be withdrawn.

Section 101 Rejection of claims 29-36

Claims 29-36 were rejected under 35 U.S.C. § 101 for allegedly being directed to non-statutory subject matter. As acknowledged by the Examiner during the December 22 interview, amended independent claim 29 is directed to statutory subject matter. Accordingly, the rejection of claim 29 and the claims dependent therefrom (claims 30-37) should be withdrawn.

Section 103 Rejection of claims 21-40

Claims 21-40 were rejected under 35 U.S.C. § 103 over Goldberg in view of Giaimo. As acknowledged by the Examiner during the December 22 telephone interview, Goldberg and Giaimo, either separately or in combination, fail to disclose or suggest all the features of claim 29. For example, claim 29 is directed to a system that comprises, *inter alia*, “a first zone player” and “a second zone player.” The “first zone player” performs “first operations” that comprise “playing first audio information.” The “second zone player” performs “second operations” that comprise “transmitting the first audio information over at least one of a local and a wide area network to the first zone player” and “playing second audio information while (i) the first zone player is playing the first audio information and (ii) the second zone player is transmitting the first audio information to the first zone player.” The second audio information “is different from the first audio information.”

Goldberg discloses an audio player device having a broadcast unit that communicates to members of a cluster. *See* para. [0263]. Referring to Fig. 33 of Goldberg, Goldberg discloses that “there are three types of communication that are used in the [audio] communication: channel A, which takes place between the members of the original cluster; channel B, which takes place between the members of the original cluster (mediated through the broadcast unit 710) and members of the sub-cluster 701; and channel C, which takes place between the members of the sub-cluster 701. Thus, communications originating from the broadcast unit 710 can be directed either through channel A or channel B, and likewise, communications originating from the broadcast receive unit 770 can be directed either at members only of the sub-cluster 701 through channel C, or to all members of the cluster 700 through both channels C and B, which is then communicated through channel A.” *See* para. [0263].

In contrast to claim 29, neither of the units 710, 770 in Goldberg “transmit[s] [] first audio information to [a] first zone player over at least one of a local and a wide area network” and “play[s] second audio information while (i) the first zone player is playing the first audio information and (ii) the second zone player is transmitting the first audio information” in the manner required by claim 29. At most, the units 710, 770 broadcast and play the same audio that the originating unit is playing. For example, the unit 770 can receive, over channel B, the audio played by unit 710. Unit 710 can also broadcast this same audio over channel A, and unit 770 can re-broadcast this same audio over channel C to receiver unit 730. None of the units 710, 770, however, play audio that is different than the audio that they are broadcasting.

Indeed, it would be contrary to the principle of operation in Goldberg to have the units 710, 770 play audio that is different than the audio they are broadcasting. For example, Goldberg discloses that “when a second individual, User B, perceives the transduced forms produced by User A DJ 200, he can then share the audio signal generated by the audio player 130 of Unit A, by use of the inter-unit transmitter/receiver 110.” *See* Goldberg, para. [0100]. If, however, the User A was playing different audio in the manner alleged in the Office Action, there would simply be no way for User B to “perceive” the audio in the manner intended by Goldberg, which would preclude User B from sharing in the audio played by User A

Giaimo fails to cure the deficiencies of Goldberg. Giaimo discloses processing “[t]wo different and separate audio data streams . . . through a personal computer (PC) system or other computing device so that the separate and distinct audio data streams are heard through separate sound transducers.” *See* Giaimo, abstract. Giaimo discloses that chat messages received over a network during execution of a multiplayer game are processed separately from sounds produced by the multiplayer game, enabling a user to hear the game sounds from speakers, separate and distinct from verbal chat messages, which are heard through earphone(s).” *Id.* Giaimo merely discloses a computer device that can process “different and separate audio data streams.” *Id.* At most, Giaimo discloses playing separate audio streams over different speakers using a computing device; however, Giaimo’s computing device does not play one audio stream while transmitting another over a local or wide area network, much less playing the audio stream while a separate zone player is playing the transmitted audio stream. Accordingly, the rejection of claim 29 and the claims depending therefrom (claims 30-36) should be withdrawn for at least the foregoing reasons, and for the additional features of these claims.

Independent claims 21 and 37 have been amended to include a combination of features generally similar to features of independent claim 29, claims 22-28 depend from base claim 21, and claims 38-40 depend from base claim 37. Therefore, the rejections of claims 21-28 and 37-40 should be withdrawn for at least the reasons discussed above with reference to claim 29, and for the additional features of these claims.

Conclusion

For at least the foregoing reasons, Applicant submits that the claims are in condition for allowance. Applicant thus respectfully requests favorable reconsideration and allowance of the claims. Applicant does not acquiesce in any assertion by the Examiner that is not expressly addressed by these remarks. The Examiner is encouraged to call the undersigned at (312) 913-0001 with any questions or comments for discussion.

Respectfully submitted,

**McDONNELL BOEHNEN
HULBERT & BERGHOFF LLP**

Date: January 24, 2017

By: /Jeffrey P. Armstrong/
Jeffrey P. Armstrong
Registration No. 54,967

Electronic Acknowledgement Receipt	
EFS ID:	28147789
Application Number:	13864249
International Application Number:	
Confirmation Number:	9806
Title of Invention:	METHOD AND APPARATUS FOR PLAYING AND PROVIDING AUDIO CONTENT IN A SYNCHRONY GROUP
First Named Inventor/Applicant Name:	Nicholas A. J. Millington
Customer Number:	107361
Filer:	Jeffrey Armstrong
Filer Authorized By:	
Attorney Docket Number:	13-0402 (MBHB 15-518-CON)
Receipt Date:	24-JAN-2017
Filing Date:	17-APR-2013
Time Stamp:	12:21:55
Application Type:	Utility under 35 USC 111(a)

Payment information:

Submitted with Payment	no
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File Listing:

Document Number	Document Description	File Name	File Size(Bytes)/ Message Digest	Multi Part /.zip	Pages (if appl.)
1		13-0402_Response.pdf	110326	yes	13
			883bd8c9d48db0f9f021272ca8e37ac41d7171ba		

	Multipart Description/PDF files in .zip description		
	Document Description	Start	End
	Amendment/Req. Reconsideration-After Non-Final Reject	1	2
	Supplemental Response or Supplemental Amendment	3	3
	Specification	4	4
	Claims	5	9
	Applicant Arguments/Remarks Made in an Amendment	10	13
Warnings:			
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Total Files Size (in bytes):		110326	
<p>This Acknowledgement Receipt evidences receipt on the noted date by the USPTO of the indicated documents, characterized by the applicant, and including page counts, where applicable. It serves as evidence of receipt similar to a Post Card, as described in MPEP 503.</p> <p><u>New Applications Under 35 U.S.C. 111</u> If a new application is being filed and the application includes the necessary components for a filing date (see 37 CFR 1.53(b)-(d) and MPEP 506), a Filing Receipt (37 CFR 1.54) will be issued in due course and the date shown on this Acknowledgement Receipt will establish the filing date of the application.</p> <p><u>National Stage of an International Application under 35 U.S.C. 371</u> If a timely submission to enter the national stage of an international application is compliant with the conditions of 35 U.S.C. 371 and other applicable requirements a Form PCT/DO/EO/903 indicating acceptance of the application as a national stage submission under 35 U.S.C. 371 will be issued in addition to the Filing Receipt, in due course.</p> <p><u>New International Application Filed with the USPTO as a Receiving Office</u> If a new international application is being filed and the international application includes the necessary components for an international filing date (see PCT Article 11 and MPEP 1810), a Notification of the International Application Number and of the International Filing Date (Form PCT/RO/105) will be issued in due course, subject to prescriptions concerning national security, and the date shown on this Acknowledgement Receipt will establish the international filing date of the application.</p>			

Under the Paperwork Reduction Act of 1995, no persons are required to respond to a collection of information unless it displays a valid OMB control number.

PATENT APPLICATION FEE DETERMINATION RECORD Substitute for Form PTO-875				Application or Docket Number 13/864,249		Filing Date 04/17/2013		☐ To be Mailed	
ENTITY: ☒ LARGE ☐ SMALL ☐ MICRO									
APPLICATION AS FILED – PART I									
(Column 1)		(Column 2)							
FOR		NUMBER FILED		NUMBER EXTRA		RATE (\$)		FEE (\$)	
☐ BASIC FEE (37 CFR 1.16(a), (b), or (c))		N/A		N/A		N/A			
☐ SEARCH FEE (37 CFR 1.16(k), (l), or (m))		N/A		N/A		N/A			
☐ EXAMINATION FEE (37 CFR 1.16(o), (p), or (q))		N/A		N/A		N/A			
TOTAL CLAIMS (37 CFR 1.16(i))		minus 20 =		*		X \$		=	
INDEPENDENT CLAIMS (37 CFR 1.16(h))		minus 3 =		*		X \$		=	
☐ APPLICATION SIZE FEE (37 CFR 1.16(s))		If the specification and drawings exceed 100 sheets of paper, the application size fee due is \$310 (\$155 for small entity) for each additional 50 sheets or fraction thereof. See 35 U.S.C. 41(a)(1)(G) and 37 CFR 1.16(s).							
☐ MULTIPLE DEPENDENT CLAIM PRESENT (37 CFR 1.16(j))									
						TOTAL			
* If the difference in column 1 is less than zero, enter "0" in column 2.									
APPLICATION AS AMENDED – PART II									
(Column 1)		(Column 2)		(Column 3)					
AMENDMENT	01/24/2017	CLAIMS REMAINING AFTER AMENDMENT		HIGHEST NUMBER PREVIOUSLY PAID FOR	PRESENT EXTRA	RATE (\$)		ADDITIONAL FEE (\$)	
	Total (37 CFR 1.16(i))	* 20	Minus	** 20	= 0	x \$80 =		0	
	Independent (37 CFR 1.16(h))	* 3	Minus	***3	= 0	x \$420 =		0	
	☐ Application Size Fee (37 CFR 1.16(s))								
	☐ FIRST PRESENTATION OF MULTIPLE DEPENDENT CLAIM (37 CFR 1.16(j))								
						TOTAL ADD'L FEE		0	
(Column 1)		(Column 2)		(Column 3)					
AMENDMENT		CLAIMS REMAINING AFTER AMENDMENT		HIGHEST NUMBER PREVIOUSLY PAID FOR	PRESENT EXTRA	RATE (\$)		ADDITIONAL FEE (\$)	
	Total (37 CFR 1.16(i))	*	Minus	**	=	X \$		=	
	Independent (37 CFR 1.16(h))	*	Minus	***	=	X \$		=	
	☐ Application Size Fee (37 CFR 1.16(s))								
	☐ FIRST PRESENTATION OF MULTIPLE DEPENDENT CLAIM (37 CFR 1.16(j))								
						TOTAL ADD'L FEE			
<p>* If the entry in column 1 is less than the entry in column 2, write "0" in column 3. ** If the "Highest Number Previously Paid For" IN THIS SPACE is less than 20, enter "20". *** If the "Highest Number Previously Paid For" IN THIS SPACE is less than 3, enter "3". The "Highest Number Previously Paid For" (Total or Independent) is the highest number found in the appropriate box in column 1.</p>									

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This collection of information is required by 37 CFR 1.16. The information is required to obtain or retain a benefit by the public which is to file (and by the USPTO to process) an application. Confidentiality is governed by 35 U.S.C. 122 and 37 CFR 1.14. This collection is estimated to take 12 minutes to complete, including gathering, preparing, and submitting the completed application form to the USPTO. Time will vary depending upon the individual case. Any comments on the amount of time you require to complete this form and/or suggestions for reducing this burden, should be sent to the Chief Information Officer, U.S. Patent and Trademark Office, U.S. Department of Commerce, P.O. Box 1450, Alexandria, VA 22313-1450. DO NOT SEND FEES OR COMPLETED FORMS TO THIS ADDRESS. **SEND TO: Commissioner for Patents, P.O. Box 1450, Alexandria, VA 22313-1450.**
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IN THE UNITED STATES PATENT AND TRADEMARK OFFICE

(Docket No. 13-0402 (MBHB 15-518-CON))

In the Application of:	)	
Nicholas A. J. Millington	)	Examiner: SURVILLO, Oleg
	)	
Application No.: 13/864,249	)	Group Art Unit: 2442
	)	
Filed: April 17, 2013	)	Confirmation No. 9806
	)	
Title: System and Method for Synchronizing	)	
Operations Among a Plurality of	)	
Independently Clocked Digital Data	)	
Processing Devices	)	

Mail Stop Amendment
Commissioner for Patents
P.O. Box 1450
Alexandria, VA 22313-1450

RESPONSE TO THE OFFICE ACTION MAILED SEPTEMBER 30, 2016

The present communication responds to the Office Action dated September 30, 2016, in the above application. Amendments to the specification begin on page 2. Amendments to the claims are reflected in the listing of claims beginning on page 3. Remarks begin on page 8.

The Office is hereby authorized to charge any underpayment of fees or credit any overpayment of fees in connection with the prosecution of the above-captioned application to Deposit Account 13-2490.

AMENDMENTS TO THE SPECIFICATION

Please amend the title of the application as follows:

~~System and Method for Synchronizing Operations Among a Plurality of Independently~~
~~Clocked Digital Data Processing Devices~~Concurrent Transmission and Playback of Audio
Information

AMENDMENTS TO THE CLAIMS

Please amend claims 21-40. Following is a completing listing of the claims, as amended.

21. (Currently Amended) In a network comprising ~~a plurality of synchrony groups, including at least a first synchrony group and a second synchrony group, wherein a first zone player is a member of the first synchrony group and a second zone player is a member of the second synchrony group~~, a method comprising:

playing first audio information via the first zone player;

transmitting the first audio information to the first zone player via the second zone player over at least one of a local and a wide area network; and

playing second audio information via the second zone player while (i) the first zone player is playing the first audio information and (ii) the second zone player is transmitting the first audio information to the first zone player, by the first zone player while the first zone player is playing a first audio information, a second audio information to the second zone player, wherein the second audio information is different from the first audio information; ~~and~~

playing, by the second synchrony group, the second audio information while the first synchrony group is playing the first audio information.

22. (Currently Amended) The method of claim 21, further comprising~~[[:]]~~ transmitting the first audio information to a third zone player~~[[, by]]~~ while the first-second zone player is:

playing the second audio information; and

transmitting the first audio information to the first zone player, the second audio information to a third zone player, wherein the third zone player is a member of the second synchrony group.

23. (Currently Amended) The method of claim ~~[[22]]~~21, further comprising:

transmitting timing information to the first zone player via the second zone player while the second zone player is playing the second audio information, by the first zone player, the second audio information to a third zone player, wherein the third zone player is a member of the second synchrony group.

24. (Currently Amended) The method of claim ~~[[23,]]~~21 wherein ~~the second audio information comprises playing, by the second synchrony group, the second audio information comprises: utilizing, by the second zone player and the third zone player, the playback timing information to synchronously playback the second audio information.~~

25. (Currently Amended) The method of claim ~~[[21,]]~~24 wherein ~~the timing information comprises at least one of further comprising: receiving, at the first zone player, the first audio information and playback timing and clock timing information from a fourth zone player; wherein the fourth zone player is a member of the first synchrony group.~~

26. (Currently Amended) The method of claim ~~[[25]]~~ 21, further comprising:
transmitting the first audio information to a third zone player via the second zone player while the second zone player is (i) playing the second audio information and (ii) transmitting the first audio information to the first zone player; and
transmitting timing information associated with the first audio information to the first and third zone players~~utilizing, at the first zone player, the playback timing information received from the fourth zone player to synchronously playback the first audio information with the fourth zone player.~~

27. (Currently Amended) The method of claim ~~[[21,]]~~28 further comprising~~[[:]]~~ receiving the analog audio information, at the first zone player, the second audio information from an audio information source, wherein the from an audio information source that is directly connected to the first zone player.

28. (Currently Amended) The method of claim ~~[[27]]~~ 21, wherein the first audio information comprises digital audio information, and wherein the method further comprises converting analog audio information into the digital audio information via the second zone player~~second audio information is received at the first zone player as analog audio, and wherein the first zone player converts the second audio information from analog to digital prior to transmission of the first audio information to the second-first zone player.~~

29. (Currently Amended) A system comprising a first ~~synchrony group~~zone player and a second ~~synchrony group~~zone player, wherein:

~~[[a]]the first zone player includes at least one processor and tangible non-transitory computer-readable memory comprising instructions that, when executed, cause the first zone player to perform first operations, the first operations comprising~~is a member of the first synchrony group; a second zone player is a member of the second synchrony group; the first synchrony group is arranged to play a playing first audio information; and

~~the first second zone player includes at least one processor and tangible non-transitory computer-readable memory comprising instructions that, when executed, cause the second zone player to perform second operations, the second operations comprising:~~

~~is arranged to transmit a second~~transmitting the first audio information over at least one of a local and a wide area network to the second-first zone player, and

playing second audio information while (i) the first zone player is playing the first audio information and (ii) the second zone player is transmitting the first audio information to the first zone player, wherein the second audio information is different from the first audio information; and the second synchrony is arranged to play the second audio information while the first synchrony group is playing the first audio information.

30. (Currently Amended) The system of claim 29 wherein the second operations further comprise ~~first zone player is further arranged to transmit~~transmitting the second-first audio information to a third zone player while the second zone player is:

playing the second audio information; and

transmitting the first audio information to the first zone player; wherein the third zone player is a member of the second synchrony group.

31. (Currently Amended) The system of claim ~~[[30,]]29~~ wherein the second operations further comprise transmitting timing information to the first zone player while the second zone player is playing the second audio information~~is further arranged to transmit playback timing information to the second zone player and the third zone player.~~

32. (Currently Amended) The system of claim ~~[[31,]]~~29 wherein the second audio information comprises timing information~~zone player and the third zone player are arranged to utilize the playback timing information to synchronously playback the second audio information.~~

33. (Currently Amended) The system of claim ~~[[9,]]~~ 32 wherein the timing information comprises at least one of ~~further comprising a fourth zone player, wherein the fourth zone player is a member of the first synchrony group, and wherein the first zone player is further arranged to receive the first audio information and playback timing and clock timing information from the fourth zone player.~~

34. (Currently Amended) The ~~method~~ system of claim ~~[[25]]~~ 29 wherein ~~the first zone player is arranged to utilize the playback~~ the second operations further comprise:

transmitting the first audio information to a third zone player while the second zone player is (i) playing the second audio information and (ii) transmitting the first audio information to the first zone player; and

transmitting timing information associated with the first audio information to the first and third zone players~~timing information received from the fourth zone player to synchronously playback the first audio information with the fourth zone player.~~

35. (Currently Amended) The system of claim ~~[[29,]]~~ 36 wherein ~~the first zone player is further arranged to receive the second audio information from an audio information source; wherein the second operations further comprise receiving the analog audio information from an audio information source that is directly connected to the first~~second zone player.

36. (Currently Amended) The ~~method~~ system of claim 29~~[[27,]]~~ wherein:

the second~~first~~ audio information further comprises digital audio information~~is received at the first zone player as analog audio, and wherein the first zone player converts; and~~

the second audio information from~~operations further comprise converting analog audio information into the digital audio information prior to the transmission of the first audio information to the second~~first zone player.

37. (Currently Amended) A tangible, non-transitory computer-readable ~~memory~~ medium having instructions stored thereon that when executed cause a first zone player to:

transmit first audio information to a second zone player over at least one of a local and a wide area network;

playing second audio information while (i) the second zone player is playing the first audio information and (ii) the first zone player is transmitting the first audio information to the second zone player;~~join a first synchrony group with a second zone player; receive a first audio information from the second zone player; play the first audio information in synchrony with the second zone player; and transmit a second audio information to a third zone player while playing the first audio information, wherein the second audio information is different from the first audio information; and, wherein the third zone player is a member of a second synchrony group;~~

transmit the first audio information to a third zone player while the first zone player is (i) playing the second audio information and (ii) transmitting the first audio information to the second zone player.

38. (Currently Amended) The computer-readable medium of claim 37, further comprising instructions that when executed cause the first zone player to:

transmit playback timing information to the second and third zone players while the first zone player is playing the second audio information.

39. (Currently Amended) The computer-readable medium of claim 37, further comprising instructions that when executed cause the first zone player to:

receive the second-first audio information from an audio information source, wherein the audio information source is directly connected to the first zone player.

40. (Currently Amended) The computer-readable medium of claim 37, further comprising instructions that when executed cause the first zone player to:

convert the second-first audio information from analog to digital prior to transmitting the second-first audio information to the first and third zone players.

REMARKS

Claims 21-40 were pending when the present Office Action was mailed. Claims 21-40 have been amended to expedite prosecution, and without conceding the merits of the rejections raised in the Office Action. In view of the above, claims 21-40 are currently pending.

In the Office Action dated September 30, 2016, claim 33 was objected to for informalities and claims 21-40 were rejected. More specifically, claims 29-36 were rejected under 35 U.S.C. § 101 for allegedly being directed to non-statutory subject matter, and claims 21-40 were rejected under 35 U.S.C. § 103 over US 2007/0142944 (Goldberg) in view of US 2005/0159833 (Giaimo).

As a preliminary matter, the undersigned attorney thanks the Examiner for engaging in a telephone conference with the undersigned and his colleagues, Chris Butts and John Tolomei, on December 22, 2016, to discuss the present Office Action, the Section 101 and 103 rejections, the Goldberg and Giaimo references, and independent claim 29. The following remarks reflect that Applicant understands that the Examiner agreed that (i) Goldberg and Giaimo fail to disclose or suggest all the features of independent claim 29, as amended above; and (ii) amended claim 29 is directed to statutory subject matter under Section 101. Applicant requests that this paper constitute the Applicant's Interview Summary. If the Examiner notices any deficiencies with this paper in this regard, he is encouraged to contact the undersigned attorney to correct such deficiencies.

Objection to Claim 33

Claim 33 has been amended to correct the typographical error identified in the Office Action. Accordingly, Applicant requests that the objection to claim 33 be withdrawn.

Section 101 Rejection of claims 29-36

Claims 29-36 were rejected under 35 U.S.C. § 101 for allegedly being directed to non-statutory subject matter. As acknowledged by the Examiner during the December 22 interview, amended independent claim 29 is directed to statutory subject matter. Accordingly, the rejection of claim 29 and the claims dependent therefrom (claims 30-37) should be withdrawn.

Section 103 Rejection of claims 21-40

Claims 21-40 were rejected under 35 U.S.C. § 103 over Goldberg in view of Giaimo. As acknowledged by the Examiner during the December 22 telephone interview, Goldberg and Giaimo, either separately or in combination, fail to disclose or suggest all the features of claim 29. For example, claim 29 is directed to a system that comprises, *inter alia*, “a first zone player” and “a second zone player.” The “first zone player” performs “first operations” that comprise “playing first audio information.” The “second zone player” performs “second operations” that comprise “transmitting the first audio information over at least one of a local and a wide area network to the first zone player” and “playing second audio information while (i) the first zone player is playing the first audio information and (ii) the second zone player is transmitting the first audio information to the first zone player.” The second audio information “is different from the first audio information.”

Goldberg discloses an audio player device having a broadcast unit that communicates to members of a cluster. *See* para. [0263]. Referring to Fig. 33 of Goldberg, Goldberg discloses that “there are three types of communication that are used in the [audio] communication: channel A, which takes place between the members of the original cluster; channel B, which takes place between the members of the original cluster (mediated through the broadcast unit 710) and members of the sub-cluster 701; and channel C, which takes place between the members of the sub-cluster 701. Thus, communications originating from the broadcast unit 710 can be directed either through channel A or channel B, and likewise, communications originating from the broadcast receive unit 770 can be directed either at members only of the sub-cluster 701 through channel C, or to all members of the cluster 700 through both channels C and B, which is then communicated through channel A.” *See* para. [0263].

In contrast to claim 29, neither of the units 710, 770 in Goldberg “transmit[s] [] first audio information to [a] first zone player over at least one of a local and a wide area network” and “play[s] second audio information while (i) the first zone player is playing the first audio information and (ii) the second zone player is transmitting the first audio information” in the manner required by claim 29. At most, the units 710, 770 broadcast and play the same audio that the originating unit is playing. For example, the unit 770 can receive, over channel B, the audio played by unit 710. Unit 710 can also broadcast this same audio over channel A, and unit

770 can re-broadcast this same audio over channel C to receiver unit 730. None of the units 710, 770, however, play audio that is different than the audio that they are broadcasting.

Indeed, it would be contrary to the principle of operation in Goldberg to have the units 710, 770 play audio that is different than the audio they are broadcasting. For example, Goldberg discloses that “when a second individual, User B, perceives the transduced forms produced by User A DJ 200, he can then share the audio signal generated by the audio player 130 of Unit A, by use of the inter-unit transmitter/receiver 110.” *See* Goldberg, para. [0100]. If, however, the User A was playing different audio in the manner alleged in the Office Action, there would simply be no way for User B to “perceive” the audio in the manner intended by Goldberg, which would preclude User B from sharing in the audio played by User A

Giaimo fails to cure the deficiencies of Goldberg. Giaimo discloses processing “[t]wo different and separate audio data streams . . . through a personal computer (PC) system or other computing device so that the separate and distinct audio data streams are heard through separate sound transducers.” *See* Giaimo, abstract. Giaimo discloses that chat messages received over a network during execution of a multiplayer game are processed separately from sounds produced by the multiplayer game, enabling a user to hear the game sounds from speakers, separate and distinct from verbal chat messages, which are heard through earphone(s).” *Id.* Giaimo merely discloses a computer device that can process “different and separate audio data streams.” *Id.* At most, Giaimo discloses playing separate audio streams over different speakers using a computing device; however, Giaimo’s computing device does not play one audio stream while transmitting another over a local or wide area network, much less playing the audio stream while a separate zone player is playing the transmitted audio stream. Accordingly, the rejection of claim 29 and the claims depending therefrom (claims 30-36) should be withdrawn for at least the foregoing reasons, and for the additional features of these claims.

Independent claims 21 and 37 have been amended to include a combination of features generally similar to features of independent claim 29, claims 22-28 depend from base claim 21, and claims 38-40 depend from base claim 37. Therefore, the rejections of claims 21-28 and 37-40 should be withdrawn for at least the reasons discussed above with reference to claim 29, and for the additional features of these claims.

Conclusion

For at least the foregoing reasons, Applicant submits that the claims are in condition for allowance. Applicant thus respectfully requests favorable reconsideration and allowance of the claims. Applicant does not acquiesce in any assertion by the Examiner that is not expressly addressed by these remarks. The Examiner is encouraged to call the undersigned at (312) 913-0001 with any questions or comments for discussion.

Respectfully submitted,
McDONNELL BOEHNEN
HULBERT & BERGHOFF LLP

Date: December 30, 2016

By: /Jeffrey P. Armstrong/
Jeffrey P. Armstrong
Registration No. 54,967

Electronic Acknowledgement Receipt	
EFS ID:	27942911
Application Number:	13864249
International Application Number:	
Confirmation Number:	9806
Title of Invention:	METHOD AND APPARATUS FOR PLAYING AND PROVIDING AUDIO CONTENT IN A SYNCHRONY GROUP
First Named Inventor/Applicant Name:	Nicholas A. J. Millington
Customer Number:	107361
Filer:	Jeffrey Armstrong
Filer Authorized By:	
Attorney Docket Number:	13-0402 (MBHB 15-518-CON)
Receipt Date:	30-DEC-2016
Filing Date:	17-APR-2013
Time Stamp:	16:58:21
Application Type:	Utility under 35 USC 111(a)

Payment information:

Submitted with Payment	no
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File Listing:

Document Number	Document Description	File Name	File Size(Bytes)/ Message Digest	Multi Part /.zip	Pages (if appl.)
1		13-0402_Response.pdf	106684	yes	11
			a56948165ac097e5f4f62712f63c59181fc18faf		

	Multipart Description/PDF files in .zip description		
	Document Description	Start	End
	Amendment/Req. Reconsideration-After Non-Final Reject	1	1
	Specification	2	2
	Claims	3	7
	Applicant Arguments/Remarks Made in an Amendment	8	11
Warnings:			
Information:			
Total Files Size (in bytes):		106684	
<p>This Acknowledgement Receipt evidences receipt on the noted date by the USPTO of the indicated documents, characterized by the applicant, and including page counts, where applicable. It serves as evidence of receipt similar to a Post Card, as described in MPEP 503.</p> <p><u>New Applications Under 35 U.S.C. 111</u> If a new application is being filed and the application includes the necessary components for a filing date (see 37 CFR 1.53(b)-(d) and MPEP 506), a Filing Receipt (37 CFR 1.54) will be issued in due course and the date shown on this Acknowledgement Receipt will establish the filing date of the application.</p> <p><u>National Stage of an International Application under 35 U.S.C. 371</u> If a timely submission to enter the national stage of an international application is compliant with the conditions of 35 U.S.C. 371 and other applicable requirements a Form PCT/DO/EO/903 indicating acceptance of the application as a national stage submission under 35 U.S.C. 371 will be issued in addition to the Filing Receipt, in due course.</p> <p><u>New International Application Filed with the USPTO as a Receiving Office</u> If a new international application is being filed and the international application includes the necessary components for an international filing date (see PCT Article 11 and MPEP 1810), a Notification of the International Application Number and of the International Filing Date (Form PCT/RO/105) will be issued in due course, subject to prescriptions concerning national security, and the date shown on this Acknowledgement Receipt will establish the international filing date of the application.</p>			

Under the Paperwork Reduction Act of 1995, no persons are required to respond to a collection of information unless it displays a valid OMB control number.

PATENT APPLICATION FEE DETERMINATION RECORD Substitute for Form PTO-875				Application or Docket Number 13/864,249		Filing Date 04/17/2013		☐ To be Mailed	
ENTITY: ☒ LARGE ☐ SMALL ☐ MICRO									
APPLICATION AS FILED – PART I									
(Column 1)		(Column 2)							
FOR		NUMBER FILED		NUMBER EXTRA		RATE (\$)		FEE (\$)	
☐ BASIC FEE (37 CFR 1.16(a), (b), or (c))		N/A		N/A		N/A			
☐ SEARCH FEE (37 CFR 1.16(k), (l), or (m))		N/A		N/A		N/A			
☐ EXAMINATION FEE (37 CFR 1.16(o), (p), or (q))		N/A		N/A		N/A			
TOTAL CLAIMS (37 CFR 1.16(i))		minus 20 =		*		X \$		=	
INDEPENDENT CLAIMS (37 CFR 1.16(h))		minus 3 =		*		X \$		=	
☐ APPLICATION SIZE FEE (37 CFR 1.16(s))		If the specification and drawings exceed 100 sheets of paper, the application size fee due is \$310 (\$155 for small entity) for each additional 50 sheets or fraction thereof. See 35 U.S.C. 41(a)(1)(G) and 37 CFR 1.16(s).							
☐ MULTIPLE DEPENDENT CLAIM PRESENT (37 CFR 1.16(j))									
						TOTAL			
* If the difference in column 1 is less than zero, enter "0" in column 2.									
APPLICATION AS AMENDED – PART II									
(Column 1)		(Column 2)		(Column 3)					
AMENDMENT	12/30/2016	CLAIMS REMAINING AFTER AMENDMENT		HIGHEST NUMBER PREVIOUSLY PAID FOR	PRESENT EXTRA	RATE (\$)		ADDITIONAL FEE (\$)	
	Total (37 CFR 1.16(i))	* 20	Minus	** 20	= 0	x \$80 =		0	
	Independent (37 CFR 1.16(h))	* 3	Minus	***3	= 0	x \$420 =		0	
	☐ Application Size Fee (37 CFR 1.16(s))								
	☐ FIRST PRESENTATION OF MULTIPLE DEPENDENT CLAIM (37 CFR 1.16(j))								
						TOTAL ADD'L FEE		0	
(Column 1)		(Column 2)		(Column 3)					
AMENDMENT		CLAIMS REMAINING AFTER AMENDMENT		HIGHEST NUMBER PREVIOUSLY PAID FOR	PRESENT EXTRA	RATE (\$)		ADDITIONAL FEE (\$)	
	Total (37 CFR 1.16(i))	*	Minus	**	=	X \$		=	
	Independent (37 CFR 1.16(h))	*	Minus	***	=	X \$		=	
	☐ Application Size Fee (37 CFR 1.16(s))								
	☐ FIRST PRESENTATION OF MULTIPLE DEPENDENT CLAIM (37 CFR 1.16(j))								
						TOTAL ADD'L FEE			
<p>* If the entry in column 1 is less than the entry in column 2, write "0" in column 3. ** If the "Highest Number Previously Paid For" IN THIS SPACE is less than 20, enter "20". *** If the "Highest Number Previously Paid For" IN THIS SPACE is less than 3, enter "3". The "Highest Number Previously Paid For" (Total or Independent) is the highest number found in the appropriate box in column 1.</p>									

This collection of information is required by 37 CFR 1.16. The information is required to obtain or retain a benefit by the public which is to file (and by the USPTO to process) an application. Confidentiality is governed by 35 U.S.C. 122 and 37 CFR 1.14. This collection is estimated to take 12 minutes to complete, including gathering, preparing, and submitting the completed application form to the USPTO. Time will vary depending upon the individual case. Any comments on the amount of time you require to complete this form and/or suggestions for reducing this burden, should be sent to the Chief Information Officer, U.S. Patent and Trademark Office, U.S. Department of Commerce, P.O. Box 1450, Alexandria, VA 22313-1450. DO NOT SEND FEES OR COMPLETED FORMS TO THIS ADDRESS. **SEND TO: Commissioner for Patents, P.O. Box 1450, Alexandria, VA 22313-1450.**
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APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
13/864,249	04/17/2013	Nicholas A. J. Millington	13-0402 (MBHB 15-518-CON)	9806
107361 7590 12/28/2016 McDonnell Boehnen Hulbert & Berghoff LLP Sonos, Inc. 300 South Wacker Drive Chicago, IL 60606			EXAMINER SURVILLO, OLEG	
			ART UNIT 2442	PAPER NUMBER
			MAIL DATE 12/28/2016	DELIVERY MODE PAPER

Please find below and/or attached an Office communication concerning this application or proceeding.

The time period for reply, if any, is set in the attached communication.

<i>Applicant-Initiated Interview Summary</i>	Application No. 13/864,249	Applicant(s) MILLINGTON, NICHOLAS A. J.	
	Examiner OLEG SURVILLO	Art Unit 2442	

All participants (applicant, applicant's representative, PTO personnel):

(1) Oleg Survillo, Primary Examiner AU 2442. (3) Christopher Butts, Reg. No.: 67,558.

(2) Jeffrey P. Armstrong, Reg. No.: 54,967. (4) John Tolomei, Reg. No. 57,846.

Date of Interview: 22 December 2016.

Type: ☒ Telephonic ☐ Video Conference
 ☐ Personal [copy given to: ☐ applicant ☐ applicant's representative]

Exhibit shown or demonstration conducted: ☐ Yes ☒ No.
 If Yes, brief description: _____.

Issues Discussed ☒ 101 ☐ 112 ☐ 102 ☒ 103 ☒ Others
 (For each of the checked box(es) above, please describe below the issue and detailed description of the discussion)

Claim(s) discussed: 29.

Identification of prior art discussed: Goldberg, Giaimo.

Substance of Interview
 (For each issue discussed, provide a detailed description and indicate if agreement was reached. Some topics may include: identification or clarification of a reference or a portion thereof, claim interpretation, proposed amendments, arguments of any applied references etc...)

Discussed proposed amendments in light of the prior art references used in the rejection. Agreed that proposed changes overcome the combination of Goldberg in view of Giaimo. Discussed making additional changes to link transmission and playback of different audio information. No agreement on exact language has been made. Agreed that proposed amendment overcomes 35 USC 101 rejection. It was agreed that further action will be taken based on Applicant's response to the outstanding Office action.

Applicant recordation instructions: The formal written reply to the last Office action must include the substance of the interview. (See MPEP section 713.04). If a reply to the last Office action has already been filed, applicant is given a non-extendable period of the longer of one month or thirty days from this interview date, or the mailing date of this interview summary form, whichever is later, to file a statement of the substance of the interview

Examiner recordation instructions: Examiners must summarize the substance of any interview of record. A complete and proper recordation of the substance of an interview should include the items listed in MPEP 713.04 for complete and proper recordation including the identification of the general thrust of each argument or issue discussed, a general indication of any other pertinent matters discussed regarding patentability and the general results or outcome of the interview, to include an indication as to whether or not agreement was reached on the issues raised.

☒ Attachment

/OLEG SURVILLO/ Primary Examiner, Art Unit 2442	
--	--

Summary of Record of Interview Requirements

Manual of Patent Examining Procedure (MPEP), Section 713.04, Substance of Interview Must be Made of Record

A complete written statement as to the substance of any face-to-face, video conference, or telephone interview with regard to an application must be made of record in the application whether or not an agreement with the examiner was reached at the interview.

Title 37 Code of Federal Regulations (CFR) § 1.133 Interviews

Paragraph (b)

In every instance where reconsideration is requested in view of an interview with an examiner, a complete written statement of the reasons presented at the interview as warranting favorable action must be filed by the applicant. An interview does not remove the necessity for reply to Office action as specified in §§ 1.111, 1.135. (35 U.S.C. 132)

37 CFR §1.2 Business to be transacted in writing.

All business with the Patent and Trademark Office should be transacted in writing. The personal attendance of applicants or their attorneys or agents at the Patent and Trademark Office is unnecessary. The action of the Patent and Trademark Office will be based exclusively on the written record in the Office. No attention will be paid to any alleged oral promise, stipulation, or understanding in relation to which there is disagreement or doubt.

The action of the Patent and Trademark Office cannot be based exclusively on the written record in the Office if that record is itself incomplete through the failure to record the substance of interviews.

It is the responsibility of the applicant or the attorney or agent to make the substance of an interview of record in the application file, unless the examiner indicates he or she will do so. It is the examiner's responsibility to see that such a record is made and to correct material inaccuracies which bear directly on the question of patentability.

Examiners must complete an Interview Summary Form for each interview held where a matter of substance has been discussed during the interview by checking the appropriate boxes and filling in the blanks. Discussions regarding only procedural matters, directed solely to restriction requirements for which interview recordation is otherwise provided for in Section 812.01 of the Manual of Patent Examining Procedure, or pointing out typographical errors or unreadable script in Office actions or the like, are excluded from the interview recordation procedures below. Where the substance of an interview is completely recorded in an Examiners Amendment, no separate Interview Summary Record is required.

The Interview Summary Form shall be given an appropriate Paper No., placed in the right hand portion of the file, and listed on the "Contents" section of the file wrapper. In a personal interview, a duplicate of the Form is given to the applicant (or attorney or agent) at the conclusion of the interview. In the case of a telephone or video-conference interview, the copy is mailed to the applicant's correspondence address either with or prior to the next official communication. If additional correspondence from the examiner is not likely before an allowance or if other circumstances dictate, the Form should be mailed promptly after the interview rather than with the next official communication.

The Form provides for recordation of the following information:

- Application Number (Series Code and Serial Number)
- Name of applicant
- Name of examiner
- Date of interview
- Type of interview (telephonic, video-conference, or personal)
- Name of participant(s) (applicant, attorney or agent, examiner, other PTO personnel, etc.)
- An indication whether or not an exhibit was shown or a demonstration conducted
- An identification of the specific prior art discussed
- An indication whether an agreement was reached and if so, a description of the general nature of the agreement (may be by attachment of a copy of amendments or claims agreed as being allowable). Note: Agreement as to allowability is tentative and does not restrict further action by the examiner to the contrary.
- The signature of the examiner who conducted the interview (if Form is not an attachment to a signed Office action)

It is desirable that the examiner orally remind the applicant of his or her obligation to record the substance of the interview of each case. It should be noted, however, that the Interview Summary Form will not normally be considered a complete and proper recordation of the interview unless it includes, or is supplemented by the applicant or the examiner to include, all of the applicable items required below concerning the substance of the interview.

A complete and proper recordation of the substance of any interview should include at least the following applicable items:

- 1) A brief description of the nature of any exhibit shown or any demonstration conducted,
- 2) an identification of the claims discussed,
- 3) an identification of the specific prior art discussed,
- 4) an identification of the principal proposed amendments of a substantive nature discussed, unless these are already described on the Interview Summary Form completed by the Examiner,
- 5) a brief identification of the general thrust of the principal arguments presented to the examiner,
(The identification of arguments need not be lengthy or elaborate. A verbatim or highly detailed description of the arguments is not required. The identification of the arguments is sufficient if the general nature or thrust of the principal arguments made to the examiner can be understood in the context of the application file. Of course, the applicant may desire to emphasize and fully describe those arguments which he or she feels were or might be persuasive to the examiner.)
- 6) a general indication of any other pertinent matters discussed, and
- 7) if appropriate, the general results or outcome of the interview unless already described in the Interview Summary Form completed by the examiner.

Examiners are expected to carefully review the applicant's record of the substance of an interview. If the record is not complete and accurate, the examiner will give the applicant an extendable one month time period to correct the record.

Examiner to Check for Accuracy

If the claims are allowable for other reasons of record, the examiner should send a letter setting forth the examiner's version of the statement attributed to him or her. If the record is complete and accurate, the examiner should place the indication, "Interview Record OK" on the paper recording the substance of the interview along with the date and the examiner's initials.



McDonnell Boehnen Hulbert & Berghoff
Law Offices

300 South Wacker Drive
Chicago, Illinois 60606-6709
www.mbhb.com

312-913-0001 phone
312-913-0002 fax

December 20, 2016

Via email: oleg.survillo@uspto.gov

Examiner Oleg Survillo
U.S. Patent & Trademark Office
P.O. Box 1450
Arlington, VA 22313-1450

RE: Interview Agenda for App. 13/864,269 (Sonos 13-0402; MBHB 15-518-CON)

Examiner Survillo:

Thank you for agreeing to interview the above-referenced application on Thursday, December 22, 2016, at 2pm ET / 1pm CT. An agenda of topics is outlined below:

Agenda for the Examiner Interview:

- Proposed amendments to claim 29
- Section 101 rejection of claim 29
- Section 103 rejection of claim 29
 - U.S. Pub. 2007/0142944 ("Goldberg")
 - U.S. Pub. 2005/0159833 ("Giaino")

Example Pending Claim:

29. (Proposed) A system comprising a first ~~synchrony group~~ zone player and a second zone player ~~synchrony group~~, wherein:

~~a the first zone player has a processor and memory comprising instructions that, when executed by the processor, cause the first zone player to perform first operations, the first operations including is a member of the first synchrony group; a second zone player is a member of the second synchrony group; the first synchrony group is arranged to playing~~ a first audio information;

~~the first second zone player has a processor and memory comprising instructions that, when executed by the processor of the second zone player, cause the second audio player to perform second operations, the second operations including:~~

~~is arranged to transmit a second transmitting the first audio information over a data network to the second first zone player, and~~
~~playing second audio information while the first zone player is playing the first audio information, wherein the second audio information is different from the first audio information; and~~
~~the second synchrony is arranged to play the second audio information while the first synchrony group is playing the first audio information.~~

I. Section 101 Rejection of claim 19 (OA, item 6, pgs. 6-7)

Amended claim 29 comprises, *inter alia*, “a first zone player” and “a second zone player” each having a corresponding “processor and memory comprising instructions that, when executed by the processor, cause the [respective] zone player to perform...operations...” Accordingly, the Applicant respectfully submits that proposed claim 29 includes tangible subject matter for at least the reason that the claim 29 includes these features.

II. Section 103 Rejection of claim 29 (OA, item 6, pgs. 6-7)

Figure 1 of the application, which shows an embodiment of the technology, is reproduced and annotated below for purposes of illustration. As shown, the system includes “a first zone player” (annotated as “ZP1”) and “a second zone player” (annotated as “ZP2”). The “first zone player” performs “first operations” that include “playing first audio information.” The “second zone player” performs “second operations” that include (i) “transmitting the first audio information over a data network to the second zone player” and (ii) “playing second audio information while the first zone player is playing the first audio information.”

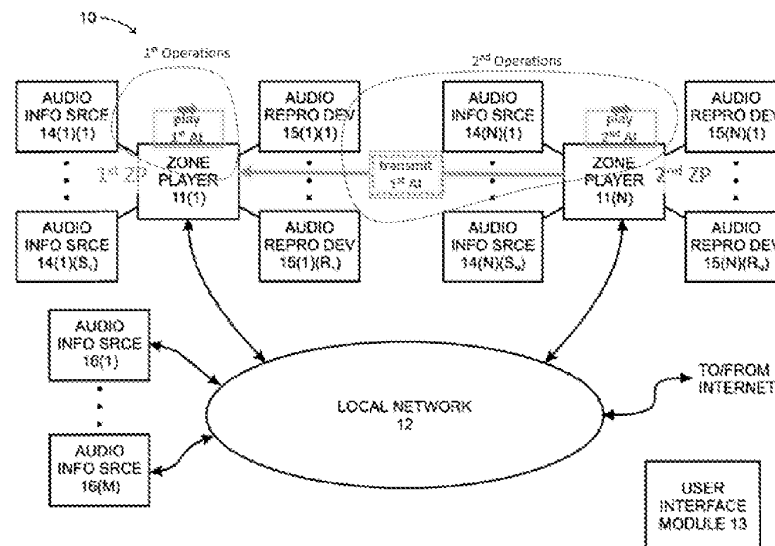


Figure 1 of the Application

Neither Goldberg nor Giaimo, either separately, or in combination disclose or suggest at least the foregoing features of proposed claim 29, much less these features in combination with the other features of claim 29.

Goldberg

Referring to paragraph [0263] and Fig. 33 of Goldberg (reproduced below), Goldberg discloses that "there are three types of communication that are used in the [audio] communication: channel A, which takes place between the members of the original cluster; channel B, which takes place between the members of the original cluster (mediated through the broadcast unit 710) and members of the sub-cluster 701; and channel C, which takes place between the members of the sub-cluster 701. Thus, a communications originating from the broadcast unit 710 can be directed either through channel A or channel B, and likewise, a communications originating from the broadcast receive unit 770 can be directed either at members only of the sub-cluster 701 through channel C, or to all members of the cluster 700 through both channels C and B, which is then communicated through channel A."

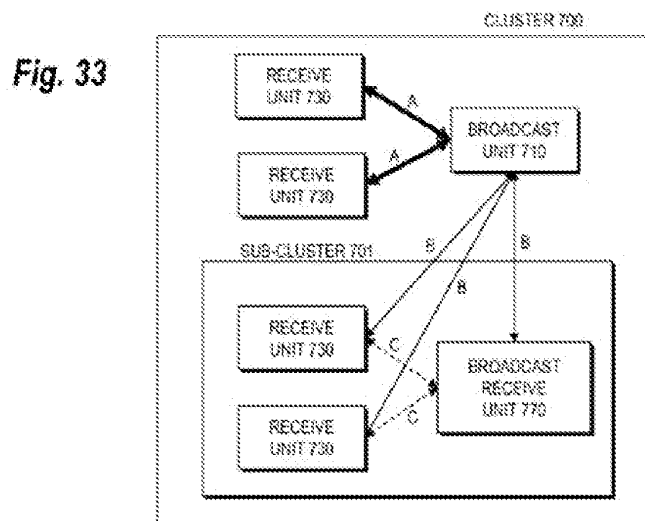


Figure 33 of Goldberg

In contrast to claim 29, neither of the units 710, 770 (i) "transmit[s] [] first audio information to [a] first zone player over a data network" and (ii) "play[s] second audio information while the first zone player is playing the first audio information" in the manner required by claim 29. At most, the units 710, 770 broadcast and play the same audio that the originating unit is playing. For example, the unit 770 can receive, over channel B, the audio played by unit 710. Unit 710 can also broadcast this same audio over channel A, and unit 770 can re-broadcast this same audio over channel C to receiver unit 730. None of the units 710, 770, however, play audio that is different than the audio that they are broadcasting.

It would be contrary to the principle of operation in Goldberg to have the units 710, 770 play audio that is different than the audio they are broadcasting. For example, Goldberg discloses that "when a second individual, User B, perceives the transduced forms produced by User A DJ 200, he can

then share the audio signal generated by the audio player 130 of Unit A, by use of the inter-unit transmitter/receiver 110." See Goldberg, para. [0100]. If, however, the User A was playing different audio in the manner alleged in the Office Action, there would simply be no way for User B to "perceive" the audio in the manner intended by Goldberg, which would preclude User B from sharing in the audio played by User A.

If you have questions, or if you would like additional information to prepare for the interview, please feel free to call me.

Best regards,

A handwritten signature in black ink, appearing to read 'J. Armstrong', with a stylized flourish at the end.

Jeff Armstrong
312-913-2104 (office)

INFORMATION DISCLOSURE STATEMENT BY APPLICANT (Not for submission under 37 CFR 1.99)	Application Number	13864249
	Filing Date	2013-04-17
	First Named Inventor	MILLINGTON; Nicholas A.J.
	Art Unit	2442
	Examiner Name	SURVILLO, Oleg
	Attorney Docket Number	13-0402 (MBHB 15-518-CON)

U.S.PATENTS						Remove
Examiner Initial*	Cite No	Patent Number	Kind Code ¹	Issue Date	Name of Patentee or Applicant of cited Document	Pages, Columns, Lines where Relevant Passages or Relevant Figures Appear
	1	5596696		1997-01-21	TINDELL; James M. et al.	
	2	6917592		2005-07-12	RAMANKUTTY; Rajesh et al.	
	3	7257398		2007-08-14	UKITA; Yoshitaka et al.	
	4	7286652		2007-10-23	AZRIEL; Gad et al.	
	5	7400644		2008-07-15	SAKAMOTO; Ichiro et al.	
	6	7907819		2011-03-15	ANDO; Hideo et al.	
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INFORMATION DISCLOSURE STATEMENT BY APPLICANT (Not for submission under 37 CFR 1.99)	Application Number		13864249
	Filing Date		2013-04-17
	First Named Inventor	MILLINGTON; Nicholas A.J.	
	Art Unit	2442	
	Examiner Name	SURVILLO, Oleg	
	Attorney Docket Number	13-0402 (MBHB 15-518-CON)	

	1	20020131398		2002-09-19	TAYLOR; James L.	
	2	20030172123		2003-09-11	POLAN; Jeffrey S. et al.	
	3	20040032421		2004-02-19	WILLIAMSON; Charles G. et al.	

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FOREIGN PATENT DOCUMENTS

Examiner Initial*	Cite No	Foreign Document Number ³	Country Code ²ⁱ	Kind Code ⁴	Publication Date	Name of Patentee or Applicant of cited Document	Pages, Columns, Lines where Relevant Passages or Relevant Figures Appear	T ⁵
	1	2001034951	JP		2001-02-09	SANYO ELECTRIC CO.	English Translation of Abstract Only	×

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NON-PATENT LITERATURE DOCUMENTS

Examiner Initials*	Cite No	Include name of the author (in CAPITAL LETTERS), title of the article (when appropriate), title of the item (book, magazine, journal, serial, symposium, catalog, etc), date, pages(s), volume-issue number(s), publisher, city and/or country where published.	T ⁵
	1	EUROPEAN PATENT OFFICE, Examination Report mailed on 24 October 2016, issued in connection with European Patent Application No. 13808623.6, 4 pages.	
	2	"Final Office Action mailed on 19 October 2016, issued in connection with U.S. Patent Application No. 13/848921, filed on 22 March 2013, 14 pages".	
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9	Non-Final Office Action mailed on 21 September 2016, issued in connection with U.S. Patent Application No. 15/080591, filed on 25 March 2016, 9 pages.
10	Non-Final Office Action mailed on 21 September 2016, issued in connection with U.S. Patent Application No. 15/080716, filed on 25 March 2016, 8 pages.
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	Examiner Name	SURVILLO, Oleg	
	Attorney Docket Number	13-0402 (MBHB 15-518-CON)	

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Examiner Signature		Date Considered	
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¹ See Kind Codes of USPTO Patent Documents at www.USPTO.GOV or MPEP 901.04. ² Enter office that issued the document, by the two-letter code (WIPO Standard ST.3). ³ For Japanese patent documents, the indication of the year of the reign of the Emperor must precede the serial number of the patent document. ⁴ Kind of document by the appropriate symbols as indicated on the document under WIPO Standard ST.16 if possible. ⁵ Applicant is to place a check mark here if English language translation is attached.

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<p>☒ The fee set forth in 37 CFR 1.17 (p) has been submitted herewith.</p>			
<p>A certification statement is not submitted herewith.</p>			
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A signature of the applicant or representative is required in accordance with CFR 1.33, 10.18. Please see CFR 1.4(d) for the form of the signature.			
Signature	/Benjamin M. Urban/	Date (YYYY-MM-DD)	2016-11-22
Name/Print	Benjamin M. Urban	Registration Number	71,591
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OPTICAL DISC AND OPTICAL DISC RECORDING PIAYBACK APPARATUS

Inventor(s): ASANO KENJI; TORASAWA KENJI; MAMIYA NOBORU; UCHIHARA KAJI; WASHIMI SATOSHI; NAKAO KENJI; HIOKI TOSHIKI; MATSUYAMA HISASHI; HORI YOSHIHIRO ± (ASANO KENJI, ; TORASAWA KENJI, ; MAMIYA NOBORU, ; UCHIHARA KAJI, ; WASHIMI SATOSHI, ; NAKAO KENJI, ; HIOKI TOSHIKI, ; MATSUYAMA HISASHI, ; HORI YOSHIHIRO)

Applicant(s): SANYO ELECTRIC CO ± (SANYO ELECTRIC CO LTD)

Classification: - international: **G11B11/105; G11B20/12; G11B20/18; G11B7/004; G11B7/007; G11B7/24;** (IPC1-7): G11B11/105; G11B20/12; G11B20/18; G11B7/007; G11B7/24

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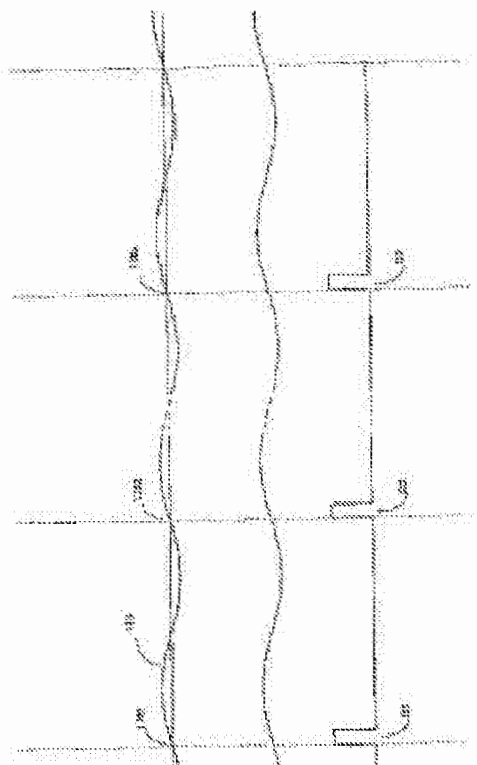
Application number: JP20000205803 20000706 Global Dossier

Priority number(s): JP19960255066 19960926 ; JP19960268893 19961009 ; JP19960278385 19961021 ; JP19960301426 19961113 ; JP19970006988 19970117 ; JP19970012790 19970127 ; JP19970025655 19970207 ; JP19970056681 19970311 ; JP19970095700 19970414 ; JP19970106368 19970423 ; JP19970109436 19970425 ; JP19970140087 19970529

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Abstract of JP2001034951 (A)

PROBLEM TO BE SOLVED: To enable the read of address information recorded as waveforms of wobbles provided at grooves of a disc for recording signals on lands and grooves even at the record playback time on not only the grooves but lands, using only one laser beam. SOLUTION: A detected wobble waveform 175 intersects time points 1761, 1762,..., 176n ascendingly in relation to a base axis, synchronizing signals D1, D2,..., Dn are generated at the points 1761, 1762,..., 176n, and hence synchronizing signals for playback signals can be generated, without forming special wobbles different from those for forming fine clock marks for generating the synchronizing signals in the tracks.



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	5 2 1		5 2 1 F
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(62) 分割の表示	特願平9-147793の分割		三洋電機株式会社
(22) 出願日	平成9年6月5日(1997.6.5)		大阪府守口市京阪本通2丁目5番5号
(31) 優先権主張番号	特願平8-255066	(72) 発明者	浅野 賢二
(32) 優先日	平成8年9月26日(1996.9.26)		大阪府守口市京阪本通2丁目5番5号 三洋電機株式会社内
(33) 優先権主張国	日本 (J P)	(72) 発明者	虎沢 研示
(31) 優先権主張番号	特願平8-268893		大阪府守口市京阪本通2丁目5番5号 三洋電機株式会社内
(32) 優先日	平成8年10月9日(1996.10.9)	(74) 代理人	100111383
(33) 優先権主張国	日本 (J P)		弁理士 芝野 正雅
(31) 優先権主張番号	特願平8-278385		
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(33) 優先権主張国	日本 (J P)		

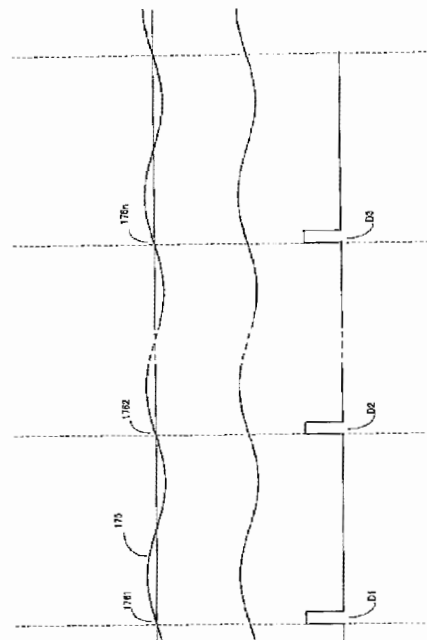
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(54) 【発明の名称】 光ディスク及び光ディスク記録再生装置

(57) 【要約】

【課題】 ランドとグルーブに信号を記録するディスクにおいてグルーブに設けられたウォブルの波形として記録されたアドレス情報をグルーブのみならず、ランドの記録再生時にも1つのレーザビームのみにより読み取ることを可能とする。

【解決手段】 検出したウォブル波形175が基軸との関係において下から上に交差する各時点1761、1762、・・・176nで同期信号D1、D2、・・・Dnを発生するので、同期信号を生成するファインクロックマークをトラックに形成するウォブルとは異なる特別のウォブルで形成しなくても再生信号の同期信号を生成することができる。



【特許請求の範囲】

【請求項1】 グループの少なくとも一方の壁をグループの幅方向をウォブリングさせてアドレス情報ブロック及びアドレス情報ブロックを形成した光ディスクにおいて、

前記アドレス情報ブロックは第1のウォブルにより形成され、

前記データ情報ブロックは前記第1のウォブルの波長より長い波長の第2のウォブルにより形成されていることを特徴とする光ディスク。

【請求項2】 ランドとグループとに記録または／および再生可能な光ディスクにおいて、

前記グループに固定波長のウォブルを連続的に形成し、前記グループの一方の壁に、アドレス情報により変調されたウォブルを所定間隔で間欠的に形成したことを特徴とする光ディスク。

【請求項3】 前記データ情報ブロックのウォブリング周期は $0.8 \sim 1.0 \mu\text{m}$ の範囲であり、前記データ情報ブロックのウォブリングの振幅は $5 \sim 25 \text{ nm}$ の範囲であることを特徴とする請求項1乃至請求項2記載の光ディスク。

【請求項4】 さらに、再生信号の洩れ込みを消去する補正量に関する情報を含むTOC領域を内周部または／および外周部に設けたことを特徴とする請求項1乃至請求項3記載の光ディスク。

【請求項5】 さらに、光磁気信号を消去した再生信号と等しくなる所定フォーマットの信号を記録した特定領域と、前記特定領域に続いて信号を記録した信号領域とを組として多数設けたことを特徴とする請求項1乃至請求項3記載の光ディスク。

【請求項6】 前記請求項1乃至請求項5記載の光ディスクを記録再生する光ディスク装置であって、1つのレーザビームを光ディスクに導く光学手段と、前記第2のウォブルを検出する検出手段と、前記検出手段により検出されたウォブル波形が下から上に基軸を交差する時点を検出して再生信号のための同期信号を生成する同期信号生成回路と、を備えたことを特徴とする光ディスク装置。

【請求項7】 前記請求項4記載の光ディスクを記録再生する光ディスク装置であって、光学手段により再生された前記補正量に基づいて前記グループの壁に設けられたウォブルの位相と振幅とを補正した補正信号を発生する補正信号発生回路と、前記補正信号発生回路により発生された補正信号を再生信号から減じる減算器と、を備えたことを特徴とする光ディスク装置。

【請求項8】 前記請求項4記載の光ディスクを記録再生する光ディスク装置であって、光学手段により再生された前記補正量に基づいて変化する補正量の範囲を決定する補正量発生回路と、

前記補正信号発生回路により決定された各補正量を再生信号から減じる減算器と、

前記減算器からの再生信号を入力して、各補正量に対するエラーレートを検出し、更に、エラーレートが最小となる補正量に対する再生信号を検出するエラーレート検出回路と、を備えたことを特徴とする光ディスク装置。

【請求項9】 前記請求項5記載の光ディスクを記録再生する光ディスク装置であって、

前記特定領域に記録された所定フォーマットの信号を再生した再生信号をメモリする波形メモリと、

再生信号を一方の端子に入力し、前記再生信号の入力に同期して他方の端子に前記波形メモリから前記所定フォーマットの信号の再生信号を入力して前記再生信号から前記所定フォーマットの信号の再生信号を減じる減算器と、を備えたことを特徴とする光ディスク装置。

【請求項10】 グループをウォブリングさせてアドレス情報ブロック及びデータ情報ブロックを形成した光ディスクを記録再生する光ディスク記録再生装置において、

再生信号をA/DするA/D変換器と、

前記A/D変換器からの信号を入力して前記グループに設けられたウォブルの1波長に相当する信号を検出する同期検波回路と、

前記同期検波回路からの信号を所定回数加算して再生信号を平均化する加算器と、

前記加算器により平均化された再生信号をメモリする波形メモリと、

前記A/D変換器からの再生信号と前記波形メモリからの平均化された再生信号とを入力して、再生信号から平均化された再生信号を減じる減算器と、を備えたことを特徴とする光ディスク装置。

【発明の詳細な説明】

【0001】

【従来の技術】

【0002】

【発明の属する技術分野】本発明は、波形のプリグループを有する光ディスク、特に、記録可能なCDディスクや記録可能光磁気ディスクの記録再生装置に好適な光ディスクとその記録再生装置に関する。ミニ・ディスクでは、回転同期信号となる搬送波をアドレス情報でFM変調した信号の波形形状のグループが刻まれており、回転制御とアドレス情報の検出に利用されている。この方法は一般に、ウォブリング法と呼ばれるものである。

【0003】また、回転同期信号となる搬送波をアドレス情報でFM変調した信号の波形形状のグループが刻まれている光ディスクにおいてグループとグループ間のランドとに記録が行われる光ディスク装置においても、ウォブリング法が提案されている。この場合の、アドレス情報の読み出し方法としては、光ピックアップのレーザ光源から発光されたレーザビームを回折格子によってメ

インビームおよび2つのサブビームの3ビームに分け、図38に示すように、メインビームをグループ中心にトラッキング制御しているときは、メインビームからのプッシュプル信号よりアドレス情報を読み出し、メインビームをランド中心に制御しているときは、サブビームからのプッシュプル信号よりアドレス情報を読み出すもの(特開平7-14172)が知られている。

【0004】

【発明が解決しようとする課題】上記した従来の光ディスクは、グループとランドに記録するためにトラックの全長はグループのみに記録するものの2倍となる。然し乍ら、グループのみにアドレス情報があるという理由から、光ピックアップのレーザ光源から発光されたレーザビームをメインビームおよび2つのサブビームの3ビームに分けるための回折格子等の光学部品が必要となる。また、レーザ光源から発光されたレーザビームのパワーを有効に使えないという欠点があった。特に、記録可能な光ディスクの場合、レーザビームのパワーの利用効率は、大きな課題となる。

【0005】この発明は上記した点に鑑みてなされたものであって、その目的とするところは、グループの波形として記録されたアドレス情報をグループのみならず、ランドの記録再生時にも1つのレーザビームのみにより読み取ることを可能とするもので、光学部品の数を減らすとともに、レーザ光源から発光されたレーザビームのパワーを損なうことなく、ディスクの記録密度を十分に高めることが可能な光ディスク記録再生装置を提供することにある。

【0006】

【課題を解決するための手段】上記の課題を解決するため、本発明は、グループの少なくとも一方の壁をグループの幅方向をウォブリングさせてアドレス情報ブロック及びアドレス情報ブロックを形成した光ディスクにおいて、前記アドレス情報ブロックは第1のウォブルにより形成され、前記データ情報ブロックは前記第1のウォブルの波長より長い波長の第2のウォブルにより形成されていることを特徴とする。

【0007】また、本発明は、ランドとグループとに記録または／および再生可能な光ディスクにおいて、前記グループに固定波長のウォブルを連続的に形成し、前記グループの一方の壁に、アドレス情報により変調されたウォブルを所定間隔で間欠的に形成したことを特徴とする。また、本発明においては、前記データ情報ブロックのウォブリング周期は0.8〜10 μ mの範囲であり、前記データ情報ブロックのウォブリングの振幅は5〜25nmの範囲であることを特徴とする。

【0008】また、本発明においては、再生信号の洩れ込みを消去する補正量に関する情報を含むTOC領域を内周部または／および外周部に設けたことを特徴とする。また、本発明においては、光磁気信号を消去した再

生信号と等しくなる所定フォーマットの信号を記録した特定領域と、前記特定領域に続いて信号を記録した信号領域とを組として多数設けたことを特徴とする。

【0009】また、本発明の光ディスク装置は、1つのレーザビームを光ディスクに導く光学手段と、前記第2のウォブルを検出する検出手段と、前記検出手段により検出されたウォブル波形が下から上に基軸を交差する時点を検出して再生信号のための同期信号を生成する同期信号生成回路とを備えたことを特徴とする。また、本発明の光ディスク装置は、光学手段により再生された前記補正量に基づいて前記グループの壁に設けられたウォブルの位相と振幅とを補正した補正信号を発生する補正信号発生回路と、前記補正信号発生回路により発生された補正信号を再生信号から減じる減算器とを備えたことを特徴とする。

【0010】また、本発明の光ディスク装置は、光学手段により再生された前記補正量に基づいて変化させる補正量の範囲を決定する補正量発生回路と、前記補正信号発生回路により決定された各補正量を再生信号から減じる減算器と、前記減算器からの再生信号を入力して、各補正量に対するエラーレートを検出し、更に、エラーレートが最小となる補正量に対する再生信号を検出するエラーレート検出回路とを備えたことを特徴とする。

【0011】また、本発明の光ディスク装置は、前記特定領域に記録された所定フォーマットの信号を再生した再生信号をメモリする波形メモリと、再生信号を一方の端子に入力し、前記再生信号の入力に同期して他方の端子に前記波形メモリから前記所定フォーマットの信号の再生信号を入力して前記再生信号から前記所定フォーマットの信号の再生信号を減じる減算器とを備えたことを特徴とする。

【0012】また、本発明は、グループをウォブリングさせてアドレス情報ブロック及びデータ情報ブロックを形成した光ディスクを記録再生する光ディスク記録再生装置において、再生信号をA/DするA/D変換器と、前記A/D変換器からの信号を入力して前記グループに設けられたウォブルの1波長に相当する信号を検出する同期検波回路と、前記同期検波回路からの信号を所定回数加算して再生信号を平均化する加算器と、前記加算器により平均化された再生信号をメモリする波形メモリと、前記A/D変換器からの再生信号と前記波形メモリからの平均化された再生信号とを入力して、再生信号から平均化された再生信号を減じる減算器とを備えたことを特徴とする。

【0013】

【発明の実施の形態】以下、図面を参照しつつ本発明の実施形態について詳述する。

<第1の実施形態>図1を参照して、光ディスクは、ランドとグループ用のアドレス部170をウォブル172とウォブル173で形成し、アドレス部170に続くデ

ータ領域171には、ウォブル174が形成されている光ディスクである。ここで、前記ウォブル172と前記ウォブル173とは異なる波形で形成される。前記ウォブル172、173、および174はグループの両側の壁に形成されている。図1に示す光ディスクでは、アドレス部170には異なるアドレスが2つ記録されており、前記ウォブル172はランド用のアドレスであり、前記ウォブル173はグループ用のアドレスである。再生時には、レーザビームが前記ウォブル172、もしくは前記ウォブル173を再生してランド、もしくはグループのアドレスを検出する。その後、レーザビームがデータ領域171においてデータを再生するとともに前記ウォブル174を再生し、検出したウォブルの波形から再生信号の同期信号を生成する。即ち、図2を参照して、検出したウォブル波形175が基軸との関係において下から上に交差する各時点1761、1762、・・・176nで同期信号D1、D2、・・・Dnを発生する。従って、図1に示す光ディスクでは、同期信号を生成するファインクロックマークをトラックに形成するウォブルとは異なる特別のウォブルで形成しなくても再生信号の同期信号を生成することができるという特徴を有する。本発明における光ディスクにおいては、同期信号を生成する前記各時点1761、1762、・・・1762nは、20～30 μ mの範囲となるようにウォブルの周波数が設定されている。前記各時点1761、1762、・・・176nの各間隔を20 μ mに、前記ウォブル174の振幅を30、40、60nmと変えて再生信号のCN比とジッタを測定した結果、図3に示す如く、前記ウォブル174の振幅が30、40、60nmと大きくなるにつれて再生信号のCN比が向上し、ジッタが減少する。前記各時点1761、1762、・・・176nの各間隔を25、30 μ mと変化させた場合にも同様の結果が得られた。

【0014】また、本発明における光ディスクは、図4に示すものでもよい。図4に示す光ディスクのアドレス部は図1における前記ウォブル174にアドレスを示すウォブル200を重畳形成したものである。図1に示す光ディスクの前記アドレス部170には2つのアドレスが記録されていたが、図4に示す光ディスクではアドレス部170には1つのアドレスがウォブル200で形成されており、前記ウォブル200はグループの一方の壁のみに形成されている。この光ディスクでは、レーザビームでウォブル200を再生し、検出されたアドレスは両隣のランド、およびグループ用のアドレスとして用いられる。また、この光ディスクにおいても、再生信号のための同期信号は図1の光ディスクと同様にして生成され、同期信号を生成するためのデータ領域に設けられたウォブルの周波数も同じである。

【0015】また更に、本発明における光ディスクは図5に示す光ディスクであってもよい。図5に示す光ディ

スクでは、アドレス部170にはデータ領域に形成される前記ウォブル174は形成されず、ウォブル210のみがグループの一方のみに形成される。この光ディスクにおいても、ウォブル210をレーザビームにより再生することにより検出されるアドレスは、両隣のランド、およびグループ用のアドレスとして用いられる。また、この光ディスクにおいても、再生信号のための同期信号は図1の光ディスクと同様にして生成され、同期信号を生成するためのデータ領域に設けられたウォブルの周波数も同じである。

【0016】また更に、本発明における光ディスクは、図6、および図7に示す光ディスクであってもよい。図6、および図7に示す光ディスクは、図5に示す光ディスクにおいて、データ部に形成される前記ウォブル174がグループの一方の壁に形成された構造のものである。また、これらの光ディスクにおいても、再生信号のための同期信号は図1の光ディスクと同様にして生成され、同期信号を生成するためのデータ領域に設けられたウォブルの周波数も同じである。

【0017】図8を参照して、上記図1、図4、図5、図6、および図7に示した光ディスクの再生装置について説明する。4分割センサー240の各領域A、B、C、およびDで検出された信号は、AとDの和[A+D]とBとCの和[B+C]が差動増幅器241に導入され、[A+D]－[B+C]が演算される。その後、[A+D]－[B+C]はトラッキングエラー信号を生成するために高域をカットするローパスフィルタ242、ウォブル検出用のバンドパスフィルタ243、アドレス復調用のバンドパスフィルタ244に送られる。前記バンドパスフィルタ244では、前記アドレス部に設けられたウォブルを検出して得られた再生信号からアドレスを復調する。また、前記バンドパスフィルタ243では、再生信号のうち、高域、低域をカットしてコンパレータ245に送り、コンパレータ245では、前記データ領域に設けられたウォブル174を検出して得られたウォブル波形が、下から上に基軸と交差する時点を検出して、その結果をPLL回路246に送る。PLL回路246は、送られてきた時点を基に再生信号のための同期信号を生成する。生成された同期信号は、ディスク回転制御回路やデータクロック生成回路に送られる。

【0018】図9を参照して、前記アドレス部170と前記データ領域174を含むディスクの詳細について説明する。ディスクの1トラック（一周）はNf個のフレームに分けられる。一フレームは2720バイトの大きさで表され、96バイトのアドレス部と、2624バイトのデータ部に分割される。光磁気信号は、主にデータ部に、NRZ I変調や(1-7)変調を使用して記録、もしくは再生される。この場合、記録されるデータのビット密度を0.22 μ m/bitとすると一フレーム当たりの長さは、4.7872mm、0.20 μ m/bit

とすると4.352mmとなる。従って、コンパクトディスク(CD)と同じ12cmサイズのディスクの場合、トラック一周当たりのフレーム数Nfは、30から87程度となる。次に、アドレス部は96バイトの長さを持ち、アドレス部の最小の一ウォブル周期を一バイトとすると、一ウォブル周期のディスク上の長さは、1.60~1.76 μ mの範囲となる。また、プリアンプル1(PA1)、プリアンプル2(PA2)に各々4バイト、アドレス1(Address1)、アドレス2(Address2)に各々42バイト、アドレスマーク(AM)に2バイト、プリアンプル3(PA3)とspaceに各々1バイトのディスク上の長さを与える。この場合、実際のデータ長としては、図10に示すように、PA1、PA2は4ビット、Address1、Address2は42ビット、AMは2ビット、PA3、spaceは1ビットが割与えられる。

【0019】更に、次に、データ領域は2624バイトの長さを持ち、PA4が24バイト、データエリアが2592バイト、PA5が8バイトから構成される。データエリアの2592バイトは、ユーザデータが2048バイト、記録信号のDC成分抑圧用データが32バイト、あるいはエラー訂正のためのデータ等から構成される。この場合、データの記録や再生を行うための同期信号を生成するためのウォブルの一周期に16バイトの長さを与えると、ディスク上の一ウォブルの長さは、ビット密度が0.22 μ m/bitの場合は28.16 μ m、0.20 μ m/bitの場合は、25.6 μ mとなる。そして、一フレームの中のデータ部には164個のウォブルが存在することになる。従って、一トラックの中に60個のフレームが存在し、ディスクが1500rpmで回転したとすると、ウォブルの周波数は、255kHzとなる。このウォブルの周波数を利用してデータを記録または/および再生するためのデータ同期信号をPLLによって生成する。これは、図11に示すように、データ変調方式としてNRZIを使用した場合、データ同期信号は、32.64MHzとなり、PLLの分周器271の比率は128分の一となる。また、一ウォブルの長さは、16バイトに限らず、一定のバイト相当の長さ、例えば、4バイト、あるいは8バイト、あるいは20バイトという長さを与えることができる。この場合、ウォブルの周波数が先の255kHzに比べて早く、あるいは遅く設定され、同期信号を生成するPLLの周波数が異なることになる。本実施の形態では、一ウォブルの周期長が5~50 μ mの範囲の値が好ましい。

【0020】ウォブル信号のSignal to Noise Ratioを確保するためには、ウォブルの振幅量が大い方がよい。一方、振幅量が大いとき、光磁気ディスクの場合、光磁気信号にウォブル周波数が漏れ込み、悪影響を与える。図12は、ウォブル信号のクロストーク量と光磁気ディスク信号のエラーレートを測定

した結果である。良好なエラーレート特性を得るためには、ウォブル信号のクロストーク量を-25dB以下にする必要がある。一方、図13は、ウォブルの振幅量と、光磁気信号へのクロストーク量を調べた例である。グループとランドの幅の比率が概ね1対1で、グループのピッチが1.0~1.28 μ mの場合、ウォブル信号のSignal to Noise Ratioを確保するとともに、光磁気ディスクの信号を精度良く再生するためには、ウォブルに $\pm 10 \sim \pm 60$ nmの振幅量を与えることが必要である。特に、ビット密度が0.15~0.24 μ m/bitで、1ウォブル長が10~32 μ mの場合、ウォブルの振幅量が $\pm 10 \sim \pm 40$ nmとなる値が好ましい。

【0021】相変化ディスク、色素系あるいは金属系の追記型の光ディスクにおいては、一ウォブル長が5~50 μ mの範囲、振幅量が $\pm 10 \sim \pm 60$ nmの範囲の値が好ましい。図10を参照して、Address1、Address2には、各々42ビットが割与えられ、Frame addressはトラック一周の中で順を付けた番号である。従って、フォーマット上では、一トラックに最大256のフレーム数(Nf)を確保することができる。Track addressは、ディスク全体のトラックを内周、もしくは外周から付けた通し番号である。従って、フォーマット上では、最大65536本のトラック数を確保することができる。PA1、PA2、およびPA3はAddress1、Address2やアドレスマークを、正確に検出するためのプリアンプルやポストアンプルとして使用される。これらの信号を記録するために使用されるウォブルの振幅は、ほぼ同じ量を与える。グループとランドの幅の比率が概ね1対1で、グループのピッチが1.0~1.28 μ mの場合、この振幅量は $\pm 15 \sim \pm 150$ nmとすることが必要である。特に、ウォブル信号のSignal to Noise Ratioを確保するとともに、アドレスマークを正確に検出するためには、 $\pm 25 \sim \pm 70$ nmの値が好ましい。Address1、Address2は、上記第1、および第2の実施の形態で説明した方法と同じ方法でディスク上に記録され、再生される。このために、アドレスが奇数のグループとアドレスが偶数のグループでは、逆相のウォブルがアドレスマークとして記録される。このために2ビットが割与えられ、(0, 1)か(1, 0)で奇数が偶数かを識別する。そして、この識別を確実にを行うために、グループとランドの幅の比率が概ね1対1で、グループのピッチが1.0~1.28 μ mの場合、ウォブルには $\pm 30 \sim \pm 150$ nmの振幅量を与える必要がある。ウォブルの振幅量としては、特に $\pm 60 \sim \pm 120$ nmの範囲の値が好ましい。

【0022】また、本実施の形態では、アドレス部のウォブルの最高周波数は8ビットデータ長を与えた。これは、記録密度との関係によって選択すればよい。一ウォ

ブル周期が $1.2\mu\text{m}$ 以上でアドレス部のエラーレートやマージンが改善され、再生を精度良く行うことができた。一方、この周期を長くすると、データのフォーマット効率が低下するので、周期長として $1.2\sim 5\mu\text{m}$ の間にする必要がある。これは、光磁気ディスクに限らず、相変化ディスク、色素系あるいは金属系の追記型の光ディスクにおいても同様の結果が得られた。また、PA1は隣接グループの位相は同期しており、トラッキング信号のオフセット補正のためにファインクロックマークの代わりとして使用することも可能という特徴を合わせ持っている。

【0023】ここで、アドレスマーク付近のディスク上のウォブルの形状は図14に示すようになる。同期信号生成用のウォブルは、アドレスマーク、spaceの後、同期信号として生成される。アドレスマークの前には、Address1やAddress2、PA1、PA2、PA3等が形成されている。また、アドレスマーク部分は、アドレスが奇数のグループとアドレスが偶数のグループでは、2ビットが割与えられ、(0, 1)か(1, 0)で逆相のウォブルとして形成されている。Address1やAddress2の識別は、アドレスマークを使用して行われる。そして、この識別を確実に行うために、グループとランドの幅の比率が概ね1対1で、グループのピッチが $1.0\sim 1.28\mu\text{m}$ の場合、ウォブルの振幅量は $\pm 30\sim \pm 150\text{nm}$ の値が好ましい。更には、 $\pm 60\sim \pm 120\text{nm}$ の値が好ましい。

【0024】また、本実施の形態では、アドレス部のウォブルの最高周波数は8ビットデータ長で与えた。これは、記録密度との関係によって選択されればよい。ウォブル周期が $1.2\mu\text{m}$ 以上でアドレス部のエラーレートやマージンが改善され、再生を精度良く行うことができた。一方、この周期を長くすると、データのフォーマット効率が低下するので、周期長として $1.2\sim 5\mu\text{m}$ の間にする必要がある。これは、光磁気ディスクに限らず、相変化ディスク、色素系あるいは金属系の追記型の光ディスクにおいても同様の結果が与えられた。また、PA1は隣接グループの位相は動記しており、上記第4の実施の形態で説明したトラッキング信号のオフセット補正のためのファインクロックマークの代わりとして使用することも可能であるという特徴を有する。

【0025】本発明に依れば、アドレスデータやデータの同期信号生成用のクロックが、ビットではなく、全てウォブルに記録された情報から検出することができる。ISOの90mm光磁気ディスクなどでは、ディスク上にアドレス信号がビットで記録されており、高速アクセス時などにおいてトラックのミスカウントを起こすことがあった。しかし、本発明に依れば、ディスクからの反射光量は殆ど一定であり、高速アクセス時などにおいてもミスカウントすることを防ぐことが出来る。

【0026】また、本発明に依れば、ミニディスクで

は、アドレス信号をウォブルの形として記録している。即ち、アドレス信号を倍フェーズ変調した後、周波数変調をかけた信号でウォブルを変調して記録している。しかし、この場合、キャリア周波数のSignal to Noise Ratioが悪くなることと、アドレス信号を周波数変調で記録しているために帯域幅が増えることにより、データの記録再生を行うための同期信号をキャリア周波数から生成することは困難である。本発明の場合、ウォブル周波数を抜き出す、バンドパスフィルターの帯域はPLLをかけるのみ必要な帯域程度であればよく、狭帯域のフィルターを用意すれば良い。このために、ウォブルの振幅は小さくて、CN比が少々悪くても、実際のSignal to Noise Ratioは良くなる。従って、ジッタの少ないPLLをかけることができ、データの記録再生を行うための同期信号を精度良く生成することができる。また、この場合、ウォブルの振幅量が小さくてもよいので、ウォブル信号が光磁気ディスクの信号に漏れ込みが少なくなるという特徴がある。

【0027】また、アドレス部とデータ領域の記録部を別にすることにより、且つアドレスデータやデータの同期信号生成用のクロックがビットではなく、全てウォブルに記録された情報から検出することができることにより、アドレスデータの光磁気信号やアクセス性能への悪影響をなくすることができる。

<第2の実施の形態>図15を参照して、第2の実施形態について説明する。図15に示す光ディスクは、ランドとグループ用のアドレス部310をウォブル311で形成し、アドレス部310に続くデータ領域313には、ウォブル312が形成されている光ディスクである。前記ウォブル312はグループの両側の壁に形成され、前記ウォブル311はグループのいずれか一方の壁に形成されている。図15に示す光ディスクでは、アドレス部310にはアドレスが1つ記録されており、前記ウォブル311はランド用のアドレスとグループ用のアドレスの両方を兼ねている。再生時には、レーザービームが前記ウォブル311を再生してランド、もしくはグループのアドレスを検出する。その後、レーザービームがデータ領域313においてデータを再生するとともに前記ウォブル312を再生し、検出したウォブルの波形から再生信号の同期信号を生成する。

【0028】また、第2の実施形態における光ディスクは、図16に示す構造のものでもよい。図16に示す光ディスクにおいては、アドレス部320には、データ部323のウォブル322にアドレス情報を記録するためのウォブル321が重畳されるように形成されているのが特徴である。また、図15、図16に示す光ディスクにおいては、アドレス部に形成される前記ウォブル311、321はアドレス情報で変調され、その波長は、データ部に形成される前記ウォブル312、322の波長

より短い。

【0029】図17を参照して、前記アドレス部310と前記データ領域313を含むディスクの詳細について説明する。ディスクの1トラック（一周）はNf個のフレームに分けられる。一フレームは2688バイトの大きさを表され、64バイトのアドレス部と、2624バイトのデータ部に分割される。光磁気信号は、主にデータ部に、NRZI変調や(1-7)変調を使用して記録、もしくは再生される。この場合、記録されるデータのビット密度を $0.22\mu\text{m}/\text{bit}$ とすると一フレーム当たりの長さは、 4.73088mm 、 $0.20\mu\text{m}/\text{bit}$ とすると 4.3008mm となる。従って、コンパクトディスク(CD)と同じ12cmサイズのディスクの場合、トラック一周当たりのフレーム数Nfは、30から87程度となる。

【0030】次に、アドレス部は64バイトの長さを持ち、アドレス部の最小の一ウォブル周期を一バイトとすると、一ウォブル周期のディスク上の長さは、 $1.60\sim 1.76\mu\text{m}$ の範囲となる。また、プリアンプル1(PA1)に8バイト、アドレスに48バイト、アドレスマーク(AM)に2バイト、プリアンプル2(PA2)に4バイト、spaceに2バイトのディスク上の長さを与える。この場合、実際のデータ長としては、図18に示すように、PA1、PA2は、それぞれ、8ビットと4ビット、Addressは48ビット、AMは2ビット、spaceは2ビットが割与えられる。

【0031】更に、次に、データ領域は2624バイトの長さを持ち、PA3が24バイト、データエリアが2592バイト、PA4が8バイトから構成される。データエリアの2592バイトは、ユーザデータが2048バイト、記録信号のDC成分抑圧用データが32バイト、あるいはエラー訂正のためのデータ等から構成される。この場合、データの記録や再生を行うための同期信号を生成するためのウォブルの一周期に16バイトの長さを与えると、ディスク上の一ウォブルの長さは、ビット密度が $0.22\mu\text{m}/\text{bit}$ の場合は $28.16\mu\text{m}$ 、 $0.20\mu\text{m}/\text{bit}$ の場合は、 $25.6\mu\text{m}$ となる。そして、一フレームの中のデータ部には164個のウォブルが存在することになる。従って、一トラックの中に60個のフレームが存在し、ディスクが1500rpmで回転したとすると、ウォブルの周波数は、252kHzとなる。このウォブルの周波数を利用してデータを記録または/および再生するためのデータ同期信号をPLLによって生成する。これは、図11に示すように、データ変調方式としてNRZIを使用した場合、データ同期信号は、32.256MHzとなり、PLLの分周器271の比率は128分の一となる。また、一ウォブルの長さは、16バイトに限らず、一定のバイト相当の長さ、例えば、4バイト、あるいは8バイト、あるいは20バイトという長さを与えることができる。この場合、

ウォブルの周波数が先の252kHzに比べて早く、あるいは遅く設定され、同期信号を生成するPLLの周波数が異なることになる。本実施の形態では、一ウォブルの周期長が $5\sim 50\mu\text{m}$ の範囲の値が好ましい。ウォブル信号のSignal to Noise Ratioを確保するためには、ウォブルの振幅量が大い方がよい。一方、振幅量が大いとき、光磁気ディスクの場合、光磁気信号にウォブル周波数が漏れ込み、悪影響を与える。このことは、図12、図13で述べたことと同じである。即ち、グループとランドの幅の比率が概ね1対1で、グループのピッチが $1.0\sim 1.28\mu\text{m}$ の場合、ウォブル信号のSignal to Noise Ratioを確保するとともに、光磁気ディスクの信号を精度良く再生するためには、ウォブルに $\pm 10\sim \pm 60\text{nm}$ の振幅量を与えることが必要である。特に、ビット密度が $0.15\sim 0.24\mu\text{m}/\text{bit}$ で、一ウォブル長が $10\sim 32\mu\text{m}$ の場合、ウォブルの振幅量が $\pm 10\sim \pm 40\text{nm}$ となる値が好ましい。一方、相変化ディスク、色素系あるいは金属系の追記型の光ディスクにおいては、一ウォブル長が $5\sim 50\mu\text{m}$ の範囲、振幅量が $\pm 10\sim \pm 60\text{nm}$ の範囲の値が好ましい。

【0032】図18を参照して、Addressには、48ビットが割与えられ、Frame addressはトラック一周の中で順を付けた番号である。従って、フォーマット上では、一トラックに最大1024のフレーム数(Nf)を確保することができる。そして、アドレス情報のデータの符号化方式には、バイフェーズ符号方式に限らず、マンチェスタ符号、NRZ、NRZI符号、マンチェスタ符号等が用いられる。Track addressは、ディスク全体のトラックを内周、もしくは外周から付けた通し番号である。この場合、ランド部L2nとグループ部G2nが同じアドレスGA2nにより与えられる。従って、フォーマット上では、最大1048576本のトラック数を確保することができる。PA1、およびPA2はAddressやアドレスマークを、正確に検出するためのプリアンプルやポストアンプルとして使用される。これらの信号を記録するために使用されるウォブルの振幅は、ほぼ同じ量を与える。グループとランドの幅の比率が概ね1対1で、グループのピッチが $1.0\sim 1.28\mu\text{m}$ の場合、この振幅量は $\pm 15\sim \pm 150\text{nm}$ とすることが必要である。特に、ウォブル信号のSignal to Noise Ratioを確保するとともに、アドレスマークを正確に検出するためには、 $\pm 25\sim \pm 90\text{nm}$ の値が好ましい。アドレスマークは、アドレス部の識別や信号の記録、再生の開始信号として使用される。そして、この識別を確実に行うために、グループとランドの幅の比率が概ね1対1で、グループのピッチが $1.0\sim 1.28\mu\text{m}$ の場合、ウォブルには $\pm 30\sim \pm 200\text{nm}$ の振幅量を与える必要がある。ウォブルの振幅量としては、特に $\pm 60\sim \pm 150$

nmの範囲の値が好ましい。また、本実施の形態では、アドレス部のウォブルの最高周波数は8ビットデータ長を与えた。これは、記録密度との関係によって選択すればよい。一ウォブル周期が $1.2\mu\text{m}$ 以上でアドレス部のエラーレートやマージンが改善され、再生を精度良く行うことができた。一方、この周期を長くすると、データのフォーマット効率が低下するので、周期長として $1.2\sim 5\mu\text{m}$ の間にする必要がある。これは、光磁気ディスクに限らず、相変化ディスク、色素系あるいは金属系の追記型の光ディスクにおいても同様の結果が得られた。

【0033】また、本実施の形態では、アドレス部のウォブルの最高周波数は8ビットデータ長で与えた。これは、記録密度との関係によって選択されればよい。一ウォブル周期が $1.2\mu\text{m}$ 以上でアドレス部のエラーレートやマージンが改善され、再生を精度良く行うことができた。一方、この周期を長くすると、データのフォーマット効率が低下するので、周期長として $1.2\sim 5\mu\text{m}$ の間にする必要がある。これは、光磁気ディスクに限らず、相変化ディスク、色素系あるいは金属系の追記型の光ディスクにおいても同様の結果が与えられた。

【0034】本実施の形態によれば、上記図1に基づく実施の形態に比べて、グループに1つのアドレスを設ければ良いので、一層のフォーマットの効率化を図ることができる。上記の説明においては、アドレス部の後に設けられるアドレスマークの有無については、述べていないが、アドレスマークを設けていても良く、アドレスマークを設けていなくても良い。アドレスマークを設ける場合には、アドレスマークの振幅は上記第5の実施の形態で説明したものと同じものを用いる。

【0035】＜第3の実施形態＞上記第1の実施形態におけるデータ領域171、または第2の実施形態におけるデータ領域313、323は、グループの両側、もしくはいずれか一方に所定周期のウォブルを設けた構造である。かかる構造の光磁気ディスクを再生した場合、グループに設けられたウォブルにより照射されたレーザービームの反射光の偏光方向が影響を受け、記録信号を正確に再生できないという問題がある。図19を参照して、更に、詳しく説明する。図19に示す光磁気ディスクの構造では、グループの両側に設けられたウォブル351は、同じ位相であるため、レーザービーム352がグループGに照射された場合、その反射光は、本来の信号の磁化に基づいた偏光波ではなく、グループGの両側に設けられたウォブル351、351によって決定されるグループの方向を示す矢印353と同じ方向の偏光波成分を有する。また、レーザービーム354が照射されたときは、その位置でのグループGの方向を示す矢印355と同じ方向の偏光波成分を有する。従って、図19に示す構造では、本来記録した信号の再生信号の成分に、グループGの両側に設けられたウォブル351、351の影

響による偏光波成分が現れ、再生特性を低下させ、その結果、記録信号を正確に検出できないという問題がある。この問題は、グループGのいずれか一方にウォブルを設けた場合にも生じるものである。以下、再生信号成分に対するウォブルの影響による偏光波成分の割合を洩れ込み量と定義する。

【0036】そこで、第3の実施形態においては、かかる上記の問題を解決し、グループGの両側に設けたウォブルが再生特性に影響を与えない光磁気ディスクを提供することを目的とする。図20を参照して、第3の実施形態について説明する。図20は、データ領域のグループの両側の壁に設けたウォブル構造を示している。グループGの両側には所定の波長Wと振幅hとのウォブル361が同位相で形成されている。本第7の実施の形態においては、上記洩れ込み量を -25dB 以下、エラーレートを 1×10^{-4} 以下にするように、ウォブル361の波長Wと振幅hを設定することを特徴としている。

【0037】図21にウォブル361の波長Wを $0.5\sim 1.0\mu\text{m}$ の範囲、振幅hを $\pm 3\sim \pm 0\text{nm}$ の範囲で変化させたときの洩れ込み量を示す。図21より、ウォブル361の波長Wが $0.5\sim 1.0\mu\text{m}$ の範囲、振幅hが $3\sim 25\text{nm}$ の範囲で、洩れ込み量が -25dB 以下となる。また、図22にウォブル361の波長Wを $0.5\sim 1.0\mu\text{m}$ の範囲、振幅hを $\pm 3\sim \pm 50\text{nm}$ の範囲で変化させたときのエラーレートを示す。上記第5の実施の形態と同じように前記ウォブル361を再生して得られる再生信号から同期信号を生成して記録再生したものである。この場合、同期信号生成回路の分周比は $1/3\sim 1/64$ に設定しており、同期信号長は $0.15\sim 0.26\mu\text{m}$ である。図22よりウォブル361の波長Wが $0.8\sim 1.0\mu\text{m}$ の範囲、振幅hが $5\sim 25\text{nm}$ の範囲で、エラーレートが 1×10^{-4} 以下となる。振幅hが $\pm 25\text{nm}$ 以上では、洩れ込み量が大きくエラーレートが劣化し、 $\pm 5\text{nm}$ 以下では、ウォブル再生信号から得られる同期信号の特性が劣化するためエラーレートが悪化する。ここで、ウォブル361の波長Wは、好ましくは、 $1.2\sim 5.0\mu\text{m}$ の範囲、更に好ましくは $1.6\sim 3.0\mu\text{m}$ の範囲である。

【0038】従って、図21、図22の結果より、洩れ込み量が -25dB 以下で、且つ、エラーレートが 1×10^{-4} 以下を実現できるウォブル361の波長Wと振幅hの範囲は、それぞれ、 $0.8\sim 1.0\mu\text{m}$ の範囲、 $\pm 5\sim \pm 25\text{nm}$ の範囲となる。尚、このウォブルの波長Wと振幅hとの値は、グループGのいずれか一方の壁にウォブルを設けた場合にも適用できる値である。また、第3の実施形態においては、アドレス情報ブロックのウォブリングの周期は、 $1.60\sim 2.0\mu\text{m}$ の範囲である。

【0039】第3の実施の形態では、データ領域のグループGの構造について説明したが、アドレス領域等のその他の部分については、上記第1、および第2の実施形

態と同じであるので、その説明は省略する。尚、本実施の形態では、上記第1、および2の実施形態より、同期信号を精度良く生成できるため低いエラーレートが得られる。

【0040】＜第4の実施形態＞第3の実施形態においては、グループの両側の壁に設けた同位相のウォブルにより照射されたレーザビームの反射光の偏光方向が影響を受け、再生特性にこの影響が現れる洩れ込みが生じるが、第4の実施形態においては、この洩れ込みを消去する方法について説明する。

【0041】図23、図24、図25を参照して、第1の消去方法について説明する。本発明に係る光磁気記録媒体39は、内周部392と外周部391にTOC領域が設けられている。第1の消去方法においては、内周部392と外周部391に設けられたTOC領域の双方、もしくはいずれか一方に洩れ込み量に関する情報を記録しておき、この情報を再生時に検出することにより、再生信号から洩れ込みを消去するというものである。図24は、洩れ込みを消去する回路構成を示すものである。端子40に再生信号が入力され、バンドパスフィルタ(BPF)41でノイズが除去された後、PLL回路42と補正信号発生回路44に送られる。PLL回路42には、図25の(a)に示すグループの両側に設けられたウォブル波形の信号が送られ、PLL回路42は(a)に示すウォブル波形から同期信号を検出し、端子43を介して図示省略したレーザ駆動回路、復号器に送られ、この同期信号に同期して信号が再生される。補正信号発生回路44は、端子45から入力される光磁気記録媒体39のTOC領域に記録されていた洩れ込み量に関する情報に基づいて、信号(a)の位相と振幅が信号(b)のウォブル波形の位相と振幅に等しくなるように補正し、減算器47に送る。また、端子46から、(b)に示すウォブル波形が重畳した光磁気再生信号が減算器47に入力される。減算器47は、信号(b)の波形から信号(a)の補正された波形を減じる演算を行い、信号(c)を生成する。生成された信号(c)は図示省略した復号器に送られ、所定の復調がされた後、再生信号として取り出される。その結果、ウォブルによる再生信号への洩れ込みが消去される。

【0042】図26、図27を参照して、第2の消去方法について説明する。第2の消去方法においては、上記図23に示した光磁気記録媒体39のTOC領域に記録された補正量に基づいて変化させる補正量を決定し、変化させた各補正量に対する再生信号のエラーレートを検出する。その後、エラーレートが最小となる補正量を決定し、決定した補正量に対する信号を再生信号として検出する。図26は第2の消去方法に用いられる回路である。補正量発生回路420には、前記TOC領域から検出された補正量が入力され、補正量発生回路420は、入力された補正量に基づいて変化させる補正量の範囲を

決定し、減算器422に送る。一方、端子421から減算器422に再生信号が入力され、減算器422は、再生信号から決定された各補正量を減じる演算を行い、その結果、エラーレート検出回路423に送る。エラーレート検出回路423は、各補正量に対するエラーレートを検出する。各補正量に対するエラーレートは図27に示す関係となるので、エラーレート検出回路423はエラーレートが最小となる補正量を決定し、決定した補正量に対する再生信号を端子424から出力する。この場合、変化させる補正量の範囲は、補正量の0.3～3倍の範囲である。

【0043】図28、図29、図30を参照して、第3の消去方法について説明する。光磁気記録媒体440は、内周部392と外周部391にTOC領域を有し、信号記録領域445には、再生信号に関する情報を記録した領域(以下、特定領域という)441、443と信号領域442、444とが組となって形成されている。前記特定領域441、443には、[11111...]、[00000...]、[1010101...]のいずれか1つの信号が記録されており信号の再生に先立ってこれらの信号を再生される。これらの信号は、光磁気信号が記録されていない再生信号と同等の信号となる。即ち、ウォブル波形に基づく信号のみとなる。従って、この信号を再生信号から減じることにより洩れ込み量を消去することが可能となる。端子450からは[11111...]、[00000...]、[1010101...]のいずれか1つの信号を再生した図30の(d)に示す再生信号が入力され、波形メモリ451にメモリされる。一方、端子452からは図30の(e)に示す再生信号が減算器453の+端子に入力され、これに同期して減算器453の-端子に波形メモリ451から(d)の信号が入力される。減算器453は、入力された(d)と(e)との信号の減算を行い、洩れ込みのない(f)の信号を得て、端子454から出力する。その後、図示省略した復号器に送られ、再生信号として取り出される。この場合は、[11111...]、[00000...]、[1010101...]の信号から洩れ込み量を検出したが、磁気ヘッド等の外部磁界印加手段により光磁気記録媒体の再生層の磁化を一方向に揃えても同様に洩れ込み量を検出できる。

【0044】図31、図32を参照して、第4の消去方法について説明する。端子470に入力された再生信号はA/D変換器471でA/D変換された後、減算器475と同期検波回路に送られる。同期検波回路472では、送られてきた再生信号から図32に示すウォブル波形の1波長に相当する再生信号を検出し、加算器473に送る。加算器473は、1波長分の再生信号を100～10000回の範囲で加算し、平均化する。その結果を、波形メモリ474に送る。A/D変換後の再生信号

は減算器475の+端子に入力され、これと同期して-端子には波形メモリから平均化された信号が入力される。減算器475は入力された再生信号から平均化された信号を減じる演算を行うことにより洩れ込みを消去できる。

【0045】図33を参照して、第5の消去方法について説明する。信号(g)の波形Bは4 bytes分の信号を表し、信号(h)の波形Cは、次の4 bytes分の信号を表す。また、波形Aは、グループの両側に形成された同位相のウォブル波形を表す。この第5の消去方法においては、信号(g)の波形Bから波形Aを減算して信号(i)を得る。また、信号(h)の波形Cに波形Aを加算して信号(j)を得る。波形Aの振幅をA1、波形Bの振幅をB1、波形Cの振幅をC1、信号(i)の振幅をBh、信号(j)の振幅をChとした場合、 $(Ch - Bh) / 2 = [(C1 + A1) - (B1 - A1)] / 2 = A1$ となるので、洩れ込み量を消去した再生信号が得られる。

【0046】＜第5の実施形態＞第5の実施形態においては、本発明に係る光ディスクの更に他の変形例について説明する。図34を参照して、第5の実施形態における光ディスクの平面構造について説明する。グループの一方の壁には、ウォブル50を、ウォブル50が形成された壁と異なる壁にはウォブル51が形成されている。ここで、再生データレートを24 MHzとした場合、前記ウォブル51の周波数は3 MHzであり、前記ウォブル50の周波数は281.25～375 kHzである。第5の実施の形態においては、前記ウォブル50と前記ウォブル51とは光ディスクの全領域に亘って形成されている。また、ウォブル50には、周波数変調でアドレス情報が記録される。

【0047】図35を参照して、ウォブル50に記録されるアドレスフォーマットについて説明する。図35は1セクター当たりのアドレスフォーマットを示したものである。アドレスフォーマットは、同期パターンとして4 bits、フレームアドレスとして24 bits、Reservedとして4 bits、ECCとして12 bitsを設けたフォーマットである。1セクターには、Data Areaが2kB含まれるため、上記説明したアドレスは2kB分のデータに対するアドレスを示すことになる。また、前記ウォブル51はデータ1 bytesに対して1個、1セクターに対して2816個形成される。この2816個/1セクターのウォブル51は、記録再生用クロックを生成する基準になるものである。また、前記ウォブル51はデータ2 bytesに対して1個、1セクターに対して1408個形成されている。

【0048】図36を参照して、本第9の実施の形態における光ディスクの再生について説明する。再生回路としては、上記図8に示した再生回路を用いる。図34に

おけるグループ1(G1)をレーザビームにより再生する場合、その再生信号は図36の(a)の信号となる。これは、G1の一方にはウォブル50が形成され、他方にはウォブル51が形成されているため、両ウォブルからの信号が重畳された形で再生信号として検出されるからである。次に、図34のL1をレーザビームにより再生する場合も、一方にウォブル51、他方にウォブル50が形成されているため、再生信号としては図36の(a)の信号となる。図34のG2、L2をレーザビームにより再生する場合も同様である。従って、得られた再生信号を前記ウォブル50からの信号と、前記ウォブル51からの信号とに分離する必要がある。

【0049】そこで、図8を参照して、プッシュプル信号として再生された(a)の信号は狭帯域のバンドパスフィルタであるBPF243とアドレス復調のバンドパスフィルタであるBPF244に送られる。BPF243に送られた再生信号は高周波数成分だけ抽出され、図36の(c)の信号が取り出される。この(c)の信号は前記ウォブル51に対応する信号である。BPF243で取り出された信号(c)はコンパレータ245に送られる。コンパレータ245は、送られた信号(c)を2値化し、2値化した信号をPLL246におくる。PLL246は送られた2値化信号から、その立ち上がりタイミングに対応してデータクロックを生成する。生成された同期信号はディスクの回転制御回路とデータクロック生成回路に送られ、回転制御とクロック生成に用いられる。一方、前記BPF244は、送られた信号(a)より低周波数成分を抽出し、信号(b)を取り出す。信号(b)は前記ウォブル50に対応する信号である。取り出された信号(b)は図37に示したFM復調回路53へ送られる。前記PLL246により生成された同期信号は、クロック分配回路56にも送られ、クロック分配回路56でクロック分配が行われ、それぞれ、FM復調回路53、Biphase復調回路54、アドレスデコーダ回路55に送られる。FM復調回路53は、前記BPFから送られてきた信号(b)をクロック分配回路56から送られてきたクロックに同期してFM復調し、Biphase復調回路54に送る。Biphase復調回路54はクロック分配回路56から送られてきたクロックに同期して、FM復調されたアドレス信号をバイフェーズ復調し、アドレスデコーダ回路55に送る。アドレスデコーダ回路55は、クロック分配回路56から送られてきたクロックに同期してアドレスを出力する。

【0050】第5の実施形態に示した光ディスクは、グループのいずれか一方の壁に形成したウォブルから同期信号を精度良く生成できるとともに、ウォブルによる再生信号への洩れ込み量が少ない光ディスクとして有効である。しかし、更に、再生特性を良くしようとすれば、上記第4の実施形態に示した洩れ込みを消去する消去回

路を適用することが可能である。

【0051】上記第1～5の実施の形態においては、光磁気ディスクに限らず、相変化ディスク、色素系あるいは金属系の追記型光ディスクであってもよい。また、光ディスクに限らず、記録媒体であれば何でもよい。また、更に、光磁気ディスクに限って言えば、記録層に記録された信号の磁区を再生層に拡大転写して再生する光磁気ディスクであってもよい。

【0052】

【発明の効果】本発明によれば、光ピックアップのレーザービームを回折格子等によって分けたりする必要がなくなり、一つのレーザービームスポットのみをディスク記録面上に集光することによってグループとランドのアドレスを検出することが可能となる。

【0053】また、本発明によれば、グループおよびランドとの双方でウォブル信号を読みながら記録再生を行うことができるので、従来の光ディスクの2倍の密度で記録を行うことができるばかりでなく、光学部品を減らすとともに、レーザー光源から発光されたレーザービームのパワーを損なうことなく、ディスクの記録密度を十分に高めることが可能となる。

【0054】また、本発明によれば、ランドとグループとから成るトラックを有する光ディスクのグループにウォブルの形で設けられたファインクロックマークを検出することによりオフセットのない、信号の記録・再生が可能となる。また、本発明によれば、アドレスデータやデータの同期信号生成用のクロックが、ビットではなく、全てウォブルに記録された情報から検出することができる。

【0055】また、本発明によれば、アドレス部とデータ領域の記録部を別にすることにより、且つアドレスデータやデータの同期信号生成用のクロックがビットではなく、全てウォブルに記録された情報から検出することができることにより、アドレスデータの光磁気信号やアクセス性能への悪影響をなくすることができる。また、本発明によれば、1つのアドレス情報でランド、およびグループのアドレスを両用できるので、1つのアドレスをウォブルで記録すれば良く、一層のフォーマットの効率化を推進できる。

【0056】また、本発明によれば、データ領域に設けたウォブルの波長を $0.8 \sim 1.0 \mu\text{m}$ 、振幅を $5 \sim 25 \text{ nm}$ の範囲とすることにより、洩れ込み量が少なく、且つ、エラーレートの小さい光磁気ディスクを実現できる。また、本発明によれば、グループの両側の壁に設けたウォブルによる再生信号への洩れ込みを消去でき、特性の良い再生信号を得ることが可能となる。

【0057】また、本発明によれば、グループの一方の壁に低周波数のウォブルを、他方の壁に高周波数のウォブルを形成することにより、ウォブルの影響による再生信号への洩れ込み量を少なくできるとともに、外部同期

信号を精度良く生成可能である。また、上記効果は、光記録媒体、光磁気記録媒体等の記録媒体においても得られる。

【0058】また、本発明によれば、信号を記録または再生する領域のグループの両側の壁には、ウォブルが形成されていないため、グループでの反射光の偏光方向がウォブルにより影響を受け、その影響が再生特性に現れるという洩れ込みがなく、再生特性がよい記録媒体が得られる。また、本発明によれば、同期信号を生成するためのグループが設けられていない領域は、68バイト毎に設けられているため同期信号を確実に生成することが可能となり、信号の記録または再生における特性を向上させることができる。

【図面の簡単な説明】

【図1】第1の実施形態におけるランド／グループ構造を示す図である。

【図2】再生されたウォブル波形から同期信号を生成する方法を説明する図である。

【図3】第1の実施形態における他のランド／グループ構造を示す図である。

【図4】第1の実施形態における更に他のランド／グループ構造を示す図である。

【図5】第1の実施形態における更に他のランド／グループ構造を示す図である。

【図6】第1の実施形態における更に他のランド／グループ構造を示す図である。

【図7】第1の実施形態における更に他のランド／グループ構造を示す図である。

【図8】第1の実施形態における光ディスクからの再生信号の処理を説明する図である。

【図9】第1の実施形態におけるディスクフォーマットを示した図である。

【図10】第1の実施形態におけるアドレスフォーマットを示した図である。

【図11】第1の実施形態における、ウォブル信号から同期信号を生成する装置ブロックを示した図である。

【図12】第1の実施形態におけるウォブル信号のクロストーク量と光磁気ディスクの信号のエラーレートとの関係を示した図である。

【図13】第1の実施形態におけるクロストークとウォブル長との関係を示した図である。

【図14】第1の実施形態におけるアドレス部、アドレスマーク部、およびデータ領域に設けられるウォブル波形の一例を示した図である。

【図15】第2の実施形態におけるランド／グループ構造を示す図である。

【図16】第2の実施形態における他のランド／グループ構造を示す図である。

【図17】第2の実施形態におけるディスクフォーマットを示した図である。

【図18】第2の実施形態におけるアドレスフォーマットを示した図である。

【図19】グループの両側に設けたウォブルが再生信号成分に悪影響を与えることを説明するための図である。

【図20】第3の実施形態におけるウォブル構造を説明する図である。

【図21】第3の実施形態におけるウォブルの波長、振幅 h と洩れ込み量との関係を示す図表である。

【図22】第3の実施形態におけるウォブルの波長、振幅 h とエラーレートとの関係を示す図表である。

【図23】第4の実施形態における光磁気記録媒体の平面構造を示す図である。

【図24】第4の実施形態における第1の消去方法に係る消去回路のブロック図である。

【図25】第4の実施形態における第1の消去方法に係る信号を示す図である。

【図26】第4の実施形態における第2の消去方法に係る消去回路を示す図である。

【図27】第4の実施形態における第2の消去方法に係る再生信号のエラーレートと補正量との関係を示す図である。

【図28】第4の実施形態における光磁気記録媒体の他の平面構造を示す図である。

【図29】第4の実施形態における第3の消去方法に係る消去回路のブロック図である。

【図30】第4の実施形態における第3の消去方法に係る信号を示す図である。

【図31】第4の実施形態における第4の消去方法に係る消去回路のブロック図である。

【図32】第4の実施形態における第4の消去方法に係る信号を示す図である。

【図33】第4の実施形態における第5の消去方法に係る信号処理を説明するための信号を示す図である。

【図34】第5の実施形態における光ディスクの平面構造を示した図である。

【図35】第5の実施形態における光ディスクに記録するアドレスフォーマットを説明する図である。

【図36】第5の実施形態におけるウォブルの再生信号を示した図である。

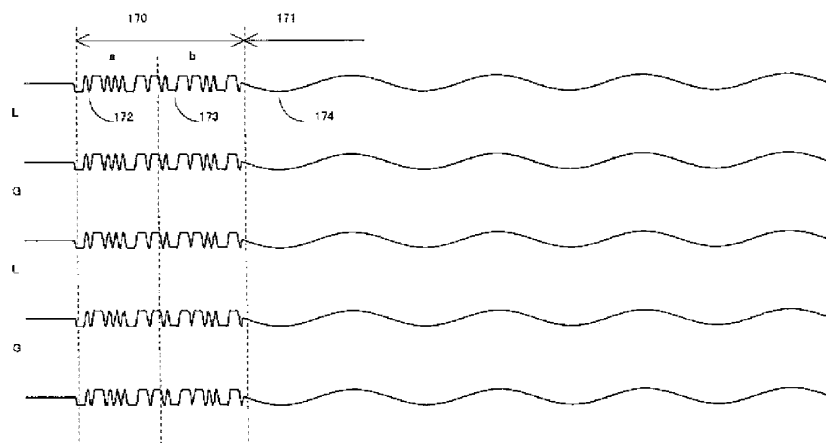
【図37】第5の実施形態におけるアドレス情報の再生を説明する回路図である。

【図38】従来方式による3つのビームを使用してランドとグループを記録再生するためのトラッキングの制御方法とアドレスの検出方法を示した図である。

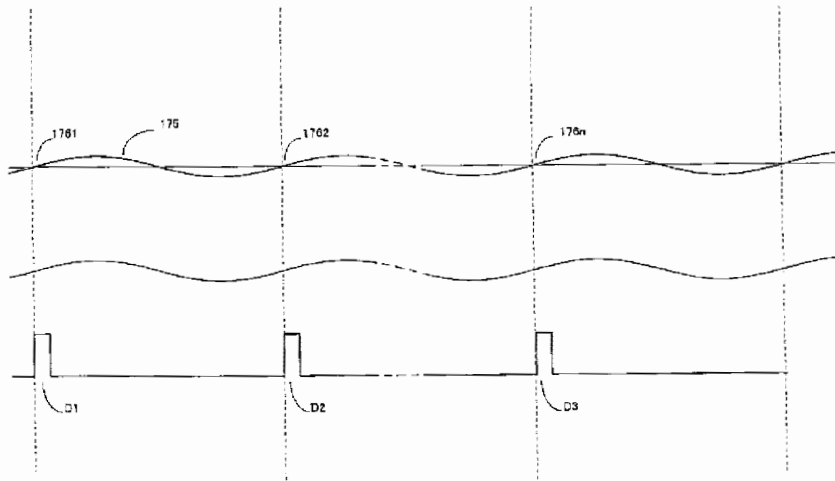
【符号の説明】

170	アドレス部
171	データ領域
172	ウォブル
420	補正量発生回路
423	エラーレート検出回路
441、443	特定領域
442、444	信号領域
445	信号記録領域
451、474	波形メモリ
471	A/D変換器
472	同期検波回路
473	加算器

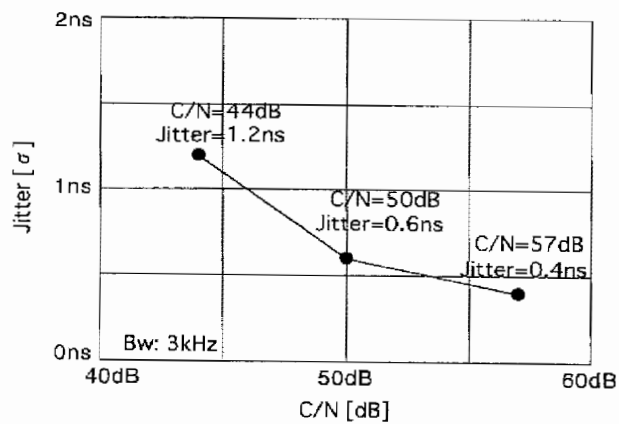
【図1】



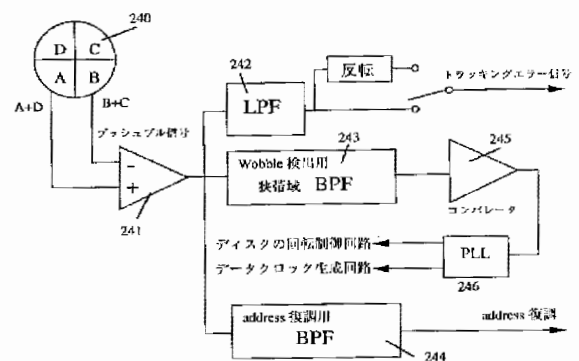
【図2】



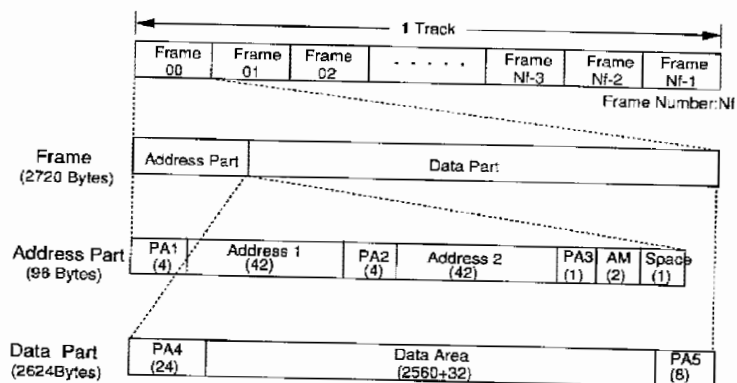
【図3】



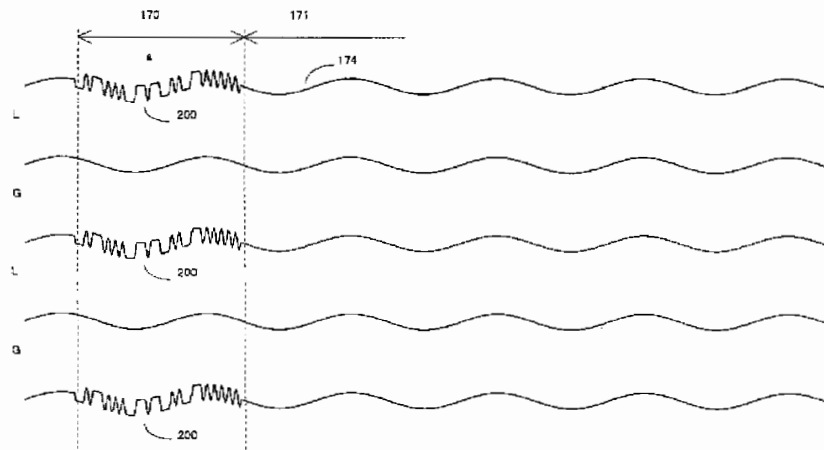
【図8】



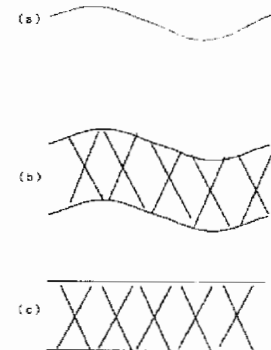
【図9】



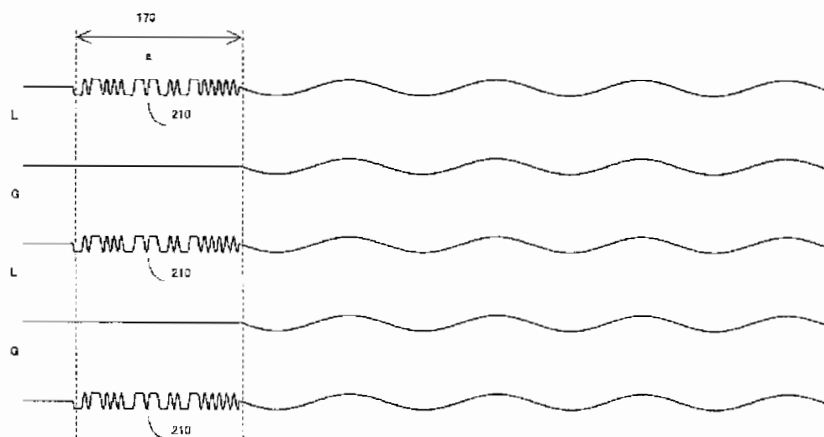
【図4】



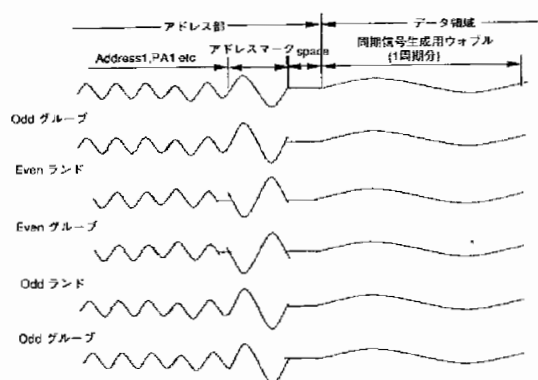
【図25】



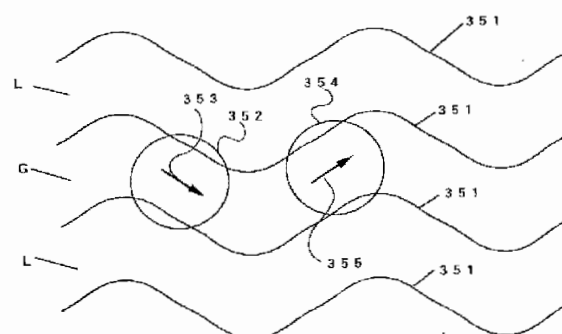
【図5】



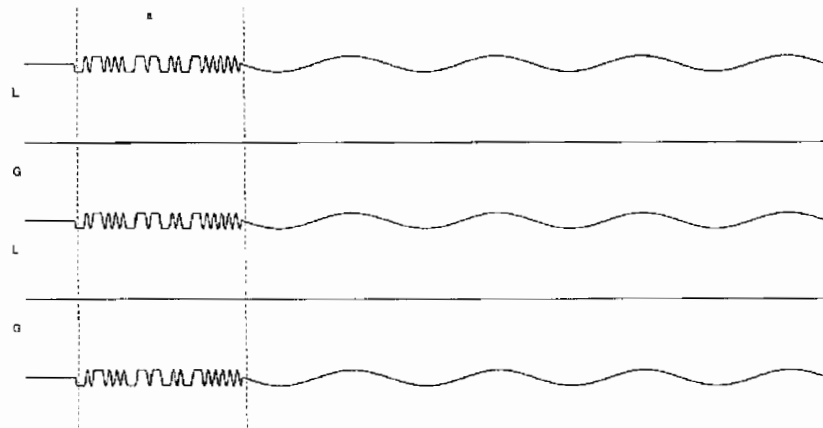
【図14】



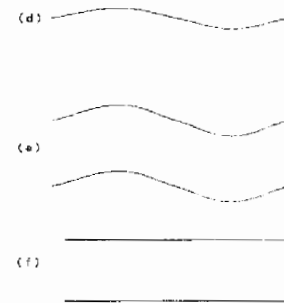
【図19】



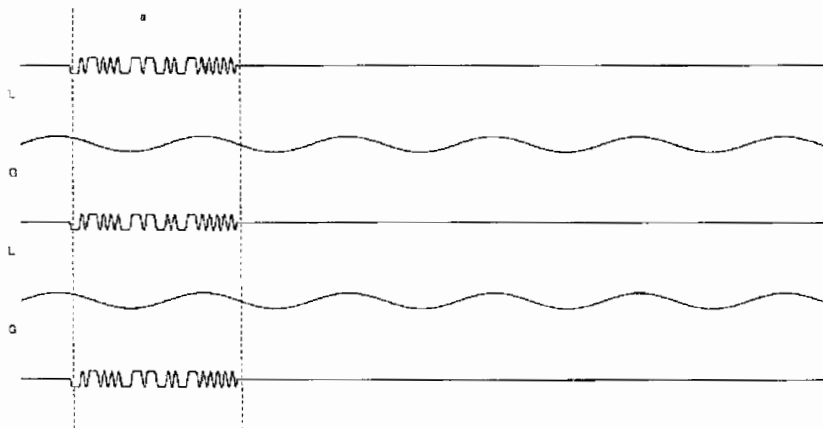
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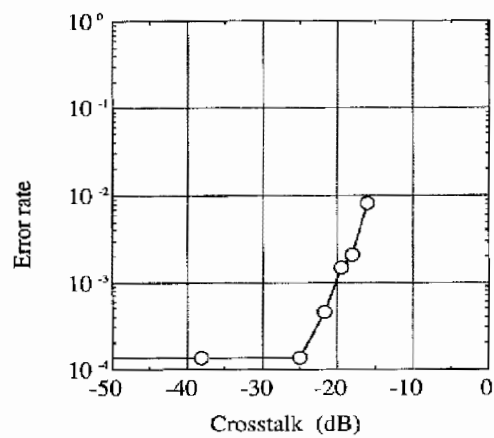
【図30】



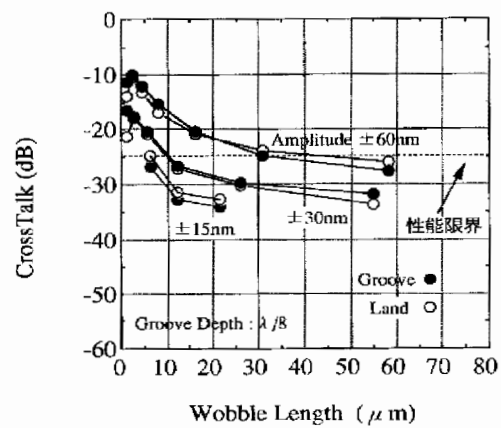
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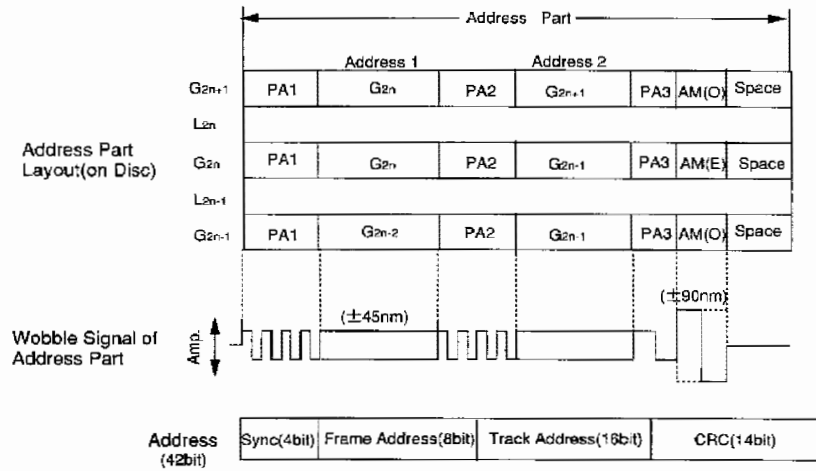
【図12】



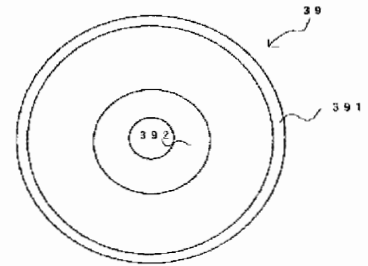
【図13】



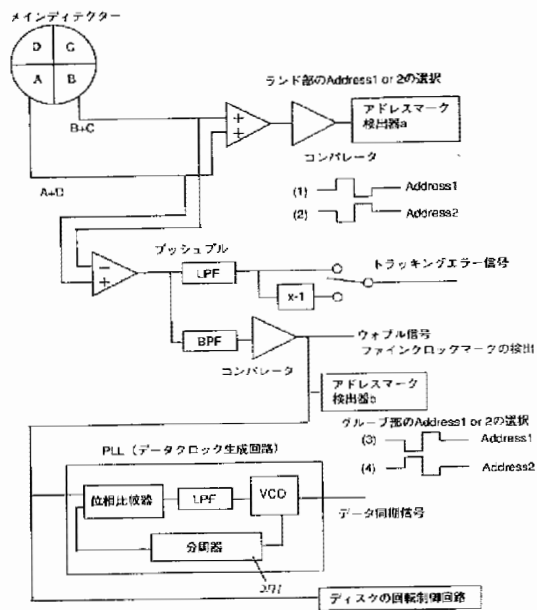
【図10】



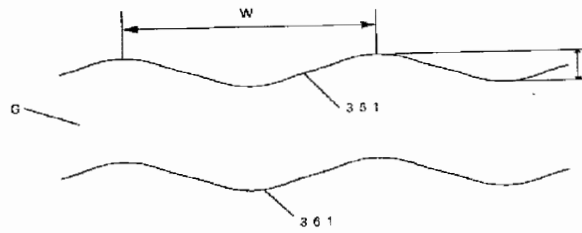
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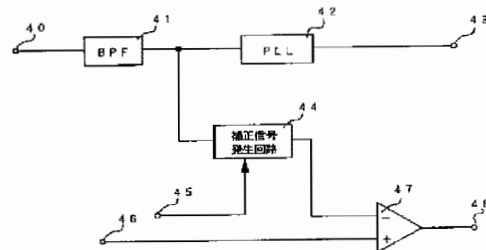
【図11】



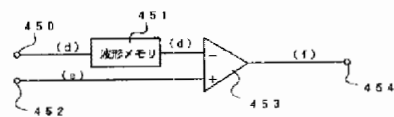
【図20】



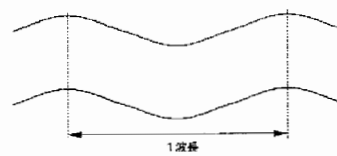
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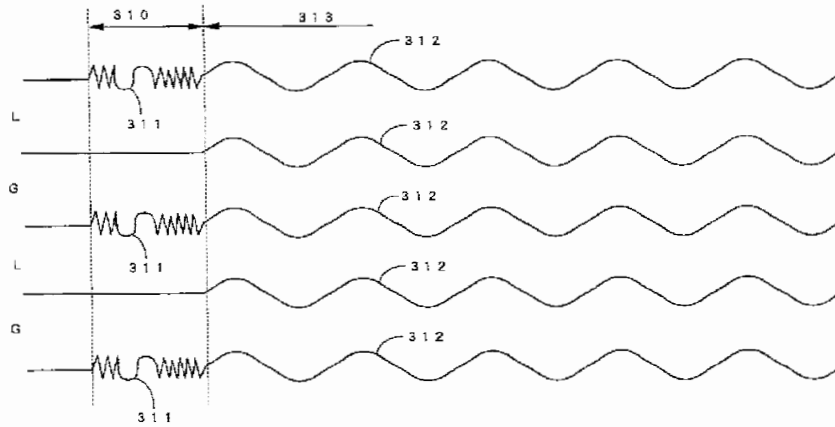
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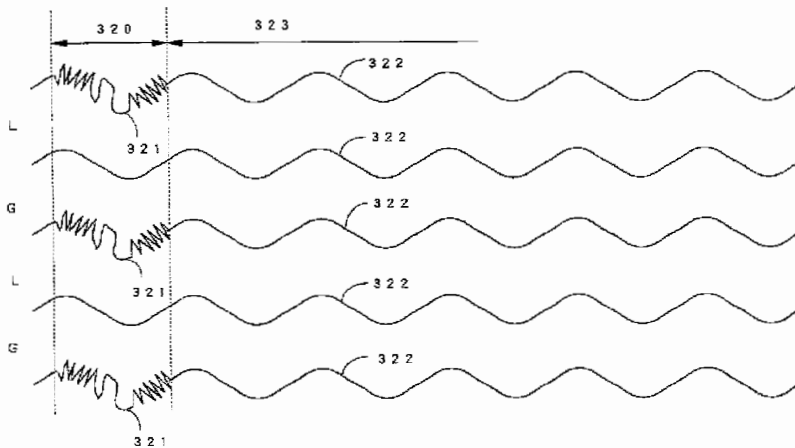
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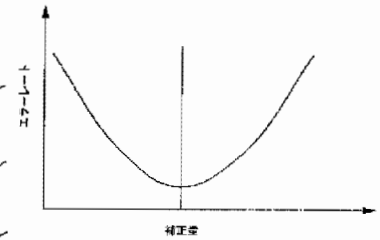
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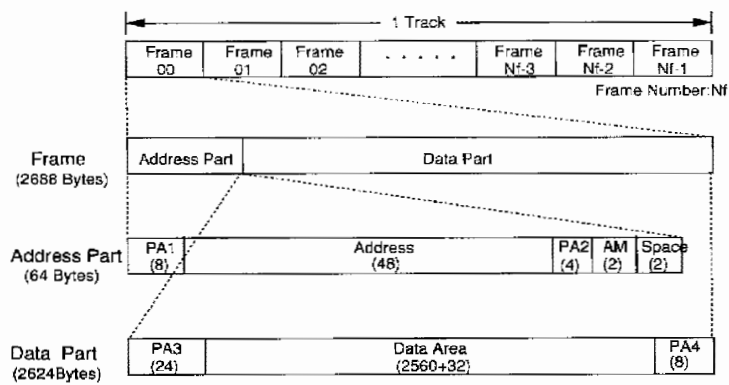
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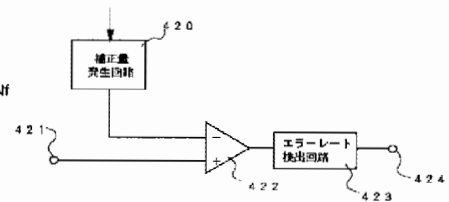
【図27】



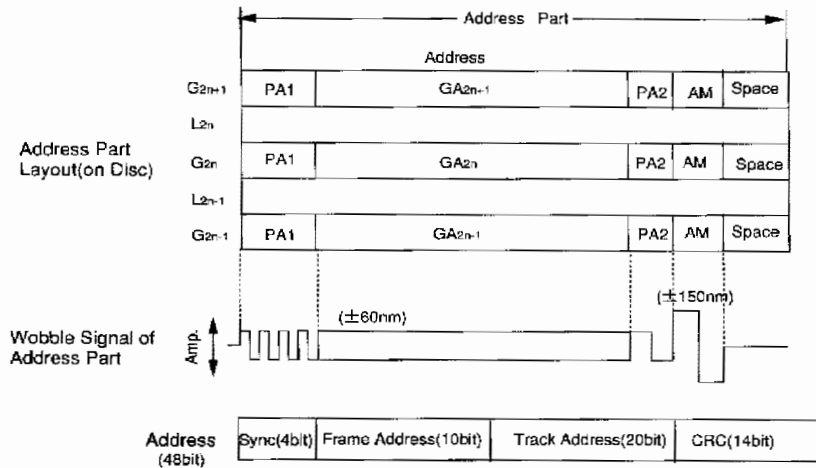
【図17】



【図26】



【図18】



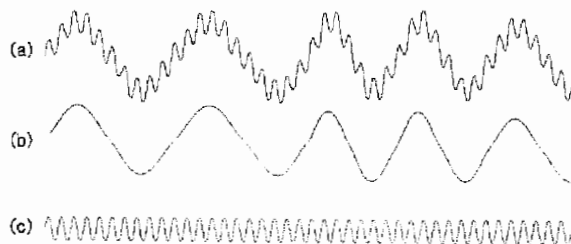
【図21】

Wobble長 (nm)	Wobble長 (μm)									
	0.5	0.8	1.2	1.6	1.88	2.08	3	5	10	
3	-40	-40	-40	-38	-39	-40	-40	-40	-42	
5	-40	-40	-39	-37	-35	-35	-40	-40	-41	
7	-39	-39	-38	-35	-32	-33	-37	-37	-41	
10	-39	-38	-35	-30	-30	-30	-32	-33	-35	
14	-35	-35	-30	-29	-28	-28	-29	-30	-34	
20	-30	-28	-26	-26	-25	-26	-28	-27	-34	
25	-28	-26	-23	-22	-20	-20	-20	-25	-32	
35	-18	-20	-20	-13	-10	-15	-16	-20	-26	
50								-15	-20	

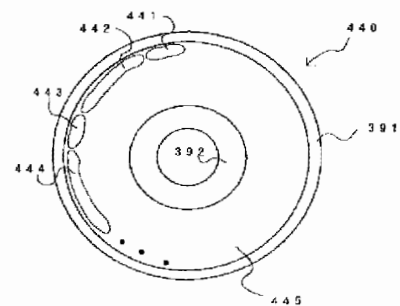
【図22】

エラーレート	Wobble長 (μm)									
	0.5	0.8	1.2	1.6	1.88	2.08	3	5	10	
3	1.0e-3	2.5e-3	2.0e-3	1.7e-3	1.3e-3	2.1e-3	1.7e-3	2.1e-3	3.2e-2	
5	1.5e-3	5.5e-3	1.0e-3	6.3e-6	5.3e-6	4.7e-6	6.9e-6	8.5e-6	5.1e-5	
7	5.6e-4	7.5e-6	7.5e-6	5.6e-6	8.6e-6	5.6e-6	7.1e-6	8.6e-6	9.7e-6	
10	8.6e-4	5.6e-6	9.6e-6	5.5e-6	4.5e-6	5.0e-6	6.7e-6	7.5e-6	8.6e-6	
14	9.6e-4	6.7e-6	8.0e-6	5.4e-6	5.5e-6	6.3e-6	5.5e-6	6.3e-6	7.8e-6	
20	5.3e-4	2.0e-5	4.3e-5	6.0e-5	8.0e-5	7.0e-5	1.2e-5	1.3e-5	8.8e-6	
25	3.2e-4	8.5e-5	4.7e-4	5.3e-3	3.1e-4	3.7e-4	5.3e-4	9.7e-5	9.6e-6	
35	1.2e-3	3.5e-3	8.0e-3	1.0e-2	2.5e-2	8.6e-3	6.5e-3	6.7e-3	8.3e-5	
50								3.2e-2	3.2e-3	

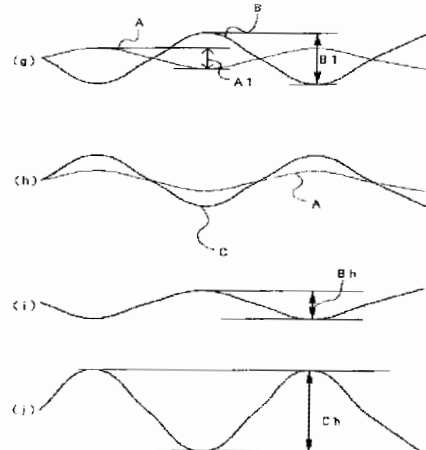
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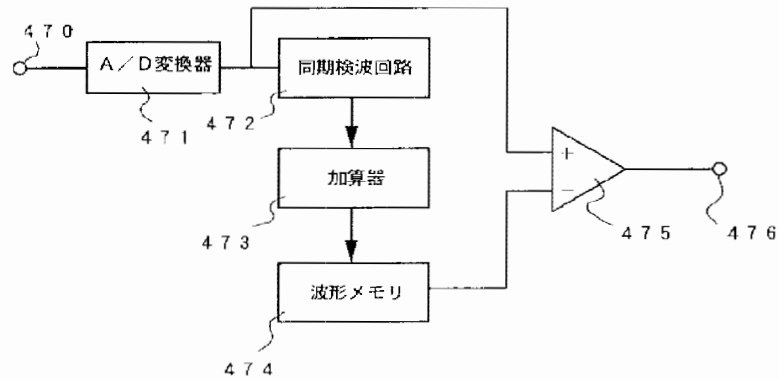
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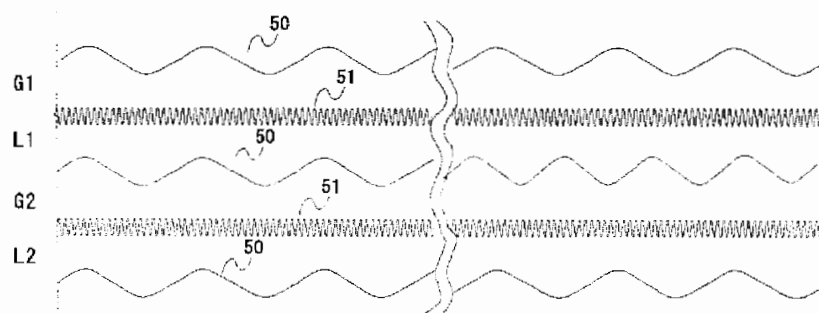
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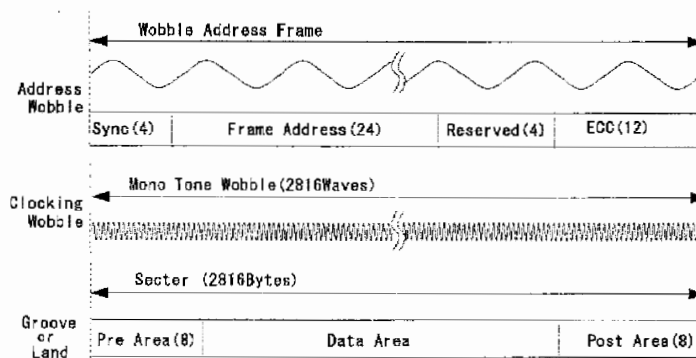
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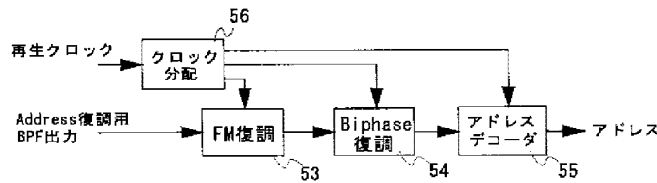
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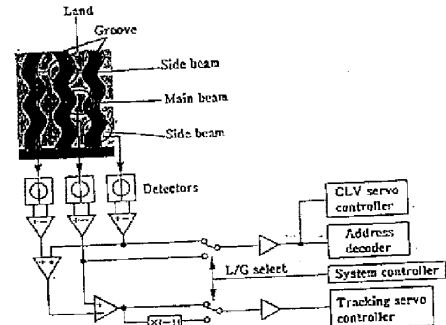
【図35】



【図37】



【図38】



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 (32) 優先日 平成8年11月13日(1996. 11. 13)
 (33) 優先権主張国 日本(JP)
 (31) 優先権主張番号 特願平9-6988
 (32) 優先日 平成9年1月17日(1997. 1. 17)
 (33) 優先権主張国 日本(JP)
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 (32) 優先日 平成9年1月27日(1997. 1. 27)
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 (32) 優先日 平成9年5月29日(1997. 5. 29)
 (33) 優先権主張国 日本(JP)
 (72) 発明者 間宮 昇
 大阪府守口市京阪本通2丁目5番5号 三
 洋電機株式会社内
 (72) 発明者 内原 可治
 大阪府守口市京阪本通2丁目5番5号 三
 洋電機株式会社内
 (72) 発明者 鷺見 聡
 大阪府守口市京阪本通2丁目5番5号 三
 洋電機株式会社内
 (72) 発明者 中尾 賢治
 大阪府守口市京阪本通2丁目5番5号 三
 洋電機株式会社内
 (72) 発明者 日置 敏昭
 大阪府守口市京阪本通2丁目5番5号 三
 洋電機株式会社内
 (72) 発明者 松山 久
 大阪府守口市京阪本通2丁目5番5号 三
 洋電機株式会社内
 (72) 発明者 堀 吉宏
 大阪府守口市京阪本通2丁目5番5号 三
 洋電機株式会社内

Electronic Patent Application Fee Transmittal				
Application Number:		13864249		
Filing Date:		17-Apr-2013		
Title of Invention:		METHOD AND APPARATUS FOR PLAYING AND PROVIDING AUDIO CONTENT IN A SYNCHRONY GROUP		
First Named Inventor/Applicant Name:		Nicholas A. J. Millington		
Filer:		Benjamin Michael Urban		
Attorney Docket Number:		13-0402 (MBHB 15-518-CON)		
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Claims:				
Miscellaneous-Filing:				
Petition:				
Patent-Appeals-and-Interference:				
Post-Allowance-and-Post-Issuance:				
Extension-of-Time:				

Description	Fee Code	Quantity	Amount	Sub-Total in USD(\$)
Miscellaneous:				
Submission- Information Disclosure Stmt	1806	1	180	180
Total in USD (\$)				180

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EFS ID:	27587863
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International Application Number:	
Confirmation Number:	9806
Title of Invention:	METHOD AND APPARATUS FOR PLAYING AND PROVIDING AUDIO CONTENT IN A SYNCHRONY GROUP
First Named Inventor/Applicant Name:	Nicholas A. J. Millington
Customer Number:	107361
Filer:	Benjamin Michael Urban
Filer Authorized By:	
Attorney Docket Number:	13-0402 (MBHB 15-518-CON)
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2	Foreign Reference	JP2001034951.pdf	1088543	no	22
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3	Non Patent Literature	European_No_138086236-1502_Examination_Report_10-24-16.pdf	125370	no	4
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4	Non Patent Literature	13848921_FOA_10-19-2016.pdf	495386	no	14
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5	Non Patent Literature	14184528_OA_11-03-2016.pdf	578767	no	17
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8	Non Patent Literature	13864252_OA_10-05-2016.pdf	327356	no	11
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INFORMATION DISCLOSURE STATEMENT BY APPLICANT (Not for submission under 37 CFR 1.99)	Application Number	13864249
	Filing Date	2013-04-17
	First Named Inventor	MILLINGTON; Nicholas A.J.
	Art Unit	2442
	Examiner Name	SURVILLO, Oleg
	Attorney Docket Number	13-0402

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	Filing Date	2013-04-17
	First Named Inventor	MILLINGTON; Nicholas A.J.
	Art Unit	2442
	Examiner Name	SURVILLO, Oleg
	Attorney Docket Number	13-0402

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	Filing Date	2013-04-17
	First Named Inventor	MILLINGTON; Nicholas A.J.
	Art Unit	2442
	Examiner Name	SURVILLO, Oleg
	Attorney Docket Number	13-0402

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14	Sony: Home Theatre System Operating Instructions for HT-IS100, Copyright 2008, 168 pages.
15	Sony: HT-IS100, 5.1 Channel Audio System, last updated November 2009, 2 pages.
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INFORMATION DISCLOSURE STATEMENT BY APPLICANT (Not for submission under 37 CFR 1.99)	Application Number		13864249
	Filing Date		2013-04-17
	First Named Inventor	MILLINGTON; Nicholas A.J.	
	Art Unit	2442	
	Examiner Name	SURVILLO, Oleg	
	Attorney Docket Number	13-0402	

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¹ See Kind Codes of USPTO Patent Documents at www.USPTO.GOV or MPEP 901.04. ² Enter office that issued the document, by the two-letter code (WIPO Standard ST.3). ³ For Japanese patent documents, the indication of the year of the reign of the Emperor must precede the serial number of the patent document. ⁴ Kind of document by the appropriate symbols as indicated on the document under WIPO Standard ST.16 if possible. ⁵ Applicant is to place a check mark here if English language translation is attached.

INFORMATION DISCLOSURE STATEMENT BY APPLICANT (Not for submission under 37 CFR 1.99)	Application Number	13864249
	Filing Date	2013-04-17
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	Art Unit	2442
	Examiner Name	SURVILLO, Oleg
	Attorney Docket Number	13-0402

CERTIFICATION STATEMENT			
Please see 37 CFR 1.97 and 1.98 to make the appropriate selection(s):			
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A signature of the applicant or representative is required in accordance with CFR 1.33, 10.18. Please see CFR 1.4(d) for the form of the signature.			
Signature	/Benjamin M. Urban/	Date (YYYY-MM-DD)	2016-10-13
Name/Print	Benjamin M. Urban	Registration Number	71,591
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Bibliographic data: CN101292500 (A) — 2008-10-22

Mobile wireless communication terminals, systems, methods, and computer program products for providing a song play list

Inventor(s): THOMAS WASSINGBO [SE] ± (WASSINGBO THOMAS)

Applicant(s): SONY ERICSSON MOBILE COMM AB [SE] ± (SONY ERICSSON MOBILE COMMUNICATIONS AB)

Classification: - **international:** H04L29/08
- **cooperative:** H04M1/72519; H04M1/72558

Application number: CN2006839085 20060816

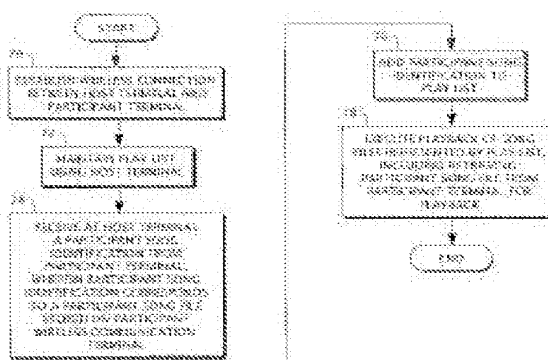
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Also published as: CN101292500 (B) WO2007023120 (A1) US2007049256 (A1) US7555291 (B2) JP2009506603 (A) more

Abstract of CN101292500 (A)

A mobile wireless communication terminal includes a wireless communication module and a controller. The wireless communication module is configured to communicate with other communication terminals over a wireless interface. The controller is configured to establish a connection with a participant wireless communication terminal via the wireless communication module, to maintain a play list representing song files to be played, to receive a participant song identification

from the participant wireless communication terminal, wherein the participant song identification corresponds to a participant song file stored on the participant wireless communication terminal, to add the participant song identification to the play list, and to thereafter execute playback of the song files represented by the play list, including retrieving the participant song file from the participant wireless communication terminal for playback.



[19] 中华人民共和国国家知识产权局

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H04L 29/08 (2006.01)



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[71] 申请人 索尼爱立信移动通讯股份有限公司

地址 瑞典隆德

[72] 发明人 T·瓦兴博

[74] 专利代理机构 中国专利代理(香港)有限公司

代理人 曾祥尧 陈景峻

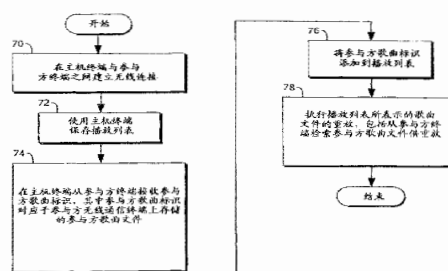
权利要求书 4 页 说明书 17 页 附图 4 页

[54] 发明名称

提供歌曲播放列表的移动无线通信终端、系统、方法和计算机程序产品

[57] 摘要

移动无线通信终端包括无线通信模块和控制器。无线通信模块配置成通过无线接口与其它通信终端进行通信。控制器配置成：经由无线通信模块与参与方无线通信终端建立连接；保存表示要播放的歌曲文件的播放列表；从参与方无线通信终端接收参与方歌曲标识，其中参与方歌曲标识对应于参与方无线通信终端上存储的参与方歌曲文件；将参与方歌曲标识添加到播放列表；以及此后执行播放列表所表示的歌曲文件的重放，包括从参与方无线通信终端检索参与方歌曲文件供重放。



1. 一种移动无线通信终端，包括：

无线通信模块，配置成通过无线接口与其它通信终端进行通信；
以及

控制器，配置成：经由所述无线通信模块与参与方无线通信终端建立连接；保存表示要播放的歌曲文件的播放列表；从所述参与方无线通信终端接收参与方歌曲标识，其中所述参与方歌曲标识对应于所述参与方无线通信终端上存储的参与方歌曲文件；将所述参与方歌曲标识添加到所述播放列表；以及此后执行所述播放列表所表示的所述歌曲文件的重放，包括从所述参与方无线通信终端检索所述参与方歌曲文件供重放。

2. 如权利要求 1 所述的无线通信终端，其中所述控制器配置成：经由所述无线通信模块与多个参与方无线通信终端建立连接；从所述多个参与方无线通信终端接收多个参与方歌曲标识，其中每个参与方歌曲标识对应于所述多个参与方无线通信终端中的相应参与方无线通信终端上存储的相应参与方歌曲文件；将所述参与方歌曲标识中的每个添加到所述播放列表；以及从所述相应参与方无线通信终端中的每个检索所述参与方歌曲文件供重放。

3. 如权利要求 1 或 2 所述的无线通信终端，其中所述控制器配置成：从所述参与方无线通信终端将所述参与方歌曲文件作为流式音乐文件进行检索和执行。

4. 如权利要求 1 或 2 所述的无线通信终端，其中所述控制器配置成：从所述参与方无线通信终端检索所述参与方歌曲文件；在所述无线通信终端上存储所述参与方歌曲文件；以及此后执行所存储参与方歌曲文件的重放。

5. 如以上权利要求中任一项所述的无线通信终端，其中所述控制器配置成向媒体中心发送重放信号，以便在所述媒体中心上重放所述

播放列表所表示的所述歌曲文件。

6. 如以上权利要求中任一项所述的无线通信终端，包括声换能器，其中所述控制器配置成使用所述声换能器来播放所述播放列表所表示的所述歌曲文件。

7. 如以上权利要求中任一项所述的无线通信终端，其中所述控制器配置成向至少一个参与方无线通信终端发送重放信号，以便在所述至少一个参与方无线通信终端上重放所述播放列表所表示的所述歌曲文件。

8. 如以上权利要求中任一项所述的无线通信终端，其中所述控制器配置成：从所述参与方无线通信终端接收表示所述参与方无线通信终端上存储的可用歌曲文件的歌曲目录列表；接收从所述可用歌曲文件选择的用户输入；以及将选择的可用歌曲文件添加到所述播放列表。

9. 如以上权利要求中任一项所述的无线通信终端，其中所述控制器配置成向所述参与方无线通信终端发送表示所述播放列表的列表信号，以便在所述参与方无线通信终端上显示所述播放列表。

10. 如以上权利要求中任一项所述的无线通信终端，其中所述控制器配置成响应用户输入而修改所述播放列表。

11. 如以上权利要求中任一项所述的无线通信终端，其中：

所述无线通信模块配置成通过直接点对点无线接口与其它通信终端进行通信；以及

所述控制器配置成：经由所述无线通信模块与所述参与方无线通信终端建立连接；从所述参与方无线通信终端接收所述参与方歌曲标识；以及全部经由所述直接点对点无线接口从所述参与方无线通信终端检索所述参与方歌曲文件供重放。

12. 如以上权利要求中任一项所述的无线通信终端，其中所述无线通信模块包括短程发射器，并且所述控制器配置成：与所述参与方无线通信终端建立连接；从所述参与方无线通信终端接收所述参与方歌曲标识；以及全部经由所述短程发射器从所述参与方无线通信终端

检索所述参与方歌曲文件供重放。

13. 如权利要求 12 所述的无线通信终端，其中所述短程发射器是蓝牙发射器。

14. 如以上权利要求中任一项所述的无线通信终端，其中所述无线通信终端包括蜂窝电话。

15. 一种用于提供歌曲播放列表的系统，所述系统包括：

主机移动无线通信终端；

参与方无线通信终端；

其中所述主机无线通信终端包括：

无线通信模块，配置成通过无线接口与其它通信终端进行通信；以及

控制器，配置成：经由所述无线通信模块与所述参与方无线通信终端建立连接；保存表示要播放的歌曲文件的播放列表；从所述参与方无线通信终端接收参与方歌曲标识，其中所述参与方歌曲标识对应于所述参与方无线通信终端上存储的参与方歌曲文件；将所述参与方歌曲标识添加到所述播放列表；以及此后执行所述播放列表所表示的所述歌曲文件的重放，包括从所述参与方无线通信终端检索所述参与方歌曲文件供重放。

16. 如权利要求 15 所述的系统，包括多个参与方无线通信终端，并且其中所述控制器配置成：与所述多个参与方无线通信终端中的每个建立连接；从所述多个参与方无线通信终端接收多个参与方歌曲标识，其中每个参与方歌曲标识对应于所述多个参与方无线通信终端中的相应参与方无线通信终端上存储的相应参与方歌曲文件；将所述参与方歌曲标识中的每个添加到所述播放列表；以及从所述相应参与方无线通信终端中的每个检索所述参与方歌曲文件供重放。

17. 一种用于使用主机无线通信终端提供歌曲播放列表的方法，所述方法包括：

在所述主机无线通信终端与参与方无线通信终端之间建立无线连

接;

使用所述主机无线通信终端保存播放列表,所述播放列表表示要播放的歌曲文件;

在所述主机无线通信终端从所述参与方无线通信终端接收参与方歌曲标识,其中所述参与方歌曲标识对应于所述参与方无线通信终端上存储的参与方歌曲文件;

将所述参与方歌曲标识添加到所述播放列表;以及

此后执行所述播放列表所表示的所述歌曲文件的重放,包括从所述参与方无线通信终端检索所述参与方歌曲文件供重放。

18. 一种用于使用主机无线通信终端提供歌曲播放列表的计算机程序产品,所述计算机程序产品包括:

计算机可用介质,其中包含有计算机可用程序代码,所述计算机可用程序代码包括:

配置成在所述主机无线通信终端与参与方无线通信终端之间建立无线连接的计算机可用程序代码;

配置成使用所述主机无线通信终端保存播放列表的计算机可用程序代码,所述播放列表表示要播放的歌曲文件;

配置成在所述主机无线通信终端从所述参与方无线通信终端接收参与方歌曲标识的计算机可用程序代码,其中所述参与方歌曲标识对应于所述参与方无线通信终端上存储的参与方歌曲文件;

配置成将所述参与方歌曲标识添加到所述播放列表的计算机可用程序代码;以及

配置成此后执行所述播放列表所表示的所述歌曲文件的重放、包括从所述参与方无线通信终端检索所述参与方歌曲文件供重放的计算机可用程序代码。

提供歌曲播放列表的移动无线通信终端、系统、方法 和计算机程序产品

技术领域

[001] 本发明涉及电子装置,更具体来说,涉及用于播放歌曲文件的电子装置、方法和计算机程序产品。

背景技术

[002] 移动电子装置如无线通信终端(例如蜂窝电话)广泛用于存储和重放歌曲文件。用户可能希望在另一个装置如媒体中心(例如 PC 或膝上型计算机、电视、音频接收器等)或者另一个无线通信终端上播放歌曲文件。如果其它装置配备了可以是集成模块或辅助(例如插接)模块的适当通信模块,则这可通过经由电缆或无线链路(例如蓝牙连接)将电子装置连接到其它装置来实现。歌曲文件可依次从一个或多个移动电子装置发送到重放装置。

发明内容

[003] 根据本发明的实施例,移动无线通信终端包括无线通信模块和控制器。无线通信模块配置成通过无线接口与其它通信终端进行通信。控制器配置成:经由无线通信模块与参与方无线通信终端建立连接;保存表示要播放的歌曲文件的播放列表;从参与方无线通信终端接收参与方歌曲标识,其中参与方歌曲标识对应于参与方无线通信终端上存储的参与方歌曲文件;将参与方歌曲标识添加到播放列表;以及此后执行播放列表所表示的歌曲文件的重放,包括从参与方无线通信终端检索参与方歌曲文件供重放。

[004] 控制器可配置成:与多个参与方无线通信终端建立连接;从

多个参与方无线通信终端接收多个参与方歌曲标识, 其中每个参与方歌曲标识对应于多个参与方无线通信终端中的相应参与方无线通信终端上存储的相应参与方歌曲文件; 将参与方歌曲标识中的每个添加到播放列表; 以及从相应参与方无线通信终端中的每个检索参与方歌曲文件供重放。

[005] 根据一些实施例, 控制器配置成从所述参与方无线通信终端将所述参与方歌曲文件作为流式音乐文件进行检索和执行。

[006] 根据一些实施例, 控制器配置成: 从参与方无线通信终端检索参与方歌曲文件; 在无线通信终端上存储参与方歌曲文件; 以及此后执行所存储参与方歌曲文件的重放。

[007] 控制器可配置成向媒体中心发送用于在媒体中心重放播放列表所表示的歌曲文件的重放信号。

[008] 根据一些实施例, 无线通信终端包括声换能器。控制器配置成使用声换能器来播放由播放列表所表示的歌曲文件。

[009] 控制器可配置成向至少一个参与方无线通信终端发送重放信号, 用于在至少一个参与方无线通信终端上重放播放列表所表示的歌曲文件。

[010] 控制器可配置成: 从参与方无线通信终端接收表示参与方无线通信终端上存储的可用歌曲文件的歌曲目录列表; 接收从可用歌曲文件选择的用户输入; 以及将选择的可用歌曲文件添加到播放列表。

[011] 根据一些实施例, 控制器配置成向参与方无线通信终端发送表示播放列表的列表信号, 用于在参与方无线通信终端显示播放列表。

[012] 控制器可配置成响应用户输入而修改播放列表。

[013] 根据一些实施例, 无线通信模块配置成通过直接点对点无线接口与其它通信终端进行通信。控制器配置成: 与参与方无线通信终端建立连接; 从参与方无线通信终端接收参与方歌曲标识; 以及全部经由直接点对点无线接口从参与方无线通信终端检索参与方歌曲文件供重放。

[014] 根据一些实施例, 无线通信模块包括短程发射器。控制器配置成: 与参与方无线通信终端建立连接; 从参与方无线通信终端接收参与方歌曲标识; 以及全部经由短程发射器从参与方无线通信终端检索参与方歌曲文件供重放。短程发射器可以是蓝牙™发射器。

[015] 无线通信终端可包括蜂窝电话。

[016] 根据本发明的其它实施例, 一种用于提供歌曲播放列表的系统包括主机移动无线通信终端和参与方无线通信终端。主机无线通信终端包括无线通信模块和控制器。无线通信模块配置成通过无线接口与其它通信终端进行通信。控制器配置成: 经由无线通信模块与参与方无线通信终端建立连接; 保存表示要播放的歌曲文件的播放列表; 从参与方无线通信终端接收参与方歌曲标识, 其中参与方歌曲标识对应于参与方无线通信终端上存储的参与方歌曲文件; 将参与方歌曲标识添加到播放列表; 以及此后执行播放列表所表示的歌曲文件的重放, 包括从参与方无线通信终端检索参与方歌曲文件供重放。

[017] 根据一些实施例, 该系统包括多个参与方无线通信终端。控制器配置成: 与多个参与方无线通信终端中的每个建立连接; 从多个参与方无线通信终端接收多个参与方歌曲标识, 其中每个参与方歌曲标识对应于多个参与方无线通信终端中的相应参与方无线通信终端上存储的相应参与方歌曲文件; 将参与方歌曲标识中的每个添加到播放列表; 以及从相应参与方无线通信终端中的每个检索参与方歌曲文件供重放。

[018] 根据本发明的其它实施例, 用于使用主机无线通信终端提供歌曲播放列表的方法包括: 建立主机无线通信终端与参与方无线通信终端之间的无线连接; 使用主机无线通信终端保存播放列表, 该播放列表表示要播放的歌曲文件; 在主机无线通信终端从参与方无线通信终端接收参与方歌曲标识, 其中参与方歌曲标识对应于参与方无线通信终端上存储的参与方歌曲文件; 将参与方歌曲标识添加到播放列表; 以及此后执行播放列表所表示的歌曲文件的重放, 包括从参与方无线

通信终端检索参与方歌曲文件供重放。

[019] 根据本发明的其它实施例,用于使用主机无线通信终端提供歌曲播放列表的计算机程序产品包括其中包含有计算机可用程序代码的计算机可用介质,计算机可用程序代码包括:配置成在主机无线通信终端与参与方无线通信终端之间建立无线连接的计算机可用程序代码;配置成使用主机无线通信终端保存播放列表的计算机可用程序代码,播放列表表示要播放的歌曲文件;配置成在主机无线通信终端从参与方无线通信终端接收参与方歌曲标识的计算机可用程序代码,其中参与方歌曲标识对应于参与方无线通信终端上存储的参与方歌曲文件;配置成将参与方歌曲标识添加到播放列表的计算机可用程序代码;以及配置成此后执行播放列表所表示的歌曲文件的重放、包括从参与方无线通信终端检索参与方歌曲文件供重放的计算机可用程序代码。

[020] 通过阅读以下附图和优选实施例的详细描述,本领域的技术人员会了解本发明的其它特征、优点和细节,这种描述只是对本发明的说明。

附图说明

[021] 图 1 是根据本发明一些实施例的移动无线通信终端以及示范基站收发器的示意图。

[022] 图 2 是示出根据本发明一些实施例的方法的流程图。

[023] 图 3 是根据本发明一些实施例、包括图 1 移动无线通信终端的歌曲文件重放系统的示意图。

[024] 图 4 是示出根据本发明一些实施例、图 1 移动无线通信终端的显示的示意图。

[025] 图 5 是根据本发明的其它实施例、包括图 1 移动无线通信终端的歌曲文件重放系统的示意图。

[026] 图 6 是根据本发明的其它实施例、包括图 1 移动无线通信终端的歌曲文件重放系统的示意图。

具体实施方式

[027] 现在将参照示出本发明实施例的附图更全面地描述本发明。但是，本发明不应当理解为局限于本文所述的实施例。相反，提供这些实施例，使得本公开是全面和完整的，并向本领域的技术人员全面地传达本发明的范围。相似的标号始终表示相似的要素。

[028] 本文所使用的术语“包括”或“包含”是开放式的，并且包括一个或多个所述特征、整数、要素、步骤、组件或功能，但并不排除存在或增加一个或多个其它特征、整数、要素、步骤、组件、功能或者它们的集合。

[029] 本文所使用的术语“和/或”包括关联的所列项中的一个或多个的任意一个和所有组合。

[030] 本文所使用的派生于拉丁短语“*exempli gratia*”的常见缩写词“例如”可用来介绍或详细说明先前所述项的一般实例或多个实例，而不是要限制这种项。在本文使用时，派生于拉丁短语“*id est*”的常见缩写词“即”可用于从更一般的叙述中详细说明某个特定项。

[031] 本文所使用的术语仅用于描述具体实施例的目的，而不是要限制本发明。本文所使用的单数形式“一个”和“该”意在也包含复数形式，除非上下文有明确说明。

[032] 除另有定义之外，本文所用的所有术语(包括科技术语)都具有与本发明所属领域的技术人员通常理解的同样的含义。还要理解，例如常用词典中定义的那些术语应当被认为具有与它们在相关领域的上下文中的含义一致的含义，并且不会以理想化或过分正式的意义来解释，除非本文中有明确定义。

[033] 要理解，当一个元件被称作“耦合到”或“连接到”另一个元件时，它可以直接耦合或连接到其它元件，或者也可存在中介元件。相反，当一个元件被说成是“直接耦合到”或者“直接连接到”另一个元件时，则不存在中介元件。此外，本文所使用的“耦合”或者“连接”可包括无线耦合或连接。

[034] 为了简洁和/或清楚起见,可能没有详细描述众所周知的功能或构造。

[035] 本发明可实施为方法、电子装置和/或计算机程序产品。因此,本发明可以硬件和/或软件(包括固件、常驻软件、微码等)实施,它们在本文一般称作“电路”或“模块”。此外,本发明可采取计算机可用或计算机可读存储介质上的计算机程序产品的形式,所述介质中包含供指令执行系统使用或与其配合使用的计算机可用或计算机可读程序代码。在本文档的上下文中,“计算机可用或计算机可读介质”可以是可包含、存储、传递、传播或传送供指令执行系统、设备或装置使用的或者与其结合使用的程序的任何介质。

[036] 参照方法和通信终端的框图和/或可操作图示来描述根据本发明的实施例。在这方面,每个框可表示代码的模块、段或部分,它包含用于实现指定逻辑功能的一个或多个可执行指令。要理解,框图和/或可操作图示的每个框、框图和/或可操作图示中的框组合可通过射频、模拟和/或数字硬件和/或程序指令来实现。这些程序指令可提供给控制器,该控制器可包括一个或多个通用处理器、专用处理器、ASIC和/或其它可编程数据处理装置,使得经由控制器和/或其它可编程数据处理装置执行的指令创建用于实现框图和/或可操作框或多个框中指定的功能/动作的部件。在一些备选实现中,框中所示的功能/动作可不按照可操作图示中所述的顺序发生。例如,连续示出的两个框实际上可以基本上同时执行,或者这些框有时可按照相反顺序执行,取决于所涉及的功能/动作。

[037] 这些计算机程序指令还可存储在计算机可用或计算机可读存储器,其可指示计算机或其它可编程数据处理装置以特定方式运行,使得计算机可用或计算机可读存储器中存储的指令产生一种制品,其中包含实现在流程图和/或框图框或多个框中规定的功能的指令。

[038] 计算机可用或计算机可读介质例如但非限制地可以是电、磁、光、电磁、红外线或半导体系统、设备、装置或者传播介质。计

计算机可读介质的更具体实例(非详尽的列表)包括下列各项: 硬盘、光存储装置、例如支持因特网或内联网的那些传输介质、磁存储装置、具有一条或多条导线的电连接、便携式计算机盘、随机存取存储器(RAM)、只读存储器(ROM)、可擦除可编程只读存储器(EPROM 或闪存速存储器)、光纤以及只读光盘存储器(CD-ROM)。

[039] 用于执行本发明操作的计算机程序代码可通过例如 Java®、Smalltalk 或 C++等面向对象的编程语言来编写。但是, 用于执行本发明操作的计算机程序代码也可通过例如“C”编程语言和/或低级汇编语言等传统面向过程的编程语言来编写。还要理解, 任何或所有程序模块的功能性也可采用分立硬件组件、一个或多个专用集成电路(ASIC)或编程的数字信号处理器或微控制器来实现。

[040] 本文所使用的“电子组件”是指有源器件, 与无源电连接器等相反。电子组件可包括处理器。

[041] 本文所使用的“流式传输”或“流式”是指例如音频或歌曲文件等文件经由数字信号连续发送到接收装置, 在其中经由适当的接收应用程序同时播放该音频或歌曲文件。数字信号通常被缓冲。

[042] 本文所使用的“通信终端”包括但不限于配置成经由例如公共交换电话网(PSTN)、数字用户线(DSL)、数字电缆或另一个数据连接/网络等有线连接和/或经由与例如蜂窝网络、卫星网络、无线局域网(WLAN)和/或另一个通信终端的无线接口接收/发送通信信号的终端。

[043] 当通信终端配置成通过无线接口进行通信时, 在本文称作“无线通信终端”或“无线终端”。无线终端的实例包括但不限于蜂窝电话、个人数据助理(PDA)、寻呼机和/或配置成通过无线通信接口传递数据的计算机, 无线通信接口可包括蜂窝电话接口、蓝牙接口、无线局域网接口(例如 802.11)、另一个 RF 通信接口和/或光/红外通信接口。

[044] 本文所使用的“移动终端”可以是便携式、可移动、安装在(基于航空、航海或陆地的)交通工具中, 或者设置成和/或配置成本地操作和/或以分布方式在地球上和/或空间的任何其它位置进行操作。

[045] 下面结合图 1-6 来描述本发明的一些实施例。本发明的一些实施例提供能够提供歌曲播放列表的移动无线通信终端，该歌曲播放列表包含来自另一个无线通信终端的一个或多个歌曲标识。

[046] 现在参照图 1，说明根据本发明一些实施例的示范移动无线通信终端 10A。无线通信终端 10A 配置成通过其间的直接无线通信接口、通过经由一个或多个蜂窝基站的另一个无线通信接口和/或通过经由无线局域网(WLAN)路由器的另一个无线通信接口与一个或多个其它无线通信终端进行数据通信。

[047] 无线通信终端 10A 可以是组成无线电话通信系统 2 的一部分的移动无线电话，如图 1 所示。系统 2 包括移动无线通信终端 10A 以及作为无线通信网络 5 组成部分的基站收发器。在本发明的一些实施例中，网络 5 包括基站收发器，该基站收发器包括定义蜂窝网络中各个小区的无线电收发器，并使用无线电链路协议与移动终端 10A(经由接口 7)和小区中的其它移动终端进行通信。要理解，在本发明的一些实施例中，许多基站收发器可通过例如移动交换中心和其它装置进行连接，以便定义无线通信网络 5。

[048] 所示实施例中的移动终端 10A 包括便携壳体组件 12、显示器 20、包括扬声器 24(即声换能器)的用户接口 22(即人机接口(MMI))、控制器 30、通信模块 32 以及存储器 34。移动终端 10A 的上述部件可包含在许多常规移动终端中，并且它们的功能性是本领域技术人员普遍知道的。移动终端 10A 还包括可存储在存储器 34 中的歌曲管理模块 40。

[049] 显示器 20 可以是任何适当的显示屏组件。例如，显示屏 28 可以是具有或没有辅助照明(例如照明面板)的液晶显示器(LCD)。

[050] 用户接口 22 可包括任何适当的输入装置，例如包括触控或触敏装置(例如触摸屏)、操纵杆、键盘/小键盘、拨号盘、方向键和/或指示装置(例如鼠标、跟踪球、触摸垫等)。扬声器 24 响应输入音频信号而产生声音。用户接口 22 还可包括耦合到音频处理器的扩音器，该

音频处理器配置成响应传入扩音器的声音而产生音频数据流。

[051] 控制器 30 可支持移动终端 10A 的各种功能。例如, 控制器 30 可以是任何市场有售的或定制的微处理器。在使用中, 移动终端 10A 的控制器 30 在显示器 20 上产生显示图像。

[052] 存储器 34 配置成存储数字信息信号和数据, 例如数字音频信号和/或数字音频文件。

[053] 通信模块 32 配置成通过一个或多个无线接口(例如本文所述的无线接口 7、112、114、116、122 和 134(图 1 和图 3))向另一个远程无线通信终端传递数据, 如本文所述。通信模块 32 可包括蜂窝通信模块、直接点对点连接模块和/或 WLAN 模块。

[054] 用蜂窝通信模块, 无线通信终端 10A 可采用例如先进移动电话服务(AMPS)、ANSI-136、全球移动通信标准(GSM)、通用分组无线电业务(GPRS)、GSM 演进增强数据速率(EDGE)、码分多址(CDMA)、宽带 CDMA、CDMA2000 和通用移动通信系统(UMTS)等一个或多个蜂窝通信协议经由网络 5 的基站进行通信。蜂窝基站可连接到移动电话交换局(MTSO)无线网络, 它又可连接到 PSTN 和/或另一个网络。

[055] 直接点对点连接模块可包括直接 RF 通信模块或直接 IR 通信模块。直接 RF 通信模块可包括蓝牙模块。用蓝牙模块, 无线通信终端 10A 可经由自组网络通过直接点对点接口进行通信。

[056] 用 WLAN 模块, 无线通信终端 10A 可通过 WLAN(例如路由器 120(图 3))采用可包括但不限于 802.11a、802.11b、802.11e、802.11g 和/或 802.11i 等通信协议进行通信。

[057] 通信模块 32 可包括通常具有发射器电路和接收器电路的收发器, 发射器电路和接收器电路经由天线分别(例如向网络 5、路由器或直接向另一个终端)发送出局射频信号和(例如从网络 5、路由器或直接从另一个终端)接收入局射频信号、如语音和数据信号。通信模块 32 可包括短程发射器和接收器, 例如蓝牙发射器和接收器。天线可以是嵌入式天线、收缩式天线或者本领域技术人员已知的任何天线, 并没

有背离本发明的范围。在移动终端 10A 与网络 5、路由器或其它终端之间发送的射频信号可包括业务和控制信号(例如寻呼信号/入局呼叫的消息),它们用来与另一方或目的地建立和保持通信。射频信号还可包括分组数据信息,例如蜂窝数字分组数据(CDPD)信息。另外,收发器可包括配置成经由 IR 端口向/从其它电子装置发送/接收红外信号的红外(IR)收发器。

[058] 移动终端 10A 还可配置成经由有线或电缆与另一个终端电耦合,以便在其间传输数字通信信号。移动终端 10A 可包括其它组件,例如配置成根据入射光生成静止图像和/或视频数据流的相机装置。

[059] 根据一些实施例,移动终端 10A 是手持移动终端。“手持移动终端”是指移动终端的外部尺寸适于且适合由典型操作者单手使用。根据一些实施例,手持移动终端 10A 的总体积小于大约 200 立方厘米。根据一些实施例,手持终端 10A 的总体积小于大约 100 立方厘米。根据一些实施例,手持移动终端 10A 的总体积在大约 50 立方厘米与 100 立方厘米之间。根据一些实施例,手持移动终端 10A 的尺寸不超过大约 200 毫米。

[060] 现在参照图 2 的流程图来描述用于使用根据本发明一些实施例的无线通信终端(例如移动终端 10A)提供歌曲播放列表的方法。参照图 2 的实施例,该方法包括在主机无线通信终端与参与方无线通信终端之间建立无线连接(框 70)。使用主机无线通信终端来保存播放列表(框 72)。播放列表表示要播放的歌曲文件。在主机无线通信终端接收来自参与方无线通信终端的参与方歌曲标识(框 74)。参与方歌曲标识对应于或者指定参与方无线通信终端上存储的参与方歌曲文件。将参与方歌曲标识添加到播放列表(框 76)。此后,执行播放列表所表示的歌曲文件的重放(框 78)。执行歌曲文件的重放包括从参与方无线通信终端检索参与方歌曲文件供重放。

[061] 根据一些实施例,在主机无线通信终端与多个参与方无线通信终端之间建立连接,并且主机无线通信终端从多个参与方无线通信

终端接收多个参与方歌曲标识,将参与方歌曲标识中的每个添加到播放列表,并从相应参与方无线通信终端中的每个检索参与方歌曲文件供重放。根据一些实施例,主机无线通信终端与参与方无线通信终端之间的通信经由直接点对点接口、如蓝牙无线 RF 连接来实现。根据一些实施例,主机无线通信终端与参与方无线通信终端之间的通信经由间接接口、例如通过 WLAN 或基于蜂窝的系统来实现。通过以下对其它实施例的描述,本发明的其它方面和实施例将显而易见。

[062] 参照图 3,其中示出根据本发明实施例的歌曲文件重放系统 100A。系统 100A 包括移动无线通信终端 10A(在本文又称作“主机无线通信终端”或“主机终端”)、多个附加的参与方无线通信终端 10B、10C、10D 和 10E(在本文又称作“参与方无线通信终端”或“参与方终端”)以及媒体中心 130。

[063] 终端 10B-E 可按照以上对于终端 10A 所述的方式来配置。终端 10B-E 各包括无线通信模块 32 和歌曲管理模块 40。但是,可根据装置的预期功能性,对于各终端 10A-E 以不同方式来配置相应的歌曲管理模块。根据一些实施例,终端 10B-E 中的所有或一些是移动无线通信终端。根据一些实施例,终端 10A-E 中的所有或一些是手持移动无线通信终端。

[064] 媒体中心 130 包括适合于将音频信号(例如数字音频信号)转换为对应声音的装置。媒体中心 130 可包括例如 PC 或膝上型计算机、电视机、音频接收器、立体声设备等。所示的媒体中心 130 包括无线通信模块 132 和声换能器(扬声器)136。无线通信模块 132 可与媒体中心 130 集成在一起,或者可以是辅助或插接的无线通信模块,例如蓝牙媒体查看器 MMV-100 模块或蓝牙媒体查看器 MMV-200 模块,均可从瑞典索尼爱立信购买。

[065] 现在采用用途的说明性实例来描述系统 100A。聚会的一组用户各具有终端 10A-E 中的相应一个。各种歌曲存储在终端 10A-E 的一个或多个(通常两个或更多)上。用户可能希望通过在媒体中心 130

播放他们的歌曲,来与聚会的其他用户共享其移动终端上存储的歌曲。用已知的可用技术,这可通过一次一个让各用户(经由蓝牙等)向媒体中心 130 一首一首地发送他或她的歌曲(即音乐音频文件)来实现。但是,这种方法需要来自每个有贡献用户的大量且积极的交互作用。这种牵连可能是不方便的,和/或无助于环境(例如聚会)。

[066] 根据本发明的实施例,通过使用主机终端 10A 创建和保存歌曲的播放列表并执行歌曲(即音乐音频文件)的重放,可以克服上述缺点。通常,各终端 10A-E 中安装了适当的歌曲管理模块 40(例如软件或固件应用程序)。

[067] 将主机终端 10A 选择为主或主机终端,而参与方终端 10B-E 各指定为从或参与方终端。主机终端 10A 的控制器 30 与参与方终端 10B-E 中的每个建立无线连接(即在通信上耦合)。根据一些实施例,无线连接是直接无线通信接口连接,根据一些实施例是直接 RF 连接,而根据一些实施例是蓝牙连接。可通过已知方式来建立连接。例如,在蓝牙连接的情况下,参与方(从)终端 10B-E 可设置成可搜索/可发现状态,而主机(主)终端 10A 发出识别可用参与方(从)终端 10B-E 的查询,从而形成微微网或个人区域网络。

[068] 如图所示,主机终端 10A 分别经由直接点对点无线接口或链路 112、114 和 116 与参与方终端 10B、10C 和 10D 进行通信,并经由无线接口或链路 118、122 通过 WLAN 路由器 120 与参与方终端 10E 进行通信。要理解,根据一些实施例,接口可以是直接(例如蓝牙)或间接(例如经由 WLAN 路由器或蜂窝网络 5)的无线接口的各种组合。根据一些预期的实施例,在主机终端 10A 与参与方终端 10B-E 之间提供的执行歌曲共享过程的所有信号均经由直接点对点无线接口来提供。根据一些预期的实施例,在主机终端 10A 与参与方终端之间提供的执行歌曲共享过程的所有信号均经由直接无线射频(RF)接口、如蓝牙接口来提供。

[069] 在建立连接之前或之后,主机终端 10A 可通过任何适当的

方法来识别和指定有效的参与方终端。根据一些实施例,主机终端 10A 将对所有或选择的一组潜在参与方终端(例如主机终端的蓝牙信号工作范围内的那些)进行轮询,或向其发送邀请。潜在参与方终端然后通过以接受邀请的方式进行应答来决定参加。作为替代或补充,主机终端 10A 可接受来自参与方终端要求参加的自发请求。主机终端 10A 可用来定义特定的组,例如将被准许参加(即,向播放列表贡献歌曲)的潜在参与方终端的列表。

[070] 主机终端 10A 在通信上还耦合到媒体中心 130,以便经由无线接口或链路 134 向通信模块 132 提供无线歌曲数据信号。或者,主机终端 10A 可使用有线或信号电缆连接到媒体中心 130。

[071] 一旦在主机终端 10A 与参与方终端 10B-E 之间建立了连接,每个参与方就可发送指定参与方终端 10B-E 上存储的歌曲的参与方歌曲标识。发送参与方终端 10B-E 的歌曲管理模块 40 可提供用户接口以帮助选择和发送歌曲标识。例如,模块 40 可生成可用所存储歌曲的列表以及用于选择歌曲发送到主机终端 10A 的复选框。

[072] 主机终端 10A 从参与方终端 10B-E 接收参与方歌曲标识。主机终端 10A 的歌曲管理模块 40 创建和保存这些歌曲标识的主机播放列表。主机播放列表表示要播放的歌曲文件。

[073] 每个歌曲管理模块 40 在关联的终端 10A-E 上提供显示 150,如图 4 所示。在主机终端 10A 上提供的显示可不同于在参与方终端 10B-E 上提供的显示。例如,可从参与方终端显示中省略显示 150 中与管理功能性相关的部分。或者,歌曲管理模块 40 可以不在参与方终端 10B-E 上提供显示。

[074] 在所示的实施例中,显示 150 包括播放列表字段 152 和控制按钮 154(例如软键)。控制按钮 154 通常仅显示在主机终端显示器上。在播放列表字段 152 显示播放列表 156。播放列表 156 包括一系列一行一行显示的歌曲标识 156A,各对应于相应的歌曲文件。如图所示,每个歌曲标识包括歌曲艺术家、歌曲名称和对应于所关联歌曲文件的

歌曲播放时间，以及提交了歌曲标识的参与方标识。要理解，也可提供或多或少和/或不同的信息。

[075] 例如，主机可使用控制按钮 154 有选择地管理歌曲列表 156 所表示的歌曲的播放。例如，“播放”和“停止”按钮可用于开始和停止向媒体中心 130 发送歌曲文件。“移动”按钮可用于对播放列表 156 中的歌曲标识 156A 确定优先顺序，或改变顺序(即改变将播放所关联歌曲文件的顺序)。“删除”按钮可用于从播放列表 156 中删除歌曲标识 156A。“详情”按钮可用于从播放列表 156 中删除歌曲标识 156A，使得不会播放所关联歌曲文件。“随机混合”按钮可用于使主机终端 10A 以随机或其它非连续顺序来执行歌曲文件的重放(即，向媒体中心 130 发送歌曲文件)。通常，主机终端 10A 将以所列的顺序来执行歌曲文件的重放，除非正在实行随机或其它非连续模式。

[076] 还可提供其它功能性。例如，主机终端 10A 可允许用户根据一个或多个选择的属性(例如有贡献的参与方、歌曲长度等)来改变歌曲标识 156A 的顺序。主机终端 10A 可配置成指定准许参与方终端 10B-E 中的哪一些向播放列表 156 添加歌曲标识。除了添加歌曲标识之外，还可使参与方终端 10B-E 能够编辑或修改歌曲列表 156。在这种情况下，主机终端 10A 可配置成允许主机选择授权或不授权参与方终端 10B-E 中的哪一些修改歌曲播放列表 156(即，启用和禁用参与方终端的播放列表编辑能力)。

[077] 当主机终端 10A 处于播放模式时，主机终端 10A 的控制器 30 将识别此后要播放的歌曲标识 156A。所识别的歌曲标识可以是接下来要播放的歌曲标识。然后，主机终端 10A 将向参与方终端 10B-E 中对应的一个发送歌曲文件请求，以便检索对应于该歌曲标识的歌曲文件。

[078] 然后，对应的参与方终端将向主机终端 10A 发送所标识歌曲文件(它存储在参与方终端上)的副本。根据一些实施例，参与方终端向主机终端 10A 发送歌曲文件，作为然后暂时存储在主机终端

10A(例如在存储器 34 中)的歌曲文件的副本。根据其它实施例,参与方终端将歌曲文件作为流式歌曲文件发送到主机终端 10A。根据一些实施例,主机终端与参与方终端之间的请求和歌曲文件信号以无线方式发送,根据一些实施例,经由直接无线 RF 连接、如蓝牙进行发送。

[079] 然后,主机终端 10A 的控制器 30 通过向媒体中心 130 发送歌曲文件,来执行所检索歌曲文件的重放。歌曲文件可存储在媒体中心 130 供重放。或者,例如当歌曲文件从始发参与方终端流式传输到主机终端 10A 时,歌曲文件可流式传输到媒体中心 130。可能优选或必要的是,如果主机终端 10A 上的所分配存储空间不够,和/或在主机终端 10A 上存储歌曲文件被阻止或者是非法的,则将歌曲文件从参与方终端流式传输到主机终端,和/或从主机终端流式传输到媒体中心 130。根据一些实施例,无论歌曲文件是流式传输的还是整个存储的,都将在重放期间或之后从主机终端 10A 自动删除(即无需用户干预)。

[080] 主机终端 10A 的控制器 30 可依次地自动(即无需用户干预)对每个歌曲标识 156A 执行上述步骤。当主机终端 10A 以这种方式遍历歌曲列表 156 时,主机可根据需要修改播放列表 156。另外,根据一些实施例,当主机终端 10A 的控制器 30 遍历播放列表 156 时,参与方终端 10B-E 可向歌曲列表 156 添加新的歌曲标识。根据一些实施例,当主机终端 10A 遍历播放列表 156 时,新的参与方终端可加入该组并提交歌曲标识。

[081] 系统 100A 可提供用于播放一群人选择的歌曲的便利而有趣的机制。系统 100A 提供重放执行功能性,并且还允许主机终端(在某些情况下是参与方终端)创建、协调、控制和/或管理歌曲播放列表。实际上,例如,聚会常客可将歌曲从他们自己的参与方终端添加到主机终端(例如聚会主人自己的移动终端)上保存的歌曲播放列表。在重放序列中轮到他们的歌曲时,歌曲文件的副本将自动发送给主机终端并播放。这样,聚会常客无需采取其它任何动作。根据一些实施例,当请求和歌曲传送信号经由直接无线通信(例如 RF 接口、如蓝牙连接)

在主机终端与参与方终端之间传递时，聚会常客可只将他或她的参与方终端放入口袋或手袋中。在执行该过程时，可添加或删除参与方。例如，当参与方离开蓝牙范围时，他或她的歌曲可从播放列表中删除或者向下移动。

[082] 现在参照图 5，其中示出根据本发明其它实施例的系统 100B，它包括主机终端 10A 和参与方终端 10B-E。系统 100B 与系统 100A(图 2)的不同之处在于，省略了媒体中心 130，并且来自歌曲播放列表 156 的歌曲文件重放经由主机终端 10A 的扬声器 24 进行。

[083] 现在参照图 6，其中示出根据本发明其它实施例的系统 100C，它包括主机终端 10A 和参与方终端 10B-E。系统 100C 与系统 100B(图 5)的不同之处在于，来自歌曲播放列表 156 的歌曲文件重放经由主机终端 10A 的扬声器 24 进行，并且另外还通过参与方终端 10B-E 的相应扬声器和/或通过在上操作上连接到参与方终端 10B-E 的一个或多个耳机(例如耳式立体声耳塞)进行。更具体来说，主机终端 10A 的控制器 30 通过与以上所述相同的方式播放每个歌曲文件，并且还向参与方终端 10B-E 发送歌曲文件供同时重放。这样，主机终端 10A 创建和管理可包含来自参与方终端 10B-E 的歌曲标识的歌曲播放列表，根据需要从适当的参与方终端 10B-E 检索歌曲文件的副本，并向参与方终端 10B-E 分发歌曲文件的副本供重放。根据一些实施例，歌曲传送信号以无线方式从主机终端 10A 发送到参与方终端 10B-E，而根据一些实施例，经由直接无线 RF 连接、如蓝牙发送。根据一些实施例，将歌曲文件流式传输到参与方终端 10B-E。主机终端 10A 可配置成准许主机指定参与方终端 10B-E 中的哪一些将接收重放信号，以及参与方终端 10B-E 中的哪一些被授权向播放列表 156 提交歌曲标识，这两组可以不同。

[084] 主机终端 10A 可配置成允许用户从上述各种选项有选择地选取用于重放的装置或装置组合。例如，主机终端 10A 可配置成仅经由主机终端 10A 的扬声器 24、经由主机终端 10A 的扬声器和参与方终

端 10B-E 的扬声器、经由参与方终端 10B-E 的扬声器和媒体中心 130 等来执行重放。要理解，当歌曲文件经由参与方终端 10B-E 的扬声器重放时，参与方终端中的一个给定终端的用户可禁用其扬声器，和/或可中继该歌曲文件以便在另一个装置(例如关联的另一个媒体中心等)上重放。

[085] 根据其它实施例，主机终端 10A 的控制器 30 可请求和/或接受从参与方终端主动提供的歌曲目录列表。歌曲目录列表包括表示参与方终端上每个歌曲文件的歌曲标识或者参与方希望可得到的所有这类歌曲文件的歌曲标识。此后，主机终端 10A 可使主机能够选择主机希望添加到播放列表的歌曲目录列表的歌曲标识。

[086] 本文所述的应用程序，包括歌曲管理模块 40，说明实现根据本发明实施例的各种特征的程序。要理解，根据本发明的实施例，可采用其它和/或附加应用程序。

[087] 虽然图 1 示出可在移动终端和/或其它电子装置中用于管理和重放歌曲文件的示范硬件/软件体系结构，但要理解，本发明不限于这样一种配置，而是要含盖能够执行本文所述操作的任何配置。例如，虽然存储器 34 示为与控制器 30 分开，但存储器 34 或者其若干部分也可视为控制器 30 的组成部分。更一般来说，虽然通过图解以具体的块示出了具体功能性，但不同块和/或其若干部分的功能性可被组合、划分和/或消除。此外，按照本发明的各种实施例，图 1 的硬件/软件体系结构的功能性可实现为单处理器系统或者多处理器系统。

[088] 给出了本公开的好处，本领域技术人员可做出许多改变和修改，并不背离本发明的精神和范围。因此必须理解，只是为了举例而阐述了所示实施例，而不应视为限制由以下权利要求书所定义的本发明。因此，以下权利要求书将理解为不仅包括字面阐述的要素的组合，而且还包括用于以基本相同方式执行基本相同功能以获得基本相同结果的所有等效要素。权利要求书因而要理解为包括以上具体说明和描述的、概念上等效的以及结合了本发明实质概念的内容。

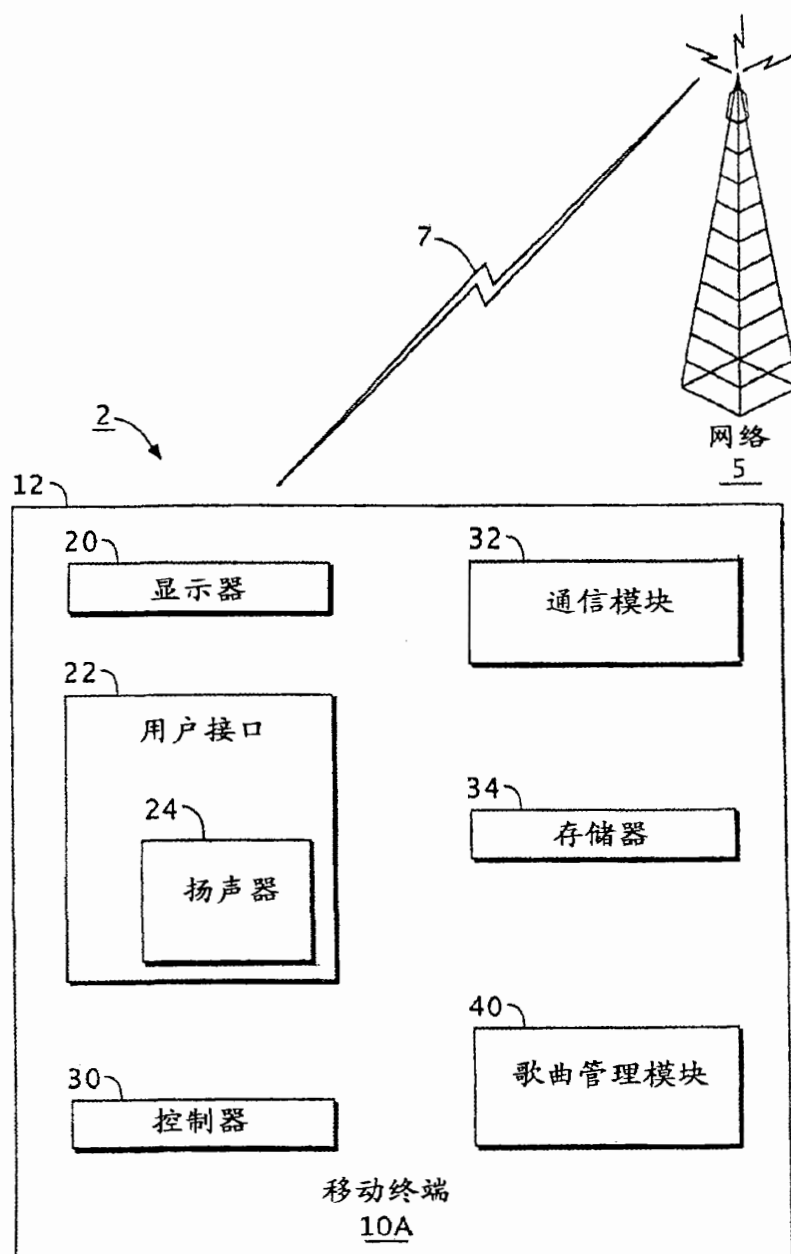
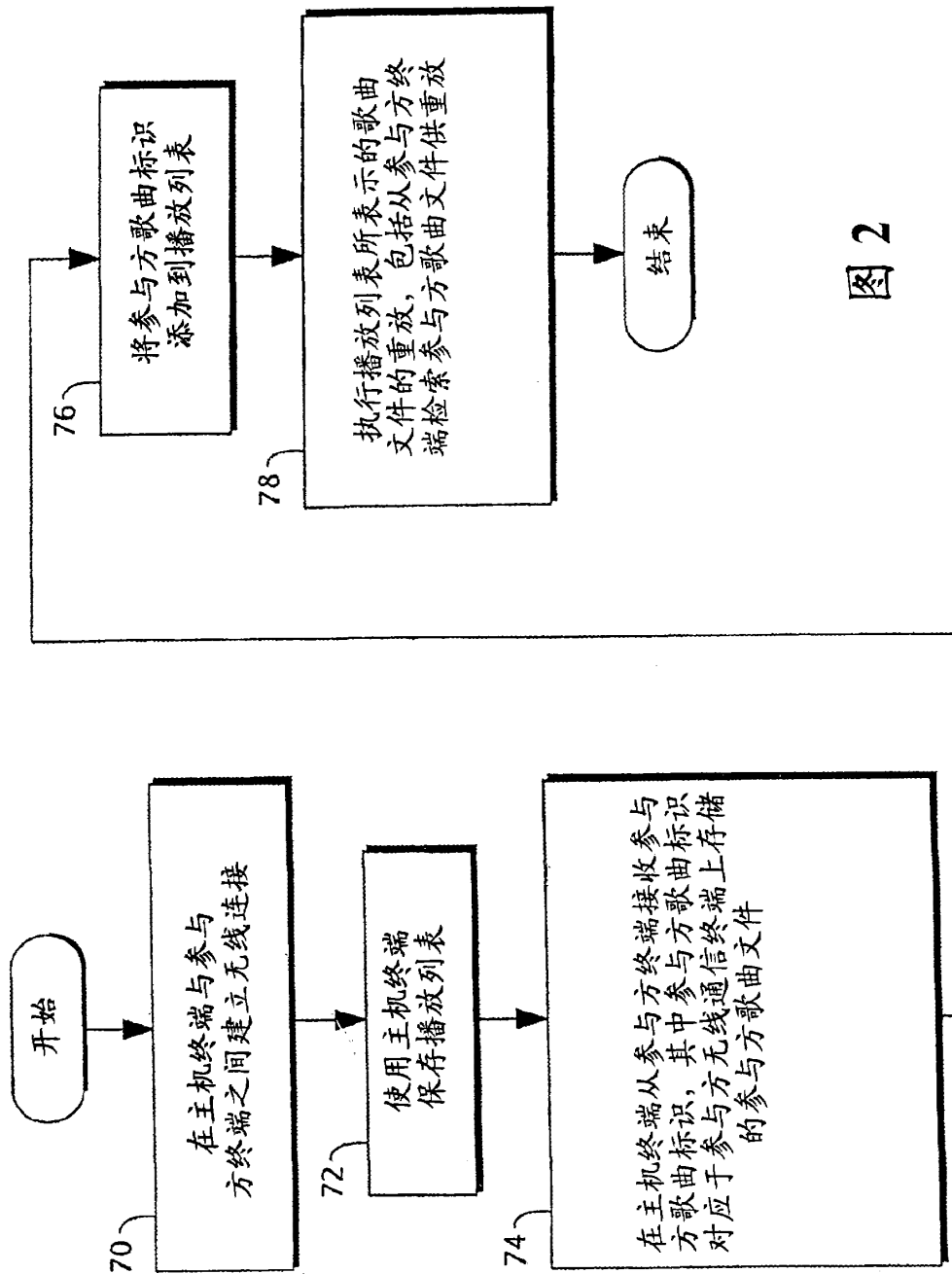


图 1



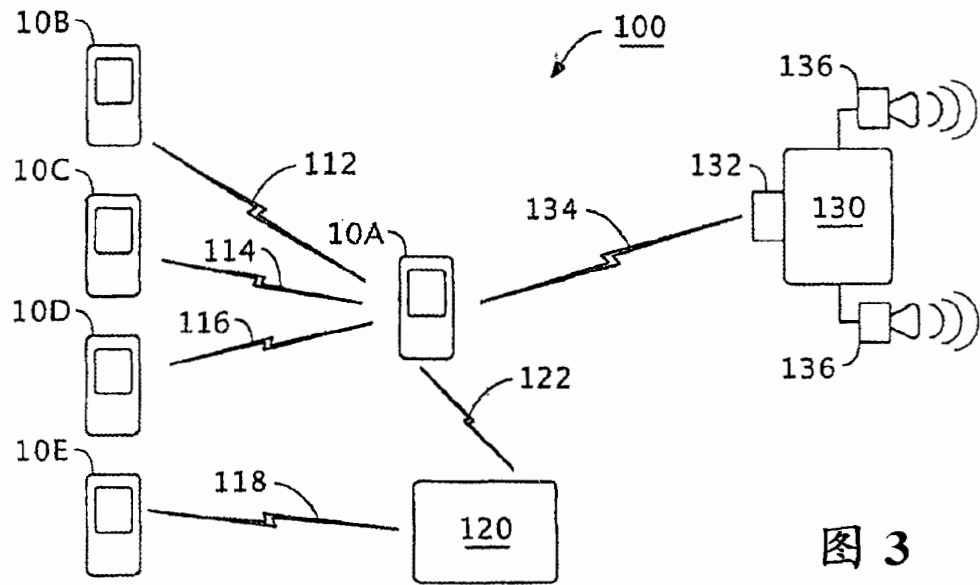


图 3

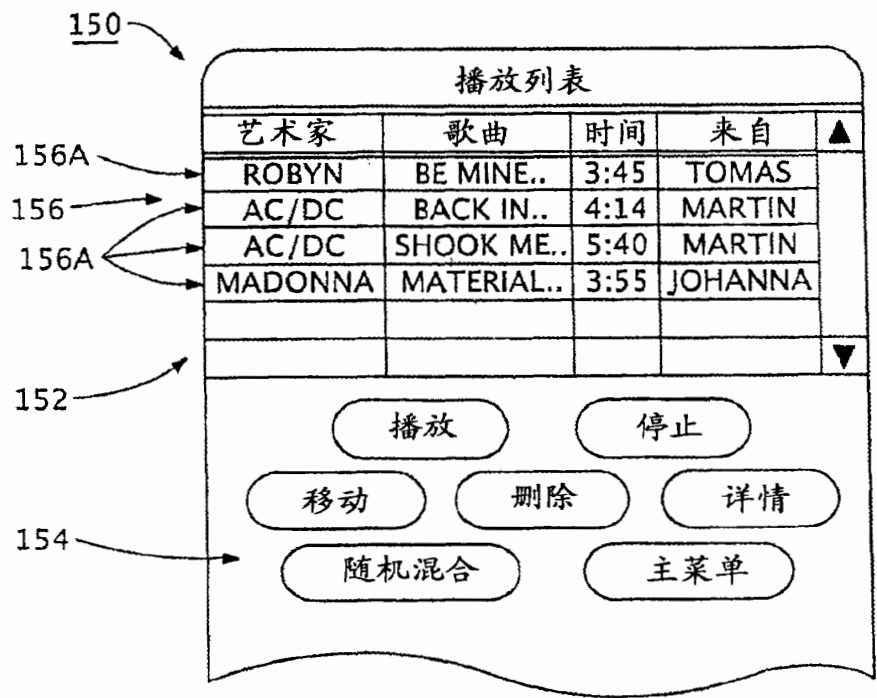
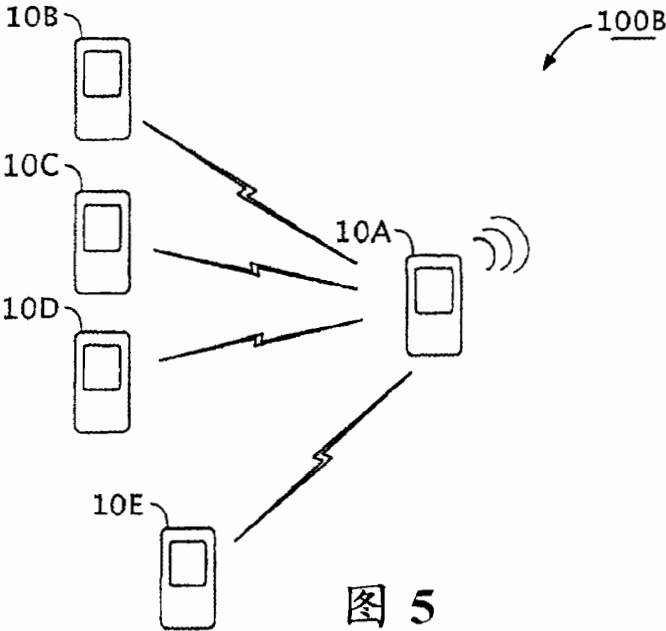
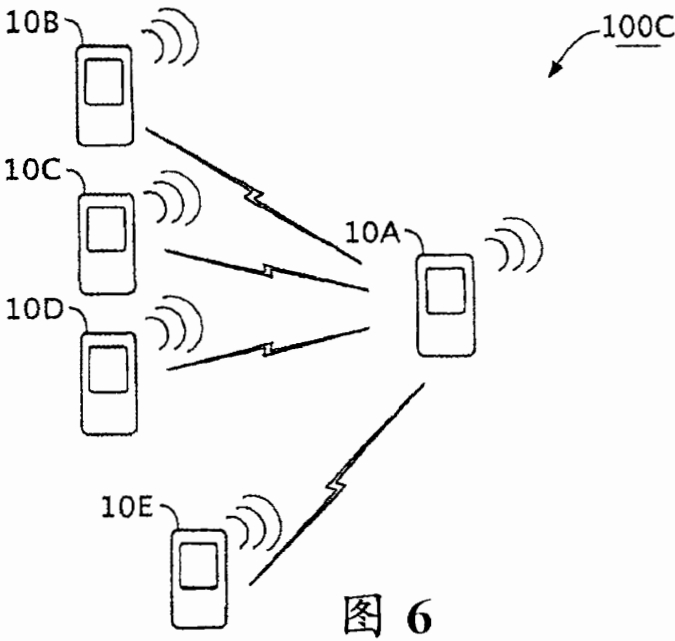


图 4



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**IN THE UNITED STATES PATENT AND TRADEMARK OFFICE
(Docket No. 13-0402)**

In re the Application of:	)	
	)	
MILLINGTON, Nicholas A.J.	)	
	)	
Application No.: 13/864,249	)	Group Art Unit: 2442
	)	
Filed: April 17, 2013	)	
	)	Examiner: Survillo, Oleg
Confirmation No.: 9806	)	
	)	
For: Method and Apparatus for Playing and	)	
Providing Audio Content in a Synchrony	)	
Group	)	

Commissioner for Patents
P.O. Box 1450
Alexandria, Virginia 22313

INFORMATION DISCLOSURE STATEMENT TRANSMITTAL LETTER

In view of the provisions set forth in 37 C.F.R. §§ 1.56, 1.97, and 1.98, Applicant submits herewith on Forms PTO/SB/08 a listing of documents that Applicant wishes to make of record in the above-identified application, along with copies of non-patent literature documents and foreign patent documents.

Documents listed in the attached Appendix and submitted herewith on Forms PTO/SB/08 arise from a patent infringement action involving Applicant that is captioned as *Sonos Inc. v. D&M Holding Inc. d/b/a the D+M Group, D&M Holdings U.S. Inc., and Denon Electronics (USA), LLC*, Civil Action No. 14-1330-RGA (D. Del. 2014). Applicant's submission of these documents in the present application is not intended as an admission that any such document constitutes prior art to the claims of the present application¹, that any such document is considered material to patentability as defined in 37 C.F.R. § 1.56(b), or that the aforementioned patent infringement action is related to the present application.

¹ For certain non-patent literature documents, the dates specified in Form PTO/SB/08 are based on allegations made by an opposing party in litigation. For the purpose of examination, Applicant respectfully requests the Office consider these documents to be available as of the specified dates. However, Applicant does not admit that the documents were available as of these dates and expressly reserves the right to challenge the availability of these documents at a later time.

Applicant respectfully requests that the listed documents be considered by the Examiner and be made of record in the present application, and that a copy of Form PTO/SB/08 be returned in accordance with M.P.E.P. §609.

Respectfully submitted,

Dated: October 13, 2016

By: /Benjamin M. Urban/
Benjamin M. Urban

McDonnell Boehnen Hulbert & Berghoff LLP
300 South Wacker Drive
Chicago, IL 60606
312 913 0001

Appendix

Patent/Pub. No.	Title	Filing Date
WO2007135581A2	A DEVICE FOR AND A METHOD OF PROCESSING AUDIO DATA	5/2/2007
WO0019693A1	METHOD AND SYSTEM OF TELECONFERENCING	9/24/1999
USD431552S	Remote control	3/3/2000
USD432525S	Handheld remote control	12/23/1999
USD412337S	Controller for a monitor camera	8/27/1998
USD416021S	Remote controller	7/27/1998
USD415496S	Remote control	3/12/1998
USD407071S	4-in-1 remote control unit	3/9/1998
USD430143S	Remote control	3/5/1998
USD410927S	Remote controller	1/27/1998
USD404741S	Control panel for an automotive audio system and climate control	1/20/1998
USD405071S	Cursor control--data entry device	10/17/1997
USD429246S	Arm support with control panel	10/8/1997
USD420006S	Remote control unit	4/16/1997
USD395889S	Remote control	3/25/1997
USD406847S	Two sided remote control	3/25/1997
USD392641S	Remote control	11/15/1996
USD397996S	Remote control with charger	9/30/1996
USD393628S	Hand-held remote control unit	8/19/1996
USD401587S	Satellite receiver remote control	4/1/1996
USD388792S	Television remote control	6/24/1996
USD389143S	Interactive display controller and telephone handset unit	4/25/1996
USD384940S	Remote controller	6/3/1996
USD380752S	Combined telephone and remote control	5/28/1996
USD382271S	Luminous remote control	5/15/1996
USD387352S	Remote control	3/19/1996
USD394659S	Remote controller	3/11/1996
USD372716S	Remote control	4/24/1995
USD379816S	Multimedia remote control	2/8/1995
USD377651S	Remote controller	10/26/1994
USD366044S	Remote controller for television receiver	9/16/1994
USD364878S	Control panel for a combined vehicular radio receiver, cassette player and digital clock	8/30/1994
USD363933S	Remote control	8/2/1994
USD362446S	Front panel for a combined radio receiver, cassette tape player and CD changer control	7/29/1994
USD364877S	Control head for a mobile radio	3/21/1994
USD356312S	Remote control unit	12/27/1993
USD350531S	Remote control for a ceiling fan	10/29/1993
USD356093S	Control Panel for a combined vehicular radio receiver, cassette player, compact disc player and digital clock	7/26/1993
USD365102S	Universal remote control unit	6/1/1993
USD357024S	Mobile radio control head	3/15/1993

USD354751S	TV remote control for the manually handicapped	12/31/1992
USD350962S	Hand held television controller	12/4/1992
USD354059S	Remote control unit	12/3/1992
US20150365987	Wireless Speaker For Receiving From A Mobile Phone Directions To Receive And Render A Playlist From A Content Server On The Internet	8/26/2015
US20110316768	SYSTEM, METHOD AND APPARATUS FOR SPEAKER CONFIGURATION	6/28/2010
US20100087089	Wall Mounted Docking Station	10/2/2009
US20110001632	Method and System for Changing an Input Source for an Electronic Device	7/1/2009
US8411883	LOUDSPEAKER APPARATUS AND SPEAKER SYSTEM	6/19/2009
US8374595	HANDHELD ELECTRONIC DEVICE AND OPERATING METHOD THEREOF	4/30/2009
US7945708	Apparatus and method of receiving data from audio/video equipment	1/9/2009
US8565455	Multiple Display Systems with Enhanced Acoustics Experience	12/31/2008
US20090124289	Audio apparatus and external adapter used for the audio apparatus	11/10/2008
US8861739	Apparatus and method for generating a multichannel signal	11/10/2008
US20090070434	SYSTEM AND METHOD FOR EFFICIENTLY ACCESSING INTERNET RESOURCES	6/6/2008
US8170222	AUGMENTED REALITY ENHANCED AUDIO	4/18/2008
US8311226	METHOD FOR ASSIGNING A PLURALITY OF AUDIO CHANNELS TO A PLURALITY OF SPEAKERS, CORRESPONDING COMPUTER PROGRAM PRODUCT, STORAGE MEANS AND MANAGER NODE	3/25/2008
US20080205070	CRADLE	1/28/2008
US7987294	UNIFICATION OF MULTIMEDIA DEVICES	10/16/2007
US7995732	Managing Audio in a Multi-Source Audio Environment	10/4/2007
US7721032	METHOD AND APPARATUS FOR MEDIATING AMONG MEDIA APPLICATIONS	8/20/2007
US8654995	Audio control system for a vehicle	4/19/2007
US8942395	Pointing element enhanced speaker system	2/12/2007
US20090011798	MOBILE PHONE	3/8/2007
US8233029	Reproduction Apparatus and System	7/14/2006
US8041062	Personal Sound System Including Multi-Mode Ear Level Module with Priority Logic	3/28/2006
US20080075295	Media playing from a docked handheld media device	8/31/2006
US7746906	DATA COMMUNICATIONS SYSTEM	7/10/2006
US20060227985	Information processing apparatus, information processing method, recording medium, and program	3/27/2006
US20060259649	INTERFACE SWITCHING APPARATUS AND METHOD	4/27/2006
US20080303947	Audio Processing Apparatus and Display Apparatus with Same	8/25/2006
US7933418	Sound reproducing apparatus and method of identifying positions of speakers	2/16/2005
US7882234	Wireless communication system, wireless communication device, and control method for establishing a one-to-one relationship	9/29/2005
US8050203	Multi-channel digital wireless audio system	12/20/2005
US8150079	Personal digital assistant and stereo reproduction method	12/14/2005
US20070087686	Audio playback device and method of its operation	10/18/2005
US7831054	Volume control	6/28/2005
US20060270395	Personal shared playback	5/25/2005
US20060222186	Multi-channel audio switch	4/5/2005
US20050197725	Music management system	2/18/2005
US7653344	Wireless digital audio/video playback system	1/5/2005

US7761176	Promotional portable music players	11/23/2004
US7688306	Methods and apparatuses for operating a portable device based on an accelerometer	11/12/2004
US8600084	Methods and systems for altering the speaker orientation of a portable system	11/9/2004
US7305694	Method and apparatus for remotely controlling a receiver according to content and user selection	9/7/2004
US7472058	AUTOMATIC AUDIO SOURCE DETECTION FOR CAPTURE DEVICES	12/28/2004
US7630501	System and method for calibration of an acoustic system	5/14/2004
US20040249490	Multichannel audio system, and head unit and slave unit used for the same	4/28/2004
US7853341	Wired, wireless, infrared, and powerline audio entertainment systems	2/20/2004
US20040171346	Method of Adding Wireless Remote Control to Audio Playback Devices Equipped for Wired Remote Control	1/5/2004
US7492912	Multi-channel speaker system and a connection system thereof	3/18/2004
US20060173844	Automatic configuration of client terminal in public hot spot	3/12/2004
US20050131558	Audio network distribution system	11/4/2004
US20050160270	Localized audio networks and associated digital accessories	11/8/2004
US7627825	Control device and method, information processing device and method, recording medium, and program	9/29/2004
USD524296	Remote controller for television set	6/14/2004
USD515072	Remote control unit	5/27/2004
USD506463	Remote control unit	4/8/2004
USD499718	Remote control	2/2/2004
USD496005	Remote control	1/21/2004
USD495333	Liquid crystal display	1/13/2004
USD499395	Remote control	1/5/2004
US20050154766	Multimedia processing system capable of auto-detecting multimedia signals to facilitate selecting and method of the same	12/23/2004
US7676044	Multi-speaker audio system and automatic control method	12/10/2004
US7548744	WIFI authentication method	12/19/2003
US7428310	Audio output adjusting device of home theater system and method thereof	12/22/2003
US7046677	Method and apparatus for time-multiplexed processing of multiple digital video programs	11/26/2003
US7430181	Method and apparatus for automatically configuring devices on a wireless network	11/26/2003
US6919771	Digital amplifier	9/25/2003
US20050060435	Middleware filter agent between server and PDA	9/17/2003
US7263110	Synchronizing clock signals of server and client devices in a network system based on power source synchronous pulse signal	9/10/2003
US7675943	Synchronous play-out of media data packets	9/4/2003
US7558224	Management of packet-based audio devices within acoustic spaces	7/29/2003
US7289631	Encryption key setting system, access point, encryption key setting method, and authentication code setting system	7/7/2003
US8442239	Audio system with balance setting based on information addresses	1/23/2003
US20030231208	Multimode interface	6/13/2003
US7676142	Systems and methods for multimedia time stretching	6/9/2003
US20040253969	Technique for discovery using a wireless network	6/6/2003
US20040185773	Mobile transceiver and electronic module for controlling the transceiver	3/18/2003
US20040183827	Methods and apparatus for implementing a remote application over a network	3/17/2003
US20040177167	Network audio systems	3/4/2003

US20030167335	System and method for network-based communication	3/3/2003
US20040168081	Apparatus and method simplifying an encrypted network	2/20/2003
US20030185400	Sound processing unit, sound processing system, audio output unit and display device	1/15/2003
US7366206	Multi-media jitter removal in an asynchronous digital home network	1/16/2003
US20040019807	System And Methodology For Providing Community-Based Security Policies	5/14/2003
USD496003	Hand-held remote control	12/16/2003
USD499086	Inline control module for communications headset adapter	12/10/2003
USD518475	Remote controller	12/2/2003
USD504872	Remote controller	11/24/2003
USD496335	Remote control	11/10/2003
USD527375	Remote control device	9/30/2003
USD497363	Liquid crystal display	9/22/2003
USD500015	Remote control device	9/19/2003
USD515557	Remote control	9/16/2003
USD501477	Keyboard control device	8/11/2003
USD491925	Remote control	7/2/2003
USD504885	Remote control	6/11/2003
USD493148	Remote controller for air conditioner	3/28/2003
USD486145	Remote control keypad	2/28/2003
US7356011	Simplified configuration and security for networked wireless devices	12/18/2002
US20040117462	Client device configuration with user entry of configuration parameters	12/12/2002
US7505889	Transcoding media system	12/5/2002
US7295548	Method and system for disaggregating audio/visual components	11/27/2002
US20030103088	User interface for a remote control application	11/6/2002
US7277547	Method for automated security configuration in a wireless network	10/23/2002
US7308188	Audio reproducing apparatus	9/30/2002
US20040059842	Network device installation	9/19/2002
US20040203590	Set-up of wireless consumer electronics device using a learning remote control	9/11/2002
US7197148	System for controlling remote speakers using centralized amplifiers, centralized monitoring and master/slave communication protocol	9/9/2002
US20030063755	System for controlling remote speakers using centralized amplifiers, centralized monitoring and master/slave communication protocol	9/9/2002
US20040042629	Automatic earpiece sensing	8/30/2002
US20040014426	Portable audio playback device with bass enhancement	7/19/2002
US20030043856	Method and apparatus for reducing synchronization delay in packet-based voice terminals by resynchronizing during talk spurts	7/2/2002
US20040001591	Robust method for achieving audio/video synchronization in MPEG decoders in personal video recording applications	6/27/2002
US7092694	Wireless communication system having a guest transmitter and a host receiver	6/19/2002
US7139981	Media management system	6/13/2002
USD477310	Remote control	10/7/2002
USD481056	Control panel for projector	9/12/2002
USD478051	Remote controller for household electric/electronic appliances	7/11/2002
USD478896	Remote control	6/28/2002
USD476643	Remote controller	6/11/2002

USD474763	Handheld remote control	5/30/2002
USD475993	TV remote control	3/21/2002
US20030219007	Reusable multi-protocol meta-architecture for Voice-over-IP playback	5/23/2002
US7096169	Virtual speaker demonstration system and virtual noise simulation	5/16/2002
US7945636	Providing a multi-tier enterprise level application	5/15/2002
US20030212802	Proximity network encryption and setup	5/9/2002
US20030198257	Method of manually fine tuning audio synchronization of a home network	4/23/2002
US20020165921	Method of multiple computers synchronization and control for guiding spatially dispersed live music/multimedia performances and guiding simultaneous multi-content presentations and system therefor	4/24/2002
US7324857	Method to synchronize playback of multicast audio streams on a local network	4/19/2002
US20030179780	Method of detecting drift between two clocks	3/20/2002
US20040037433	Multi-channel wireless professional audio system	8/21/2002
US20020131761	Information recording medium, apparatus and method for recording/reproducing information to/from the medium	1/16/2002
US7206618	Removable customizable inserts and faceplate for electronic devices	1/11/2002
US20020101357	System for transmitting control commands to electronic devices	1/4/2002
US7092528	Apparatus and methods for reducing noise audible from a speaker	11/6/2001
US7120168	System and method for effectively performing an audio/video synchronization procedure	11/20/2001
US20020078293	Method and system for selecting and controlling devices in a home network	10/12/2001
US20060072489	Home network using wireless module	10/11/2001
US7260616	Communication hub with automatic device registration	8/13/2001
US6937988	Methods and systems for prefilling a buffer in streaming data applications	8/10/2001
US6873862	Wireless headphones with selective connection to auxiliary audio devices and a cellular telephone	7/24/2001
US20030014486	Distributed audio network using networked computing devices	7/16/2001
US7161939	Method and system for switching among independent packetized audio streams	6/29/2001
US6987767	Multiplexer, multimedia communication apparatus and time stamp generation method	6/28/2001
US6778073	Method and apparatus for managing audio devices	6/26/2001
US7136934	Multimedia synchronization method and device	6/19/2001
US6653899	Audio amplifier power and temperature controller having network detecting and associated methods	6/15/2001
US20020137505	Audio detection for hands-free wireless	5/24/2001
US20010050991	Portable audio devices	5/18/2001
US7146260	Method and apparatus for dynamic configuration of multiprocessor system	4/24/2001
US7130316	System for frame based audio synchronization and method thereof	4/11/2001
US6741961	Low power audio processor that multiplexes component distribution signals	3/14/2001
US20010043456	Portable computer with low power audio CD-player	3/8/2001
US8477958	Networked sound masking system	2/26/2001
US6456783	Data structure of stream data, and recording and playback method thereof	1/8/2001
US20040223622	Digital wireless loudspeaker system	4/6/2001
USD462945	Volume control for a communications headset	11/13/2001
USD479520	Audiovisual control device	9/15/2001
USD468297	Karaoke controller with microphone	7/27/2001
USD466108	Remote control	6/15/2001

USD452520	Remote control for a vertical digital projector	3/13/2001
USD462340	Remote control	1/18/2001
US20020124097	Methods, systems and computer program products for zone based distribution of audio signals	12/29/2000
US20020072817	System and method for request, delivery and use of multimedia files for audiovisual entertainment in the home environment	12/11/2000
US6915347	Associating multiple display units in a grouped server environment	12/6/2000
US20020072816	Audio system	12/7/2000
US7548851	Digital multimedia jukebox	10/11/2000
US7457948	Automated authentication handling system	9/29/2000
US6826283	Method and system for allowing multiple nodes in a small environment to play audio signals independent of other nodes	7/27/2000
US6859460	System and method for providing multimedia jitter buffer adjustment for packet-switched networks	5/31/2000
US6654720	Method and system for voice control enabling device in a service discovery network	5/9/2000
US6684060	Digital wireless premises audio system and method of operation thereof	4/11/2000
US6654956	Method, apparatus and computer program product for synchronizing presentation of digital video data with serving of digital video data	4/10/2000
US6535121	Fire department station zoned alerting control system	4/6/2000
US6816510	Method for clock synchronization between nodes in a packet network	2/9/2000
US6901439	Method of adding a device to a network	1/20/2000
USD462339	Remote control	12/22/2000
USD478069	Communication interface with external controls	11/1/2000
US7558635	Control device and control method for fixing an information output order for several information sources, especially audio sources	9/5/2000
US7966388	Network management system and graphical user interface	12/21/1999
US6636269	Video timing system and method	8/18/1999
US6463474	Local authentication of a client at a network device	7/2/1999
US6046550	Multi-zone lighting control system	6/16/1999
US6356871	Methods and circuits for synchronizing streaming data and systems using the same	6/14/1999
US6931557	INFORMATION PROCESSING APPARATUS, POWER CONTROL METHOD AND RECORDING MEDIUM TO CONTROL A PLURALITY OF DRIVING UNITS ACCORDING TO THE TYPE OF DATA TO BE PROCESSED	4/7/1999
US6493832	Communication apparatus which handles a time stamp	3/29/1999
US6859538	Plug and play compatible speakers	3/17/1999
US6061457	Waterproof microphone and speaker	2/22/1999
US6353172	Music event timing and delivery in a non-realtime environment	2/2/1999
USD444475	Set of keys for a computer keyboard	8/23/1999
USD444475	Set of keys for a computer keyboard	8/23/1999
US6169725	Apparatus and method for restoration of internal connections in a home audio/video system	10/30/1998
US6269406	User group synchronization to manage capabilities in heterogeneous networks	10/19/1998
US6487296	Wireless surround sound speaker system	9/30/1998
US6449642	METHOD AND SYSTEM FOR INTEGRATING A CLIENT COMPUTER INTO A COMPUTER NETWORK	9/15/1998
US6313879	Synchronization method and decoder	9/11/1998
US6466832	High quality wireless audio speakers	8/24/1998

US6321252	System and method for data streaming and synchronization in multimedia groupware applications	7/17/1998
US6442443	Information processing apparatus and power saving apparatus	7/17/1998
US6148205	Method and apparatus for secure registration within an in-home wireless network	6/30/1998
US6195435	Method and system for channel balancing and room tuning for a multichannel audio surround sound speaker system	5/1/1998
US5946343	Digital wireless speaker system	4/24/1998
US6301012	Automatic configuration of a network printer	4/24/1998
US5999906	Sample accurate audio state update	3/4/1998
US6418150	Method and apparatus for calibrating an IEEE-1394 cycle master	2/20/1998
US6127941	Remote control device with a graphical user interface	2/3/1998
US6246701	Reference time clock locking in a remultiplexer for video program bearing transport streams	1/14/1998
US6469633	Remote control of electronic devices	1/5/1998
US6038614	Active volume control with hot key	1/5/1998
US5852722	System and method for automatic configuration of home network computers	12/8/1997
US6212282	Wireless speaker system	10/31/1997
US6018376	Synchronous reproduction apparatus	8/18/1997
US6704421	Automatic multichannel equalization control system for a multimedia computer	7/24/1997
US6029196	Automatic client configuration system	6/18/1997
US5956025	Remote with 3D organized GUI for a home entertainment system	6/9/1997
US6101195	Timing correction method and apparatus	5/28/1997
US6181383	Method and apparatus for preserving synchronization of audio and video presentation when splicing transport streams	5/28/1997
US5987611	System and methodology for managing internet access on a per application basis for client computers connected to the internet	5/6/1997
US6310652	Fine-grained synchronization of a decompressed audio stream by skipping or repeating a variable number of samples from a frame	5/2/1997
US5990884	Control of multimedia information with interface specification stored on multimedia component	5/2/1997
US6081266	Interactive control of audio outputs on a display screen	4/21/1997
US5991307	Method and apparatus for packet transmission	3/28/1997
US5832024	Digital wireless speaker system	3/26/1997
US5875233	Audio record and playback through a standard telephone in a computer system	2/18/1997
US6088063	Data encoding method and apparatus for outputting encoded data at a designated time	1/29/1997
US5905768	MPEG audio synchronization system using subframe skip and repeat	12/20/1996
US5848152	Communication device having interchangeable faceplates and active keypad cover	12/4/1996
US5818948	Architecture for a universal serial bus-based PC speaker controller	10/23/1996
US5960006	MPEG decoding system adjusting the presentation in a predetermined manner based on the actual and requested decoding time	9/23/1996
US5742623	Error detection and recovery for high rate isochronous data in MPEG-2 data streams	7/11/1996
US5661665	Multi-media synchronization	6/26/1996
US5774016	Amplifier system having prioritized connections between inputs and outputs	4/9/1996
US5812201	Data synchronizing apparatus and method thereof	12/27/1995
US5640388	Method and apparatus for removing jitter and correcting timestamps in a packet stream	12/21/1995
US5790543	Apparatus and method for correcting jitter in data packets	9/25/1995
US5761320	Audio distribution system having programmable zoning features	6/26/1995

US5984512	Method for storing video information	6/7/1995
US5625350	Audio/video communication system and method	12/27/1994
US5515345	Control unit for automobile audio system	8/23/1994
US5553314	Method of configuring a communication unit using a wireless portable configuration device	4/12/1994
US5740235	User-programmable paging system controller having priority-based, multiple input paging access capability for selectively activating one or more audio/visual paging output devices	2/9/1994
US5361381	Dynamic equalizing of powered loudspeaker systems	12/17/1993
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Electronic Patent Application Fee Transmittal				
Application Number:		13864249		
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Filer:		Benjamin Michael Urban/Liz Furiasse		
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Patent-Appeals-and-Interference:				
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File Listing:					
Document Number	Document Description	File Name	File Size(Bytes)/ Message Digest	Multi Part /.zip	Pages (if appl.)
1	Other Reference-Patent/App/Search documents	Sonos_v_DM_Opening_Brief_Support_Def_Part_Mot_Judg_5-6-2016.pdf	4496109	no	27
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2	Other Reference-Patent/App/Search documents	SONOS_v_DM_OPENING_Claim_Construction_Brief_9-9-2016.pdf	24360442	no	26
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3	Other Reference-Patent/App/Search documents	Sonos_v_DM_Plaintiffs_Response_in_Opposition_5-27-2106.pdf	25234778	no	24
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4	Other Reference-Patent/App/Search documents	Sonos_v_DM_SECOND_AMENDED_COMPLAINT_2-27-2015.pdf	7809394	no	49
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			89f5e4d1f1c4ebe0ef65f1ab91b070d0c6abdd9a4		
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30	Information Disclosure Statement (IDS) Form (SB08)	13-0402_MBHB15-518-CON_IDS_third_13OCT2016.pdf	1055286	no	6
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31	Foreign Reference	CN101292500A.pdf	1591837	no	27
			58dd12ffcaaa5f82a265c69b19abeafbad7cd915		
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			10bccc87e174a8955b2e144c80ea7f0f3bebd5f4		
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			a8876bc0bcf06a52ff39254b2a9a74db3c4facad		
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New Applications Under 35 U.S.C. 111

If a new application is being filed and the application includes the necessary components for a filing date (see 37 CFR 1.53(b)-(d) and MPEP 506), a Filing Receipt (37 CFR 1.54) will be issued in due course and the date shown on this Acknowledgement Receipt will establish the filing date of the application.

National Stage of an International Application under 35 U.S.C. 371

If a timely submission to enter the national stage of an international application is compliant with the conditions of 35 U.S.C. 371 and other applicable requirements a Form PCT/DO/EO/903 indicating acceptance of the application as a national stage submission under 35 U.S.C. 371 will be issued in addition to the Filing Receipt, in due course.

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Examiner Initial*	Cite No	Patent Number	Kind Code ¹	Issue Date	Name of Patentee or Applicant of cited Document	Pages, Columns, Lines where Relevant Passages or Relevant Figures Appear
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Electronic Acknowledgement Receipt	
EFS ID:	27175765
Application Number:	13864249
International Application Number:	
Confirmation Number:	9806
Title of Invention:	METHOD AND APPARATUS FOR PLAYING AND PROVIDING AUDIO CONTENT IN A SYNCHRONY GROUP
First Named Inventor/Applicant Name:	Nicholas A. J. Millington
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File Listing:

Document Number	Document Description	File Name	File Size(Bytes)/ Message Digest	Multi Part /.zip	Pages (if appl.)
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<p>This Acknowledgement Receipt evidences receipt on the noted date by the USPTO of the indicated documents, characterized by the applicant, and including page counts, where applicable. It serves as evidence of receipt similar to a Post Card, as described in MPEP 503.</p> <p><u>New Applications Under 35 U.S.C. 111</u> If a new application is being filed and the application includes the necessary components for a filing date (see 37 CFR 1.53(b)-(d) and MPEP 506), a Filing Receipt (37 CFR 1.54) will be issued in due course and the date shown on this Acknowledgement Receipt will establish the filing date of the application.</p> <p><u>National Stage of an International Application under 35 U.S.C. 371</u> If a timely submission to enter the national stage of an international application is compliant with the conditions of 35 U.S.C. 371 and other applicable requirements a Form PCT/DO/EO/903 indicating acceptance of the application as a national stage submission under 35 U.S.C. 371 will be issued in addition to the Filing Receipt, in due course.</p> <p><u>New International Application Filed with the USPTO as a Receiving Office</u> If a new international application is being filed and the international application includes the necessary components for an international filing date (see PCT Article 11 and MPEP 1810), a Notification of the International Application Number and of the International Filing Date (Form PCT/RO/105) will be issued in due course, subject to prescriptions concerning national security, and the date shown on this Acknowledgement Receipt will establish the international filing date of the application.</p>					

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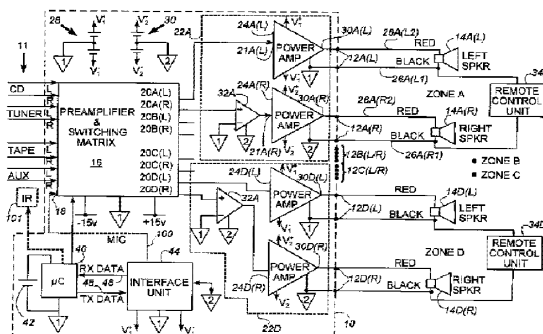
(71) BONNEVILLE, MARC ETIENNE,
3020 Quillivan Drive, CUMBERLAND, O1 (CA).

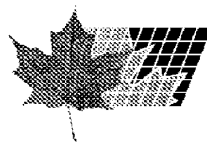
(72) BONNEVILLE, MARC ETIENNE (CA).

(74) THOMAS ADAMS & ASSOC.

(54) TRANSMISSION DE LA PUISSANCE ET/OU DU SIGNAL ENTRE UN CENTRE DE DISTRIBUTION AUDIO ET PLUSIEURS TRANSDUCTEURS AUDIO ELOIGNES
 (54) TRANSMISSION OF POWER AND/OR SIGNALLING BETWEEN AN AUDIO DISTRIBUTION UNIT AND A PLURALITY OF REMOTE AUDIO TRANSDUCERS

(57) A remote-controlled audio system comprises sets of loudspeakers (14A-14D) connected to power amplifier units (22A - 22D) in an audio distribution unit (10) to which several audio sources, such as a compact disc unit, a tape deck, etc., are connected. Remote units (34) are provided, one in each zone with a set of loudspeakers. The loudspeakers and the remote unit are connected to the audio distribution unit by four wires whereby audio and data signals and power are transmitted to the loudspeakers and the remote unit, respectively. Each power amplifier unit may have a pair of power amplifiers, each connected to one of the loudspeakers in the set, referenced to a first ground (1) and a second ground (2), respectively. The second ground (2) is maintained at a prescribed voltage relative to the first ground. The remote unit (34A) is connected between the "ground" terminals of the two loudspeakers of the set, i.e. those connected via the speaker cables to the first and second grounds in the audio distribution unit. The remote unit has a transmitter unit which transmits control signals to the audio distribution unit via the "ground" cables. In the audio distribution unit, an interface circuit connected between the grounds conveys the control signals to a control device which adjusts the settings of the audio distribution unit in dependence thereupon. The remote units may receive power from the audio distribution unit via the transmission channel to supply their internal components.





(72) BONNEVILLE, MARC ETIENNE, CA

(71) BONNEVILLE, MARC ETIENNE, CA

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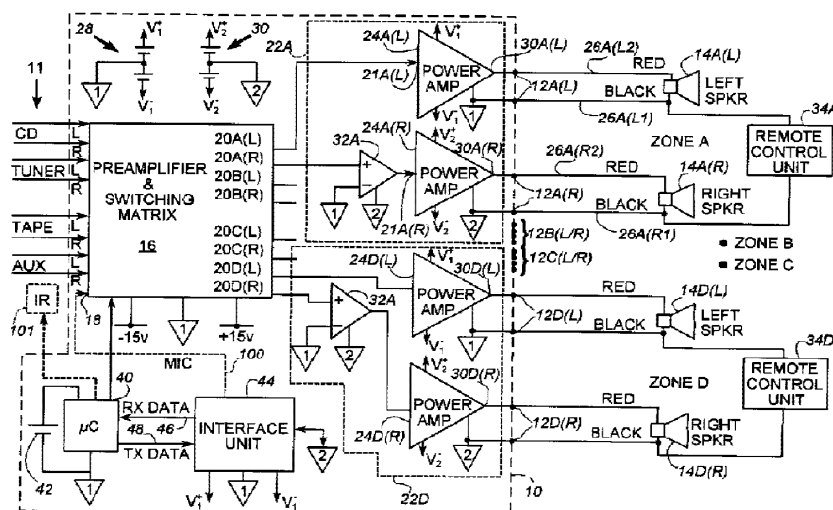
(30) 1999/09/23 (2,283,577) CA

(54) TRANSMISSION DE LA PUISSANCE ET/OU DU SIGNAL

ENTRE UN CENTRE DE DISTRIBUTION AUDIO ET
PLUSIEURS TRANSDUCTEURS AUDIO ELOIGNES

(54) TRANSMISSION OF POWER AND/OR SIGNALLING BETWEEN

AN AUDIO DISTRIBUTION UNIT AND A PLURALITY OF
REMOTE AUDIO TRANSDUCERS



(57) A remote-controlled audio system comprises sets of loudspeakers (14A-14D) connected to power amplifier units (22A - 22D) in an audio distribution unit (10) to which several audio sources, such as a compact disc unit, a tape deck, etc., are connected. Remote units (34) are provided, one in each zone with a set of loudspeakers. The loudspeakers and the remote unit are connected to the audio distribution unit by four wires whereby audio and data signals and power are transmitted to the loudspeakers and the remote unit, respectively. Each power amplifier unit may have a pair of power amplifiers, each connected to one of the loudspeakers in the set, referenced to a first ground (1) and a second ground (2), respectively. The second ground (2) is maintained at a prescribed voltage relative to the first ground. The remote unit (34A) is connected between the "ground" terminals of the two loudspeakers of the set, i.e. those connected via the speaker cables to the first and second grounds in the audio distribution unit. The remote unit has a transmitter unit which transmits control signals to the audio distribution unit via the "ground" cables. In the audio distribution unit, an interface circuit connected between the grounds conveys the control signals to a control device which adjusts the settings of the audio distribution unit in dependence thereupon. The remote units may receive power from the audio distribution unit via the transmission channel to supply their internal components.



ABSTRACT OF THE DISCLOSURE

A remote-controlled audio system comprises sets of loudspeakers (14A -
5 14D) connected to power amplifier units (22A - 22D) in an audio distribution
unit (10) to which several audio sources, such as a compact disc unit, a tape
deck, etc., are connected. Remote units (34) are provided, one in each zone
with a set of loudspeakers. The loudspeakers and the remote unit are
connected to the audio distribution unit by four wires whereby audio and data
10 signals and power are transmitted to the loudspeakers and the remote unit,
respectively. Each power amplifier unit may have a pair of power amplifiers,
each connected to one of the loudspeakers in the set, referenced to a first
ground (1) and a second ground (2), respectively. The second ground (2) is
maintained at a prescribed voltage relative to the first ground. The remote unit
15 (34A) is connected between the "ground" terminals of the two loudspeakers
of the set, i.e. those connected via the speaker cables to the first and second
grounds in the audio distribution unit. The remote unit has a transmitter unit
which transmits control signals to the audio distribution unit via the "ground"
cables. In the audio distribution unit, an interface circuit connected between
20 the grounds conveys the control signals to a control device which adjusts the
settings of the audio distribution unit in dependence thereupon. The remote
units may receive power from the audio distribution unit via the transmission
channel to supply their internal components.

TRANSMISSION OF POWER AND/OR SIGNALLING BETWEEN AN AUDIO
DISTRIBUTION UNIT AND A PLURALITY OF REMOTE AUDIO TRANSDUCERS

BACKGROUND OF THE INVENTION

5 TECHNICAL FIELD

The invention relates to the transmission of power and/or signalling between an audio distribution unit and a plurality of remote audio transducers and is especially applicable to multi-zone audio systems or entertainment centres comprising an audio distribution unit for driving several
10 sets of loudspeakers in different zones, such as different rooms of a house.

BACKGROUND ART

A multi-zone audio/entertainment system may have a plurality of loudspeaker sets, each set located in a different zone, such as a room of a
15 house or other building. Typically, such a system comprises an audio distribution unit having inputs for audio signals from several sources (tuner, CD player, tape player, etc.), output ports for connection to the sets of loudspeakers, and a switching matrix for connecting the sources to respective ones of the loudspeaker sets. Some known systems provide local control of
20 volume by means of an autotransformer, but these are susceptible to distortion, heating and poor frequency response. It is desirable for the user to be able to control settings of the audio distribution unit from the zones where the loudspeakers are located.

It is known to provide remote control of audio source selection, volume
25 and tone control, balance adjustment, and so on, by means of keypad units, one for each zone. Each keypad unit communicates with a microcontroller in the audio distribution unit to provide control of most of its functions. A disadvantage of such systems is that each keypad unit is connected to the audio distribution unit by a multi-conductor cable which comprises some
30 conductors for exchanging control signals between the remote keypad unit and the audio distribution unit, and others for supplying power from the audio distribution unit to the remote keypad unit. It is expensive to provide such multi-conductor cabling for every keypad, and such expense is especially

unacceptable when adding remote control to an existing multi-zone loudspeaker system, which usually would require installation of multi-conductor cables in addition to existing two-conductor loudspeaker cables.

It is also known to use wireless communication links between remote control units and an audio distribution unit. Disadvantageously, however, the radio frequency circuitry is relatively expensive and susceptible to interference, and the remote control unit requires a local power supply. Also, speaker drive conductors would usually still be required unless the audio drive also used wireless links. The latter systems either are expensive or deliver poor quality results.

SUMMARY OF THE INVENTION

The present invention seeks to eliminate, or at least mitigate, the disadvantages of these known remote control systems.

According to one aspect of the present invention, there is provided apparatus comprising an audio distribution unit (10) having means (11) for providing audio signals from audio sources and output ports (12A...12D) for supplying sets of audio transducers, each said set comprising at least a first audio transducer (14A(L)) and a second audio transducer (14A(R)) and having associated therewith at least one remote unit (34A), the first and second audio transducers (14A(L), 14A(R)) and the remote unit (34A) being connected to the audio distribution unit (10) by a set of four conductors, the apparatus further comprising means (22A...22D, 44, 76, 80-96) for supplying audio signals to the audio transducers (14A(L), 14A(R)) and transferring at least one of power and data signals between said audio distribution unit (10) and said remote unit (34A), all by way of the four conductors.

In one embodiment of this first aspect, the audio distribution unit (10) comprises first and second power amplifiers (24A(L), 24A(R)), a first conductor (26A(L1)) connects one terminal of each of said first and second audio transducers (14A(L), 14A(R)) and one terminal of the remote unit (34A) to a ground terminal of the audio distribution unit (10), a second (26A(L2)) of the conductors connects a drive-signal terminal of said first audio transducer (14A(L)) to a corresponding output terminal (30A(L)) of said first power

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amplifier (24A(L)), a third conductor (26A(R1)) connects a second terminal of the remote unit (34A) to an interface unit (44) of the audio distribution unit (10), and a fourth conductor (26A(R2)) connects a drive-signal terminal of said second audio transducer (14A(R)) to a corresponding output terminal (30A(R)) of said second power amplifier (24A(R)), the audio signals being supplied by way of the first, second and fourth conductors and the at least one of power and data signals being transferred by way of the first and third conductors.

In another embodiment of the first aspect, the audio distribution unit (10) comprises first and second power amplifiers (24A(L), 24A(R)) each connected to a respective one of two grounds (1,2) that are separated electrically from each other, a first conductor (26A(L1)) connects one drive-signal terminal of the first audio transducer (14A(L)) to a ground terminal of the first power amplifier (24A(L)), a second conductor (26A(L2)) connects the other drive-signal terminal of the first audio transducer (14A(L)) to an output terminal (30A(L)) of the first power amplifier (24A(L)), a third conductor (26A(R1)) connects one terminal of the second audio transducer (14A(R)) to a ground terminal of the second power amplifier (24A(R)), and a fourth conductor (26A(R2)) connects the other drive-signal terminal of the second audio transducer (14A(R)) to a corresponding output terminal (30A(R)) of the second power amplifier (24A(R)), the audio signals being supplied to the first audio transducer (14A(L)) by way of the first and second conductors, and to the second audio transducer (14A(R)) by way of the third and fourth conductors, and said at least one of power and data signals being transferred by way of the first (26A(L1)) and third (26A(R1)) conductors.

According to a second aspect of the invention, there is provided apparatus comprising an audio distribution unit having input means for audio signals from audio sources and output ports connected to at least one set of audio transducers. The set of audio transducers comprises at least a first audio transducer and a second audio transducer connected to a respective set of said output ports by a set of conductors comprising first, second, third and fourth conductors. The apparatus further comprises at least one remote unit associated with the set of audio transducers. The audio distribution unit has a first ground and a separate second ground. The first conductor and the

second conductor connect respective drive-signal terminals of the first audio transducer to the first ground and a line terminal of a first of the set of output ports, respectively, of the audio distribution unit. The third conductor and the fourth conductor connect respective drive signal terminals of the second audio
5 transducer to the second ground and a line terminal of a second of the set of output ports, respectively, of the audio distribution unit. The remote unit is connected to the first ground and the second ground. The apparatus further comprises transferring means for transferring at least one of power and data signals between the audio distribution unit and the remote unit by way of a
10 transmission path including the first ground and second ground.

Preferably, the or each remote unit is connected to the first ground and the second ground by the first conductor and third conductor, respectively.

The transferring means may comprise means for maintaining a potential difference between the first and second grounds, and the remote unit then may
15 comprise a power extraction circuit connected to the first ground and second ground for extracting power for operation of components of the remote unit.

The apparatus may comprise a plurality of such sets of audio transducers connected by a corresponding plurality of sets of conductors and having a corresponding plurality of associated remote units. Each remote unit may then
20 be connected to the first ground and second ground in the audio distribution unit by conductors of the same set of conductors that connect the associated audio transducers to the audio distribution unit. Alternatively, one or more of the remote units may be connected to the audio distribution unit by at least one of the conductors of a set connecting a different set of audio transducers
25 to the audio distribution unit, i.e. not the set of audio transducers with which that remote unit is associated.

Where the apparatus comprises a plurality of audio transducer sets, the transferring means may include means at each remote unit for including an identifier in its transmitted data signals and means at the audio distribution unit
30 for detecting the identifier and identifying therefrom the corresponding remote unit.

The or each remote unit may comprise modulator means for providing signals modulated in dependence upon data input thereto, and the audio

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distribution unit then may have a corresponding demodulator means connected between the grounded conductors for detecting and demodulating such modulated signals. The audio distribution unit then may comprise control means, such as a microcomputer, responsive to the demodulated signals for
5 controlling operation of the audio distribution unit.

The audio distribution unit also may have modulation means for transmitting onto the transmission path signals modulated according to data signals received from components of, or attached to, the audio distribution unit. The remote unit then may have a corresponding demodulator for
10 detecting and demodulating such signals received via the transmission path from the audio distribution unit.

The remote unit and audio distribution unit may employ any suitable bandpass data transmission scheme, such as Frequency Shift Keying or Amplitude Shift Keying, for communicating said signals.

15 Preferably, the audio distribution unit comprises a preamplifier unit and an audio power amplifier unit for supplying audio signals from the preamplifier unit to the set of audio transducers, the audio power amplifier unit comprising first and second power amplifiers coupled to the first and second audio transducers, respectively, and having their ground terminals coupled to the first
20 and second grounds, respectively. The first power amplifier then has its input connected directly to a corresponding first audio output port of the preamplifier unit and the second power amplifier has its input coupled to the corresponding second audio output port of the preamplifier unit by way of a coupling device arranged to permit a potential difference between the ground terminals of the
25 second power amplifier and the preamplifier unit, respectively.

In one preferred embodiment, the first power amplifier has a ground terminal and an output terminal connected by said first and second conductors, respectively, to said one of the two audio transducers, the ground terminal being connected to said first ground, and an input terminal connected to a first
30 audio output port of the preamplifier unit. The second power amplifier has a ground terminal and an output terminal connected by said third and fourth conductors, respectively, to said other of the two audio transducers, and an input terminal connected to an output of the coupling device, the coupling

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device having its input connected to the second audio output port of the preamplifier unit, and the second power amplifier ground terminal being connected to said second ground.

The audio distribution unit may have a third ground separate from both
 5 the first ground and the second ground, the set of output ports comprise a third output port, the one set of audio transducers comprise a third audio transducer and the set of conductors further comprise a fifth conductor and a sixth conductor connecting drive terminals of the third audio transducer to ground and line terminals, respectively, of the third output port. The remote unit then
 10 may be connected also to the third ground, and the transferring means be connected to the reference, second and third grounds for transference of power to the remote unit via one pair of the three grounds and transference of the data signals via a different pair of the three grounds. Alternatively, signalling might be transferred via either or both of the channels.

15 The apparatus may comprise a second set of audio transducers connected to a corresponding second set of output ports of the audio distribution unit by a second set of first, second, third and fourth conductors and the audio distribution unit have a third ground separate from both the first ground and second ground, with a first audio transducer of the second set
 20 connected to the first ground and a second audio transducer of the second set connected to the third ground, a second remote unit being associated with the second set of audio transducers and connected to the audio distribution unit by a transmission path, conveniently different from the first transmission path. The transferring means then may be coupled to the third ground for
 25 transmission of either or both of power and signalling between the audio distribution unit and the second remote unit connected to the third ground and first ground.

Each remote unit may be connected to conductors connecting to the audio distribution unit the set of audio transducers with which it is co-located;
 30 or connected to conductors connecting a different set of audio transducers. In either case, signals to and from the remote unit will include an identifier identifying the location of the remote unit.

The apparatus may be provided with an intercom feature, whether the remote unit is connected to the audio distribution unit by a multiconductor cable or via the loudspeaker conductors. Thus, the remote unit may comprise voice circuitry for producing voice signals in response to a microphone and
5 transmitting the voice signals to the audio distribution unit via the two grounds and means for adding to the signals at least one address of a remote unit, the audio distribution unit comprising means for detecting the voice signals and applying the detected voice signals to an input of the audio distribution unit, the audio distribution means comprising means for detecting the address and,
10 in dependence thereon, directing the voice signals to one or more audio transducer sets other than that from which the voice signals originated.

According to a third aspect of the invention, there is provided apparatus comprising an audio distribution unit and a plurality of sets each of at least one audio transducer connected by a signal transmission path to the audio
15 distribution unit, at least one of the sets of at least one audio transducer being remote from the audio distribution unit and having a remote unit associated therewith, the remote unit comprising voice circuitry for transmitting voice signals from a microphone via a transmission channel interconnecting the remote unit and the audio distribution unit, and means for adding to the signals
20 at least one address corresponding to one of the sets of audio transducers, and the audio distribution unit comprises means for extracting the voice signals from the transmission channel and applying the extracted voice signals to an input of the audio distribution unit, and means for detecting the address and, in dependence thereon, directing the voice signals to one or more audio
25 transducer sets other than that from which the voice signals originated.

The transmission channel may comprise conductors connected to the audio transducers, or separate conductors.

The audio distribution unit may have separate grounds, and the transmission channels be coupled between those grounds.

30 According to a fourth aspect of the invention, there is provided apparatus comprising an audio distribution unit having means for providing audio signals from at least one audio source controllable by infrared control signals, at least one audio transducer at a location remote from the audio

distribution unit and connected thereto by a signal transmission path, and a remote unit associated with the audio transducer and connected thereto by a transmission channel, the apparatus further comprising an infrared transmitter for transmitting said infrared control signals for controlling said audio source,
5 and wherein the remote unit comprises a receiver for receiving, from an infrared remote control device, infrared signals corresponding to said infrared control signals, generating in response thereto corresponding source control signals, and transmitting the source control signals to the audio distribution unit by way said transmission channel, the audio distribution unit having means for
10 detecting said source control signals and controlling the infrared transmitter in dependence thereupon to generate said infrared control signals.

In embodiments of any of these aspects of the invention, the audio distribution unit and the audio transducers may be supplied separately. Hence, according to a fifth aspect of the invention, there is provided an audio
15 distribution unit for use as part of the apparatus of the first aspect, and having means for providing audio signals from audio sources, a first ground, a separate second ground, at least one set of output ports each having a line terminal and a ground terminal, such set of output ports including a first output port and a second output port having their respective ground terminals connected to the
20 first ground and second ground, respectively, for connection to a first and a second of a corresponding set of audio transducers, and means for transferring by way of said first ground and said second ground, and said first and second output ports, at least one of power and data signals between said audio distribution unit and a remote unit associated with said set of audio
25 transducers.

According to a sixth aspect of the invention, there is provided an audio distribution unit for use as part of apparatus according to the second aspect, namely with a plurality of sets each of at least one audio transducer connected by a signal transmission path to the audio distribution unit, at least one of the
30 sets of at least one audio transducer being remote from the audio distribution unit and having a remote unit associated therewith, the remote unit comprising voice circuitry (102) for transmitting voice signals from a microphone via a transmission channel interconnecting the remote unit and the audio distribution

unit, and means for adding to the signals at least one address corresponding to one of the sets of audio transducers, said audio distribution unit comprising means for extracting the voice signals from the transmission channel, and means for detecting the address and, in dependence thereon, directing the
5 voice signals to one or more audio transducer sets other than that from which the voice signals originated.

According to a seventh aspect of the invention, there is provided an audio distribution unit having means for providing audio signals from at least one audio source controllable by infrared control signals and configured for use
10 with at least one audio transducer at a location remote from the audio distribution unit and connected thereto by a signal transmission path, and a remote unit associated with the audio transducer and connected thereto by a transmission channel, the remote unit comprising a receiver for receiving, from an infrared remote control device, infrared signals corresponding to said infrared
15 control signals, generating in response thereto corresponding source control signals, and transmitting the source control signals to the audio distribution unit by way said transmission channel, the audio distribution unit having an infrared transmitter for transmitting said infrared control signals for controlling said audio source, and means for detecting said source control signals and
20 controlling the infrared transmitter in dependence thereupon to generate said infrared control signals.

Various other objects, features, aspects and advantages of the present invention will become more apparent from the following detailed description, taken in conjunction with the accompanying drawings, of preferred
25 embodiments of the invention, which are described by way of example only.

BRIEF DESCRIPTION OF DRAWINGS:

Figure 1 is a schematic diagram of an audio system comprising an audio distribution unit and several sets of loudspeakers, each set located in a
30 different zone and having an associated remote unit;

Figure 2 is a more detailed block schematic diagram of a line interface unit of the audio distribution unit; and

Figure 3 is a more detailed block schematic diagram of the remote unit;

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Figure 4 is a schematic circuit diagram of a power extraction circuit of the remote unit;

Figure 5 is a schematic circuit diagram of a microphone and a microphone interface of a first modification;

5 Figure 6 illustrates an alternative coupling device for coupling audio signals in the audio distribution unit;

Figure 7 illustrates modifications to parts of the audio system of Figure 1 to provide different grounds for different sets of loudspeakers;

10 Figure 8 illustrates a modification which uses three grounds to provide two transmission channels to a set of three loudspeakers, one channel for transmitting power and the other for signalling;

Figure 9 illustrates a modified interface unit for the modified apparatus of Figure 8;

15 Figure 10 illustrates modifications to a remote unit for use with the modified apparatus of Figure 8; and

Figure 11 illustrates a further embodiment of the invention in which two speakers and a remote control in a zone are connected to the audio unit by four conductors, one being a common ground conductor.

20 DESCRIPTION OF THE PREFERRED EMBODIMENT

Referring to Figure 1, a multi-zone audio system comprises an audio distribution unit 10 which has several stereo inputs. Figure 1 shows inputs 11 for a compact disc player (CD), a radio tuner (TUNER) and a tape unit (TAPE), and an auxiliary input (AUX). Usually, there would be additional inputs for
 25 other sources, such as a video cassette recorder, digital video disc player (DVD) and so on. The audio distribution unit 10 has four sets of output ports 12A, 12B, 12C and 12D connected to four sets of audio transducers, specifically loudspeakers, located in four zones A, B, C and D, respectively. Each set of the output ports comprises a stereo pair of output ports 12A(L),
 30 12A(R),...12D(L), 12D(R). For clarity, only two sets of loudspeakers, namely loudspeakers 14A(R) and 14A(L) and 14D(R) and 14D(L) are shown. The letters (R) and (L) designate RIGHT and LEFT stereo channels, respectively, and the letters A and D designate the zones in which the sets of loudspeakers are

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located. Although Figure 1 shows only two loudspeakers in each set, it could comprise more than two loudspeakers, such as might be used in a "surround sound" or home theater system.

The audio distribution unit 10 comprises a preamplifier unit 16 which
 5 includes the usual preamplifier and a switching matrix allowing some or all of the inputs 11 to be connected simultaneously to respective sets of the loudspeakers. The preamplifier unit 16 also has a data input port 18 for receiving data signals to control operation of the audio distribution unit 10, to vary, for example, volume, bass, treble and balance, and to operate the
 10 switching matrix to make or alter input/source selections. The construction of such a preamplifier unit 16 is known to persons skilled in this art, so it will not be described in detail here.

The audio distribution unit 10 also comprises four identical stereo power amplifier units (22A...22D), one for each set of loudspeakers, connected to the
 15 four sets of stereo output ports 12A, 12B, 12C and 12D, respectively.

The loudspeakers 14A(L) and 14A(R) have the usual "RED" and "BLACK" drive signal input terminals, indicating the polarity of the windings so that the pair of loudspeakers can be connected to the associated power amplifiers with the same polarity, thereby ensuring proper stereo reproduction.
 20 For convenience of illustration, only two of the stereo power amplifier units, 22A and 22D, are shown, and only one unit, 22A, will be described. Stereo power amplifier unit 22A comprises two identical audio power amplifiers 24A(L) and 24A(R), connected to loudspeakers 14A(L) and 14A(R), respectively, by a set of loudspeaker cables comprising first and second
 25 conductors 26A(L1) and 26A(L2) connecting the BLACK and RED terminals of LEFT loudspeaker 14A(L) to the ground terminal and line terminal 30A(L), respectively, of amplifier 24A(L) via output port 12A(L) and third and fourth conductors 26A(R1) and 26A(R2) connecting BLACK and RED terminals of the RIGHT loudspeaker 14A(R) to the ground terminal and line output terminal
 30 30A(R) respectively, of amplifier 24A(R) via output port 12A(R). LEFT stereo channel power amplifier 24A(L) has its input connected to LEFT channel line output terminal 20A(L) of preamplifier unit 16, and its power supply terminals connected to a first power supply 28 which supplies voltages V_1^+ and V_1^- , with

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amplitude of, say, +35 volts and -35 volts, respectively, relative to a first, reference ground designated as ground "1". The GROUND terminal of LEFT power amplifier 24A(L) is connected to first ground "1". The preamplifier unit 16 is shown grounded to ground 1 and supplied with power at ± 15 volts, 5 conveniently derived from power supply 28 by means of a suitable regulator circuit (not shown).

The RIGHT channel power amplifier 24A(R) is powered by supply voltages V_2^+ and V_2^- , also, say, +35 volts and -35 volts, respectively, derived from a second power supply 30, which supplies voltages V_2^+ and V_2^- relative 10 to a second ground designated as ground "2". The GROUND terminal of power amplifier 24A(R) is connected to second ground "2". Hence, the BLACK terminals of loudspeaker 14A(L) and 14A(R) are connected to first ground 1 and second ground 2, respectively.

The signal input 21A(L) of power amplifier 24A(L) is connected directly 15 to the LEFT channel output terminal 20A(L) of the preamplifier unit 16. The signal input 21A(R) of power amplifier 24A(R), however, is connected to the corresponding RIGHT channel output terminal 20A(R) of preamplifier unit 16 by way of a coupling device, specifically, a differential amplifier 32A which has its output terminal connected to the input 21A(R) of RIGHT channel power 20 amplifier 24A(R), its non-inverting input terminal (+) connected to the RIGHT channel output terminal 20A(R) of the preamplifier unit 16, and its inverting input terminal (-) connected to first ground 1. The GROUND terminal of differential amplifier 32A is connected to second ground 2.

The differential amplifier 32A will maintain the difference between its 25 output and ground 2 so that it is the same as the difference between its non-inverting and inverting inputs. The signal applied to the input 21A(R) of power amplifier 24A(R), relative to first ground 1, will be the sum of the RIGHT channel audio signal from preamplifier terminal 20A(R) and any voltage difference between first ground 1 and second ground 2. Because the output 30 30A(R) of power amplifier 24A(R) is referenced to second ground 2, however, only the audio signal will appear on the RED and BLACK loudspeaker conductors 26A(R1) and 26A(R2), as a differential mode signal. Any variations in the voltage between first ground 1 and second ground 2, however, will

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appear on both conductors 26A(R1) and 26A(R2), i.e. as common mode signals, and hence will not be "seen" by loudspeaker 14A(R). Accordingly, grounds 1 and 2 and the two loudspeaker ground (BLACK) conductors 26A(L1) and 26A(R1) may be used as a transmission path or channel for
 5 "common mode" transmission of power and signalling, as will be described later.

It should be appreciated that the connections between the loudspeaker cable conductors and the RED and BLACK terminals could be transposed, i.e. the RED terminals of the loudspeakers 14A(L) and 14A(R) could be connected
 10 to the grounds 1 and 2, instead of the BLACK terminals.

A remote control unit 34A for controlling the loudspeakers in zone A is connected across the conductors 26A(L1) and 26A(R1) and hence between the respective BLACK terminals of the loudspeakers 14A(R) and 14A(L).

Power amplifier unit 22D is identical to power amplifier unit 22A and so
 15 will not be described in detail. In Figure 1, its components have the same reference numbers as the corresponding components in power amplifier 24A, but with the zone suffix "D".

The system includes means for transferring power and signalling between the audio distribution unit 10 and the remote unit 34A via the above-
 20 mentioned transmission path. At the audio distribution unit 10, the transferring means includes a microcontroller 40, with its own 5 volt power supply 42 referenced to first ground 1, and a line interface unit 44 supplied by power supply 28 and referenced to first ground 1. The line interface unit 44 is connected to the microcontroller 40 by a first link 46 for data signals (RX)
 25 received from the remote control units 34A to 34D and a second link 48 for data signals (TX) from the microcontroller 40 which are to be transmitted to the remote control units 34A to 34D. The output of line interface unit 44 is connected to second ground 2 of power supply 30 and maintains second ground 2 at a potential difference of about 20 volts d.c. relative to ground 1.
 30 It should be noted, therefore, that second ground 2 and voltages V_2^+ and

V_2^- "float" relative to ground 1 and voltages V_1^+ and V_1^- with a potential

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difference of about 20 volts d.c., subject to variations due to modulation signals, as will be described later.

In operation, the line interface unit 44 maintains this potential difference of about 20 volts d.c. between the first and third conductors 26A(L1) and 5 26A(R1) and likewise between the corresponding "grounded" conductors of the other loudspeaker sets in zones B, C and D. Because the remote control units are connected between these "grounded" conductors, they are able to extract power at 20 volts d.c. from the interface unit 44. At each of the remote control units 34A to 34D, the afore-mentioned transferring means includes 10 means for transmitting signals to the microcontroller 40 via the signal transmission channel defined by the pair of "grounded" conductors to which it is connected, specifically by applying modulated signals to them. Also, the line interface unit 44 can transmit modulated signals to the remote units via the transmission channels.

15 Referring to Figure 2, a suitable line interface unit 44 comprises a transconductance amplifier and D.C. voltage source unit 50 formed by an amplifier 52 which is supplied with voltages V_1^+ and V_1^- from power supply 28 (Figure 1) and has a non-inverting input (+) coupled to a 20 volt d.c. source 54 by a resistor 56. The source 54 has its negative terminal connected to first 20 ground 1. The inverting input (-) of amplifier 52 is connected to its output by a first feedback loop comprising a resistor 58. An output resistor 60 connects the output of the amplifier 52 to ground 2. A second feedback loop is formed by a capacitor 62 and a resistor 64 connected in series between the second ground 2 and the non-inverting input (+) of amplifier 52. In operation, and 25 when quiescent unit 50 maintains a potential difference of about 20 volts d.c. between grounds 1 and 2.

An output terminal of a modulator 66 is connected to the inverting input (-) of amplifier 52 by a second capacitor 68 and resistor 70, in series. The input of the modulator 66 is connected to the link 48 to receive digital data 30 signals (TX) from the microcontroller 40 (Figure 1). The modulator 66 may use Frequency Shift Keying (FSK), Amplitude Shift Keying (ASK), or any other type of modulation suitable for digital data transmission. When half duplex FSK is used, suitable mark and space carrier frequencies might be 100 kHz for "1" and

90 kHz for a zero. Full duplex FSK could be implemented using two additional frequencies.

A demodulator 72 has its input connected to the output of the transconductance amplifier unit 50 (and hence to second ground 2), and its
5 output connected to the link 46 to pass demodulated digital data signals (RX) received on second ground 2 to the microcontroller 40 (Figure 1).

In use, signals received on link 48 from the microcontroller 40, for example to give the status of the controls of audio distribution unit 10, are modulated by modulator 66 and supplied to amplifier 52, which transmits them
10 onto the second ground 2 and hence broadcasts them to all of the remote control units 34A-34D. Each remote control unit has a unique address assigned during a set-up procedure and mapped to the corresponding zone by suitable entries in the memory of the microcontroller 40. The microcontroller 40 will include in each broadcast signal the address of the remote control unit
15 34A-34D for which the signal is intended. The remote control unit for which the signal is intended will detect its own address in the signal, process the signal accordingly, using known techniques, and provide a suitable response, such as displaying a setting for one of the parameters of the audio distribution unit 10. Conversely, signals applied to one of the transmission channels by
20 one of the remote control units 34A-34D, when received in the interface unit 44, will be demodulated by demodulator 72 and transmitted via link 46 to the microcontroller 40. All of the remote control units 34A-34D are effectively connected in parallel between first grounds 1 and second ground 2, and all transmitters in both the interface unit 44 and, as will be described later, the
25 remote control units, have low output impedances, of the order of 100 ohms. Consequently, as soon as one of them transmits a signal, all of the others will receive it. Each of the remote control units 34A-34D is arranged to inhibit operation of its own transmitter while its own receiver is receiving a signal, as is common in transceivers. This same feature will operate to inhibit the
30 transmitters of all other remote control units as soon as one of them begins to transmit, thereby avoiding "collisions" if two users using different ones of the remote control units 34A-34D attempt to control the audio distribution unit 10 at the same time.

The remote control units 34A-34D are identical, so only one will be described. As shown in Figure 3, a suitable remote control unit 34A comprises a power extraction circuit 76 connected across terminals 77 and 79 which, in use, are connected to the BLACK terminals of the LEFT and RIGHT
 5 loudspeakers, respectively, and hence across "grounded" first and third conductors 26A(L1) and 26A(R1), respectively. The power extraction circuit 76 supplies the various components of the remote control unit 34A, including a second microcontroller 84, at a voltage of 5 volts d.c., and has a "backlight" output 85 which supplies the backlight of a display 86. (For clarity, the supply
 10 connections to second microcontroller 84 and display 86 only are shown). The backlight supply is turned on and off by the microcontroller 84 by way of a control signal line 87 connecting a backlight on/off output of the microcontroller 84 to the power extraction circuit 76. The connection of terminal 77 to the BLACK terminal of the left loudspeaker 14A(L), i.e. to first
 15 conductor 26A(L1) and indirectly to ground 1, constitutes the ground for the remote control unit 34A.

The third conductor 26A(R1) (BLACK terminal) of the RIGHT loudspeaker 14A(R) is connected via an isolating capacitor 78 to an input of a receiver amplifier 80, the output of which is connected to the input of a demodulator
 20 82, which operates conversely to the modulator 66 in the interface unit 44 to demodulate received signals. The output of the demodulator 82 is coupled to an input of the second microcontroller 84 which controls display 86. An input device 88, for example a keypad having a set of pushbuttons, or a touch-screen overlay on display 86, is coupled to a second input of the
 25 microcontroller 84 for manual inputting of instructions or data by a user.

An output of microcontroller 84 is connected to an input of a modulator 90 which corresponds to the demodulator 72 in interface unit 44. The output of the modulator 90 is connected to an input of a transmitter amplifier 94, the output of which is coupled to the third conductor 26A(R1), and hence the
 30 second ground 2, by way of a (100Ω) termination resistor 96 which connects the output of the transmitter amplifier 94 to the isolating capacitor 78.

The power extraction circuit 76, microcontroller 84, modulator 90, demodulator 82, and amplifiers 80 and 94 constitute parts of the afore-

mentioned transferring means, the other parts being in the audio distribution unit 10.

The power extraction circuit 76, shown in detail in Figure 4, comprises a first transistor 110 with its collector connected to third conductor 26A(R1) and its emitter coupled to first conductor 26A(L1) by way of a capacitor 112. A resistor 114 and capacitor 116 are connected in series between the transistor emitter and first conductor 26A(L1). The capacitors 112 and 116 and resistor 114 form a filter to remove digital noise. A 5.6 volt Zener diode 118 and feed resistor 120 are connected in series between the conductors 26A(R1) and 26A(L1). The cathode of the Zener diode 118 is connected to the base of transistor 110 and, by way of a smoothing capacitor 122, to the local ground, i.e. conductor 26A(L1). The transistor 110 acts as a voltage follower with a voltage drop equivalent to that of one diode, and the 5 volt supply for the various components of the remote control unit is taken from the junction between resistor 114 and capacitor 116, on line 124.

The backlight supply circuitry comprises a second transistor 126 connected as a constant current source, with its emitter connected by way of a resistor 128 to conductor 26A(R1) and its collector connected to the backlight output 85 and hence to display 86 (Figure 3). Transistor 126 is biased by means of a pair of diodes 132 and 134 connected in series between conductor 26A(R1) and the base of transistor 126. A second smoothing capacitor 136 is connected in parallel with the diodes 132 and 134 to smooth switching transients.

The backlight supply transistor 126 is turned on and off by means of a switching transistor 138 which has its emitter grounded (to the local ground - conductor 26A(L1)), its collector coupled by way of a resistor 140 to the cathode of diode 134 and its base coupled by way of a resistor 142 and control line 87 to the backlight on/off output of the microcontroller 84 (Figure 3). The microcontroller 84 is programmed so that, as soon as the user operates the input device 88 (Figure 3), for example presses a key, it turns on transistor 138 which turns on the backlight current.

The microcontroller 84 stores the address for the remote unit in the system. When a user presses a key, the microcontroller 84 extracts the

address of its own remote unit and includes it in a data signal, which then is transmitted to the microcontroller 40 in the audio distribution unit 10, allowing the latter to determine the source of the data signal. The microcontroller 40 will control the audio distribution unit 10 according to the content of the data
5 signals received from the remote unit 34A and will communicate to the transmitting remote unit 34A such data signals as are appropriate.

Referring again to Figure 3, when the system is in use, signals received from the interface unit 44 via conductors 26A(R1) and 26A(L1) are amplified by receiver amplifier 80, which also provides bandpass filtering to limit the
10 signal to the transmit bandwidth of the modulator 66 (Figure 2), demodulated by demodulator 82 and supplied to the microcontroller 84, which deciphers the signals and controls the display 86 accordingly. Conversely, when a user inputs information or commands by way of input device 88, corresponding digital data signals from the microcontroller 84 are modulated by modulator 90,
15 amplified by transmitter amplifier 94 and applied to the conductor 26A(R1).

As mentioned previously, each of the remote control units 34A-34D has the ability to turn off its own transmitter when its receiver is receiving a data signal. Thus, the microcontroller 84 continuously monitors the output of the demodulator 82 and compares it with the input to the modulator 90. If they
20 are not the same, a "collision" is assumed. If the microcontroller 84 is not driving the modulator 90 and transmitter amplifier 94, and the demodulator 82 is outputting a data signal (the demodulator output has a data valid line and a data line), the microcontroller 84 inhibits its own transmissions. In a similar manner, in the audio distribution unit 10 (Figure 1), the microcontroller 40 will
25 monitor RX data line 46 and inhibit its own transmissions as required. In essence, the arrangement is like a network, with the audio distribution unit 10 acting as a hub to which the remote control units 34A-34D are connected by their respective transmission channels formed by pairs of loudspeaker conductors. When any network element is transmitting, the other elements
30 wait until the transmission ends before seizing the opportunity to transmit themselves. The network analogy also applies to the use of unique addresses for the remote units, enabling selective communications using known network signalling techniques.

Various modifications and substitutions may be made to the above-described embodiment without departing from the scope of the present invention. Thus, it is envisaged that a full wave rectifier could be interposed between the conductors 26A(R1) and 26A(L1) and the power extraction circuit 76 of each of the remote control units 34A-34D, so that the power supply to the remote control units would be independent of the polarity of the potential difference on the corresponding transmission channel. This would, of course, require the use of a polarity-insensitive signalling scheme, such as FSK or ASK.

It is also envisaged that the system could be modified to add an intercom feature, in which case the signals transmitted between the audio distribution unit 10 and the remote unit(s) 34A-34D would include baseband voice signals as well as the data signals, the latter being above baseband.

Figures 2 and 3 illustrate, in dashed lines, a modification to the interface unit 44 two alternative modifications to the remote unit 34A, which would permit intercom use, it being understood that similar modifications would be made to the other remote units 34B-34D. As shown in dashed lines in Figure 3, remote control unit 34A could have a microphone 96A connected by way of a microphone amplifier 98A and intercom ON/OFF switch 99A (conveniently controlled by microcontroller 84) directly to the conductor 26A(R1) allowing transmission of voice signals via the conductors 26A(L1) and 26A(R1) to the interface unit 44. As shown in broken lines in Figure 2, the interface unit 44 could then have a capacitor 104 or other suitable voice circuitry for extracting the voice signal and supplying it to an extra input (not shown), such as the microphone input MIC, of preamplifier unit 16, by way of a suitable connection 100 shown in broken lines in Figure 1.

Alternatively, and preferably, the voice signal could be transmitted by modulating the "backlight" current of the power extraction unit 76. Suitable additional circuitry for doing so is shown in chain-link lines in Figures 3 and 4. Thus, referring to Figure 3, microphone 96A' is connected to microphone amplifier and interface circuit 102, the output of which is connected to an input of the power extraction circuit 76. As shown in Figure 4, within the power extraction circuit 76, the output of the microphone amplifier and interface unit 102 (Figure 3) is coupled to the base of transistor 126 by way of an additional

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capacitor 144 and an additional resistor 146 connected in series. An additional resistor 148 is inserted in series with the base of transistor 126. Resistors 146 and 148 are equal in value, suitably 100 ohms, so about one half of the voice signal is applied to the base of the transistor 126. The voice signal modulates the backlight current on the collector of transistor 126, which typically has a quiescent value of about 100 mA. The backlight supply circuitry of power extraction circuit 76 is a constant current source. Consequently, variations in the backlight current will appear as changes in the voltage across first ground 1 and second ground 2 in the audio distribution unit 10. When the user speaks, the backlight current will vary in proportion to the output of microphone amplifier and interface unit 102.

It is also envisaged that the voice signal could be used to modulate the total current consumed by the remote unit.

The system may be capable of paging, in which case the message is broadcast to all of the other zones, or directed messaging, in which case the message is sent to one or more zones selected by the user. A user wishing to send, for example, broadcast voice signals, will press a corresponding key or symbol on the input device 88. The microcontroller 84 will respond by turning on the backlight (if it is not already on), detecting the "intercom" selection, generating a corresponding data signal including the address of the local remote unit 34A and transmitting it to microcontroller 40 in the audio distribution unit 10, at the same time operating switch 99A to enable voice transmissions. On receipt of the "intercom" data signal on line 46, the microcontroller 40, which is suitably programmed, will detect the address of remote unit 34A only and issue a control signal to the preamplifier unit 16 to cause the switching matrix to route the voice signals from the MIC input 18 to the loudspeaker sets in all of the other zones B, C and D, muting or reducing their existing signals as appropriate. When, subsequently, the user speaks into the microphone 96A, the voice signal will be broadcast to the loudspeakers in the other zones B, C and D. It will be appreciated that, to avoid feedback problems, the voice signal will not be transmitted to the audio transducer set in the zone from which the voice signal originated.

If the user wished to direct the message to a particular zone, he would first select the "intercom" mode and then identify the reception zone using the input device 88. The microcontroller 84, which would have all of the zone addresses stored in memory, would select the appropriate address and include
 5 it in the message, as a "recipient" address, together with the "sender" zone address. At the audio distribution unit 10, the microcontroller 40 would detect the recipient zone address and control the switching matrix so as to direct the message to the one zone only.

It will be appreciated that this intercom system could also be modified
 10 to allow the user to input several recipient addresses and send the same voice message to several selected zones, but not broadcast to all of them.

As shown in Figure 5, a suitable microphone amplifier and interface unit 102 comprises an amplifier 150 having its non-inverting input connected to the microphone 96A' by way of an ON/OFF switch 152 and a blocking capacitor
 15 154 in series. The switch 152 may be a CMOS analog switching device or other suitable device controlled by the microcontroller 84 by way of line 156. The poles of the switch 152 are connected to local ground by resistors 158 and 160, respectively. As depicted in Figure 5, the microphone 96A' is an electret microphone comprising an internal field effect transistor (not shown)
 20 connected between the 5 volt supply and, by way of a bias resistor 162, local ground. The inverting input of amplifier 150 is connected to local ground by way of a resistor 164 and, by way of a feedback resistor 166, to the output of amplifier 150, which output is connected to the capacitor 144 in power extraction unit 76 (Figure 4).

25 In use, the user will select the intercom function by means of the input device 88. The microcontroller 84 will respond by turning on the backlight current (if it is not already on), closing normally-open switch 152, and, as before, transmitting data signals to select the loudspeaker sets to receive the voice signals. When the user speaks into the microphone 96A', the amplified
 30 voice signal will modulate the current flowing through constant current source transistor 126 (Figure 4). At voice frequencies, the constant current flows through a substantially constant impedance, i.e. the output impedance of

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interface unit 44. As a result the potential difference across the conductors 26A(R1) and 26A(L1) will be modulated with the voice signal.

As before, at the interface unit 44, a capacitor 104 connected between the input of the demodulator 72 and the line 100 will extract the voice signal
5 and supply it to the appropriate input 18 of the preamplifier unit 16.

It is envisaged that the intercom arrangement could be used with a multiconductor system, the transmission channel via which the voice signals are transmitted being defined by conductors other than those used to transmit the audio signals to the audio transducers.

10 Where a zone is equipped with a set of more than two loudspeakers, with a correspondingly greater number of "ground" wires, any two of the ground wires could be selected to form the transmission channel, providing that they were connected to the separate first ground 1 and second ground 2, so that the signalling was common mode with respect to the power amplifiers and
15 loudspeakers and did not interfere with the audio signals.

For example, if there were four loudspeakers in a particular zone, two with their respective "ground" conductors connected to first ground 1 and two with their respectively "ground" conductor connected to second ground 2, the remote unit could be connected between either one of the first ground 1
20 conductors and either one of the second ground 2 conductors.

It is also envisaged that the audio distribution unit 10 could have at least a third ground 3, isolated from both first ground 1 and second ground 2, and the "ground" conductors of one or more of the audio transducer sets could be connected between the first ground 1 and the third ground 3, thereby forming
25 a separate transmission channel and providing better segregation between signals from the different remote units. Modifications to do so are illustrated in Figure 7 which shows a modified power amplifier unit 22B and the loudspeakers of zone B, together with an additional interface unit 44/2 and additional power supply 29 in the audio distribution unit 10. Other components
30 of the system, i.e. for zones A, C and D, will be as shown in Figure 1.

As shown in Figure 7, the third power supply 29 provides voltages V_3^+ and V_3^- , say +35 volts and -35 volts, relative to third ground 3,

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conveniently, but not necessarily, the same as that between first ground 1 and second ground 2. The second interface unit 44/2, which is similar in construction to first interface unit 44, is connected between the microcontroller 40 and third ground 3, and maintains third ground 3 at a predetermined voltage
 5 relative to first ground 1. It also may have a capacitor (not shown) for coupling voice signals via line 100' to a microphone input, or other suitable audio input, of the preamplifier unit 16 if the intercom feature is provided. Data links 46/2 and 48/2 couple received and transmitted data signals (RX2, TX2), respectively, between the microcontroller 40 and the third ground 3. The
 10 power amplifier unit 22B for zone B is generally similar to the power amplifier unit 22A of Figure 1 and is connected to zone B loudspeakers 14B(L) and 14B(R) in a similar manner. It differs, however, in that the "ground" terminals of differential amplifier 32B and power amplifier 24B(R), and hence the "grounded" loudspeaker conductor 26B(R1), i.e. connected to the BLACK
 15 terminal of loudspeaker 14B(R), are connected to the third ground 3. Consequently, the transmission channel for zone B is defined by conductors 26B(L1) and 26B(R3) which connect the zone B remote unit 34B to first ground 1 and third ground 3, respectively.

As before, the differential amplifier 32B ensures that the audio signal
 20 from preamplifier output 20B(R) appears as a differential signal across the conductors 26B(R1) and 26B(R2) while variations in the potential difference between ground 1 and ground 3 appear on both of these conductors, i.e. in common mode, and so are not "seen" by the loudspeaker 14B(R).

Figure 8 illustrates how, when the set of audio transducers in a zone
 25 includes a third audio transducer, it would be possible to provide a third ground in the audio distribution unit 10 and connect the "ground" conductor of the third audio transducer to the third ground.

Thus, Figure 8 shows a modified power amplifier unit 22A' connected to three loudspeakers 14A(L), 14A(R) and 14A(C) in zone A. The power
 30 amplifier unit 22A' comprises three power amplifiers 24A(L), 24A(R) and 24A(C) connected to the LEFT, RIGHT and CENTRE channel loudspeakers 14A(L), 14A(R) and 14A(C), respectively, by a set of six conductors 26A(L1) and 26A(L2), 26A(R1) and 26A(R2), and 26A(C1) and 26A(C2) connected to

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output ports 12A(L), 12(R) and 12(C). The ground terminals of power amplifiers 24A(L), 24A(R) and 24A(C) are connected to grounds 1, 2 and 3, respectively. The power amplifiers 24A(R) and 24A(C) are connected by differential amplifiers 32A(1) and 32A(2), respectively, to the corresponding
 5 outputs of preamplifier unit 16 and are supplied from power supplies 30 and 29, respectively. The audio distribution unit 10 has a modified interface unit 44' which has two outputs, one connected to ground 2 and the other connected to ground 3. A suitably modified remote unit 34A' is connected to all three ground conductors 26A(L1), 26A(R1) and 26A(C1)

10 As shown in more detail in Figure 9, the modified interface unit 44' supplies power to the modified remote unit 34A' via grounds 1 and 3 and signalling via grounds 1 and 2. Thus, the modified interface unit 44' comprises a modulator 66 and demodulator 72 connected between data lines 48 and 46, respectively, and an amplifier 52. Amplifier 52 has its non-inverting input
 15 connected to the output of modulator 66 and its output connected to second ground 2 by a resistor 60. The input of demodulator 72 is connected to second ground 2. A feedback resistor 64 interconnects the output and inverting input of amplifier 52 and the inverting input is coupled to first ground 1 by resistor 56. The modulator 66, demodulator 72 and amplifier 52 operate
 20 in a similar manner to those shown in Figure 2 to communicate signals between microcontroller 40 and remote unit 34A' via the transmission channel defined by grounds 1 and 2 and the loudspeaker conductors 26A(L1) and 26A(R1).

Power is supplied to the remote unit 34A' via first ground 1 and third ground 3 and conductors 26A(L1) and 26A(C1) by a separate power supply
 25 68 which has its negative pole coupled to first ground 1 and its positive pole coupled to third ground 3 by way of a current limiting resistor 55.

Figure 10 illustrates corresponding modifications to the remote unit 34A'. In particular, the voltage regulator comprises transistor 110, Zener diode 118, capacitors 122 and 112 and resistor 120, as in the voltage regulator of
 30 Figure 4, but without the additional smoothing capacitor 116 and resistor 114. The backlight circuitry is not shown in Figure 10 but would be similar to that shown in Figure 4.

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The receiver amplifier 80 and transmitter amplifier 94 have their ground terminals coupled to first ground 1 via conductor 26A(L1). The input of receiver amplifier 80 is connected to second ground 2 via conductor 26A(R1) and the output of transmitter amplifier 94 is connected by way of resistor 96
5 to second ground 2. Other components of remote unit 34A' will be configured in a similar manner to those of remote unit 34A (Figure 3).

With such an arrangement, the transmission channel formed by conductors 26A(L1) and 26A(C1), between first ground 1 and third ground 3, is used to transfer power from the audio distribution unit 10 to the remote unit
10 34A' and the other transmission channel defined between conductors 26A(L1) and 26A(R1), first ground 1 and second ground 2, is used to transfer signals between the audio distribution unit 10 and the remote unit 34A'.

It should be appreciated that the intercom arrangement described as a modification with reference to Figures 1 to 5 could be adapted for use with the
15 embodiment of Figure 8.

It is also envisaged that, in any of the foregoing embodiments, each remote unit could provide an interface for an infrared remote control device. As illustrated in broken lines in Figure 3, an infrared receiver 106, of the kind used in remote-controlled audio or video equipment, for receiving infrared
20 signals transmitted by a separate infrared remote control device 107, may be coupled to the microcontroller 84 of remote unit 34A. The microcontroller 84 then would be programmed to transmit corresponding signals onto the signal transmission channel. At the audio distribution unit 10, the "infrared" data signals (or source control signals) would be demodulated and detected by the
25 interface unit 44 along with other data signals from the remote units 34A-34D and passed to the microcontroller 40, which would supply corresponding source control signals to an infrared transmitter 101 (shown in broken lines in Figure 1) for communicating with one or more of the various source components. In order to differentiate the source control or "infrared" data
30 signals from other data signals, the microcontroller 84 in the remote unit would add an identifier to the source control signal, for example, several control bits. The microcontroller 40 would detect the infrared identifier in the received data signal RX and control the infrared transmitter 101 to generate a corresponding

infrared control signal and broadcast it to the infrared receiver(s) on the front(s) of the source unit(s). It should be appreciated that the infrared signals could substitute for, or supplement, the other data signals.

Most infrared remote controls use ASK modulation. Consequently,
5 where the remote unit is equipped with such an infrared receiver, the modulator could use ASK to transmit the signals from the infrared remote control device 107 for controlling the source unit (CD, tape, etc.) and use FSK to transmit the data signals for controlling operation of the audio distribution unit. The same demodulator could be used for both FSK and ASK. When the microcontroller
10 40 detected that the received signal was ASK modulated, it would pass it directly to the afore-mentioned infrared transmitter 101. If it detected that the signal was FSK modulated, however, the microcontroller 40 would determine the content of the signal and process it accordingly.

It should be appreciated that the infrared signals passed between the
15 infrared remote control device 107 and the infrared receiver 106 need not be identical to the infrared control signals passed between the infrared transmitter 101 and the audio source unit(s), so long as they convey the appropriate information. It would be convenient, of course, if they were substantially identical.

20 Various other modifications are envisaged. Thus, one or more of the differential amplifiers 32A - 32D which allow the required referencing of the signal from terminal 20A(R) to the second ground 2 could be replaced by an isolation amplifier 32A' which, as shown in Figure 6, would be connected in a similar manner to the differential amplifier 32A (Figure 1) but would provide
25 electrical isolation between its inputs and outputs, hence providing better common mode rejection. Suitable isolation amplifiers may use optical, magnetic, capacitive, or other suitable forms of isolation. It is also envisaged that a transformer could be used instead of the differential amplifier.

Where it is desired to use a larger potential difference than can be
30 tolerated by the differential amplifier 32A, a capacitor could be interposed between the input of the differential amplifier 32A and the preamplifier unit 16.

If it is required to transmit power only, the differential amplifier 32A could be replaced by a suitable capacitor connected directly between the

preamplifier output terminal 20A(R) and the input of power amplifier 24A(R). In the embodiment of Figure 8, therefore, the second differential amplifier 32A(2) could be replaced by a suitable capacitor connected directly between the preamplifier output 20A(C) and the input of power amplifier 24A(C).

5 It would be possible for the microcontrollers 40 and 84 to perform the required modulation, in which case modulators 66 (Figure 2) and 90 (Figure 3) could be omitted. It is also envisaged that the power amplifier units 24A(L)...24D(R) could be duplicate channels of a multi-channel power amplifier unit. The power amplifier units 24A(L)...24D(R), the interface unit 44, the
10 microcontroller 40, and the remote control units 34A-34D could be supplied to a customer without the preamplifier unit 16, for addition to an existing preamplifier unit capable of external control by microcontroller 40 or the like. The microcontroller 40 also could be omitted if the preamplifier unit 16 already had a microcontroller capable of interfacing with the interface unit 44.

15 Although the above-described embodiments transmit both power and signalling between the audio distribution unit 10 and the remote control units 34A-34D, the invention embraces systems which transmit power only, or signalling only. The appropriate components could then be omitted.

 Although a four-zone audio system has been described, it will be
20 appreciated that more zones could be added simply by adding loudspeaker sets and duplicating the power amplifier units; or fewer zones served with correspondingly fewer power amplifier units and loudspeaker sets. Appropriate modifications would, of course, be made to other components, such as the switching matrix and the programming of the microcontrollers 40 and 84.

25 It should be appreciated that, although an audio system has been described, specifically, the invention is applicable to audio-video systems, such as home theatre systems, which have sets of several loudspeakers in different zones. In such a case, the audio distribution unit 10 could be combined with a video distribution unit which itself might have several sources and a suitable
30 switching matrix. The video sources also might be distributed. The same microcontroller might then control the audio and video distribution.

 It should be noted that more than one remote control unit could be mapped to the same zone, for example a large room.

It should also be noted that a particular remote unit could be physically associated (co-located) with the loudspeakers of one zone, and associated logically by having an address linked to that zone by the microcontroller 40, yet be connected to the ground conductors of loudspeakers in a different zone.

5 For example, if two sets of audio transducers were located in adjacent rooms, the conductors from both sets of audio transducers might follow a common path back to the audio distribution unit, perhaps initially within a wall shared by the two rooms. In such a case, it might be convenient for both remote units to be connected to the same pair of grounded conductors and
10 share the same transmission channel. The microcontroller 40 would be able to discriminate between their signals because each remote unit would include its own address with any signals it transmitted and detect its own address in the broadcast signals it received.

Although, in the above-described embodiments, each remote unit is
15 connected to the loudspeaker terminals, each remote unit could be connected to the loudspeaker conductors anywhere along their length, or even be connected to the audio distribution unit by a pair of conductors of a multiconductor cable, which were connected to the two grounds.

It is also envisaged that any of the foregoing embodiments of the
20 invention could be modified so that two speakers and a remote unit in a zone shared four conductors, one of which connected the "ground" terminals of the remote unit and speakers to a common ground of the associated power amplifiers. Such an arrangement will now be described with reference to Figure 11. The system shown in Figure 11 is similar to that shown in Figure
25 1 and corresponding components have the same reference number. The system of Figure 11 differs from that of Figure 1, however, as follows: (i) the power amplifiers 24A(L), 24A(R), ..., 24D(L), 24D(R) share the same ground 1; (ii) the input terminals 21A(R), ..., 21D(R) of the RIGHT channel power amplifiers 24A(R), ..., 24D(R), respectively, are connected directly to output ports
30 20A(R), ..., 20D(R), respectively, of the preamplifier unit 16; (iii) the second power supply unit 30 and ground 2 are omitted; (iv) both loudspeakers 14A(L) and 14A(R) have their BLACK terminals connected in common to the first conductor 26A(L1) and thereby to ground 1; (v) the terminal of remote unit

34A which, in the system of Figure 1, would have been connected to the BLACK terminal of speaker 14A(R) is connected directly to the interface unit 44 by a conductor 26A(R1)'; (vi) in zone D, both loudspeakers 14D(L) and 14D(R) have their BLACK terminals connected in common to the conductor 26D(L1) and thereby to ground 1; (vii) the terminal of remote unit 34D which, in the system of Figure 1, would have been connected to the BLACK terminal of speaker 14D(R) is connected directly to the interface unit 44 by a conductor 26D(R1)'. At the interface unit 44, the two conductors 26A(R1)' and 26D(R1)' are connected in common. The primes signify that, in a practical system, the conductors 26A(R1)' and 26D(R1)' would be the loudspeaker conductors 26A(R1) and 26D(R1) of Figure 1 connected differently at their respective ends.

The other terminals of the remote units 34A and 34D are connected to respective BLACK terminals of the audio transducers 14A(L) and 14D(L), respectively, as in the embodiment of Figure 1.

In this modified system, one of the usual four conductors used to connect the loudspeakers to the audio distribution unit 10, two conductors per loudspeaker, is reallocated. Thus, in zone A, the first conductor 26A(L1) serves to connect the BLACK terminals of both of the two loudspeakers 14A(L) and 14A(R) to the audio distribution unit ground 1, and the second and fourth conductors 26A(L2) and 26A(R2), with the first conductor, supply the RIGHT and LEFT channel audio signals to the LEFT and RIGHT loudspeakers, respectively. The third conductor 26A(R1)', and the first conductor 26A(L1) supply power to the remote unit 34A and communicate control signals to and from it. The speakers and remote units in the other zones are connected in a similar manner.

Generally, it is envisaged that, providing the distance between the loudspeakers and the audio distribution unit 10 is not too great, cross talk or reduction in stereo separation will not be unacceptable. Where the situation precludes it, perhaps because the distance is much greater, the embodiments which use separate grounds would be used instead.

Although the above-described embodiments have been shown, and described, with the audio distribution unit connected to the various sets of

audio transducers, the invention comprehends audio distribution units and remote units supplied as a kit, without the audio transducers. Moreover, although the above-described embodiments have input means for receiving audio signals from separate sources, the input means could itself comprise one
5 or more of such sources, for example as a tuner-amplifier unit with input ports for connecting to a tape deck, CD player, DVD player, and so on.

Although embodiments of the invention have been described and illustrated in detail, it is to be clearly understood that the same are by way of illustration and example only and not to be taken by way of limitation, the spirit
10 and scope of the present invention being limited only by the appended claims.

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CLAIMS

What is claimed is:

1. Apparatus comprising an audio distribution unit (10) having means (11) for providing audio signals from audio sources and output ports (12A...12D)
 5 for supplying sets of audio transducers, each said set comprising at least a first audio transducer (14A(L)) and a second audio transducer (14A(R)) and having associated therewith at least one remote unit (34A), the first and second audio transducers (14A(L), 14A(R)) and the remote unit (34A) being connected to the audio distribution unit (10) by a set of four conductors, the apparatus further
 10 comprising means (22A...22D, 44, 76, 80-96) for supplying audio signals to the audio transducers (14A(L), 14A(R)) and transferring at least one of power and data signals between said audio distribution unit (10) and said remote unit (34A), all by way of the four conductors.

- 15 2. Apparatus according to claim 1, wherein the audio distribution unit (10) comprises first and second power amplifiers (24A(L), 24A(R)), a first conductor (26A(L1)) connects one terminal of each of said first and second audio transducers (14A(L), 14A(R)) and one terminal of the remote unit (34A) to a ground terminal of the audio distribution unit (10), a second (26A(L2)) of the
 20 conductors connects a drive-signal terminal of said first audio transducer (14A(L)) to a corresponding output terminal (30A(L)) of said first power amplifier (24A(L)), a third conductor (26A(R1)) connects a second terminal of the remote unit (34A) to an interface unit (44) of the audio distribution unit (10), and a fourth conductor (26A(R2)) connects a drive-signal terminal of said
 25 second audio transducer (14A(R)) to a corresponding output terminal (30A(R)) of said second power amplifier (24A(R)), the audio signals being supplied by way of the first, second and fourth conductors and the at least one of power and data signals being transferred by way of the first and third conductors.

- 30 3. Apparatus according to claim 1, wherein the audio distribution unit (10) comprises first and second power amplifiers (24A(L), 24A(R)) each connected to a respective one of two grounds (1,2) that are separated electrically from each other, a first conductor (26A(L1)) connects one drive-signal terminal of

the first audio transducer (14A(L)) to a ground terminal of the first power amplifier (24A(L)), a second conductor (26A(L2)) connects the other drive-signal terminal of the first audio transducer (14A(L)) to an output terminal (30A(L)) of the first power amplifier (24A(L)), a third conductor (26A(R1)) connects one terminal of the second audio transducer (14A(R)) to a ground terminal of the second power amplifier (24A(R)), and a fourth conductor (26A(R2)) connects the other drive-signal terminal of the second audio transducer (14A(R)) to a corresponding output terminal (30A(R)) of the second power amplifier (24A(R)), the audio signals being supplied to the first audio transducer (14A(L)) by way of the first and second conductors, and to the second audio transducer (14A(R)) by way of the third and fourth conductors, and said at least one of power and data signals being transferred by way of the first (26A(L1)) and third (26A(R1)) conductors.

4. Apparatus comprising an audio distribution unit (10) having means (11) for providing audio signals from audio sources and output ports (12A...12D) each having a line terminal and a ground terminal, at least one set of audio transducers, said set comprising at least a first audio transducer (14A(L)) and a second audio transducer (14A(R)) connected to a respective set (12A(L), 12A(R)) of said output ports by a set of conductors comprising a first conductor (26A(L1)), a second conductor (26A(L2)), a third conductor (26A(R1)) and a fourth conductor (26A(R2)), said apparatus further comprising at least one remote unit (34A) associated with said set of audio transducers, wherein the audio distribution unit (10) has a first ground (1) and a separate second ground (2), said set of output ports including a first output port (12A(L)) and a second output port (12A(R)) connected to the first ground (1) and second ground (2), respectively, said first conductor (26A(L1)) and said second conductor (26A(L2)) connecting respective drive-signal terminals of said first audio transducer (14A(L)) to the ground terminal and line terminal, respectively, of said first output port (12A(L)), and said third conductor (26A(R1)) and said fourth conductor (26A(R2)) connecting respective drive signal terminals of said second audio transducer (14A(R)) to the ground terminal and line terminal, respectively, of said second output port (12A(R)),

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the remote unit (34A) being connected to said first ground (1) and said second ground (2), the apparatus further comprising means (22A...22D,44,76,80-96) for transferring at least one of power and data signals between said audio distribution unit (10) and said remote unit (34A) by way of said first ground (1) and said second ground (2).

5. Apparatus according to claim 4, wherein the remote unit (34A) is connected to the first ground (1) and second ground (2) by the first conductor (26A(L1)) and the third conductor (26A(R1)), respectively, the power or data signals being transferred by way of these first and second conductors.

6. Apparatus according to claim 4, wherein the remote unit (34A) is connected to said first ground (1) and said second ground (2) by conductors other than the conductors connected to the drive signal terminals of the audio transducers (26A(L1),26A(L2),26A(R1),26A(R2)).

7. Apparatus according to claim 4, comprising a plurality of said sets of at least two audio transducers (14A(L),14A(R),...14D(L),14D(R)) connected to a corresponding plurality of sets of said output ports, and a corresponding plurality of remote units (34A...34D), each remote unit being connected to said first ground (1) and second ground (2) and associated with a respective one of the sets of audio transducers, and wherein the transferring means (44,76,80-96) comprises, at each of said remote units (34A-34D), means (84,90,94) for transmitting to said audio distribution unit (10) data signals including an identifier specific to the set of audio transducers with which that remote unit is associated and, at the audio distribution unit (10), means (40,44) for receiving said data signals and using said identifier to identify the remote unit, whereby the data signal was transmitted.

8. Apparatus according to claim 4, comprising a plurality of said sets of audio transducers (14A(L),14A(R),...14D(L),14D(R)) connected to a plurality of said sets of output ports (12A...12D) by a plurality of said sets of conductors, and a corresponding plurality of said remote units (34A,...34D), connected to

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said first ground (1) and second ground (2), wherein at least one of said remote units (34A,...34D) is co-located with one of said sets of audio transducers and connected to the first ground (1) and the second ground (2) by one or more conductors of a said set of conductors connecting a different set of audio
 5 transducer units (14A,...14D) to the audio distribution unit (10) and the transferring means (44,76,80-96) comprises, at each of said remote units (34A-34D), means (84, 90, 94) for transmitting to said audio distribution unit (10) data signals each including an identifier specific to the set of audio transducers with which that remote unit is associated and, at the audio
 10 distribution unit (10), means (40,44) for receiving said data signals and using said identifier to identify the transmitting remote unit.

9. Apparatus according to claim 4, wherein the audio distribution unit (10) has a third ground (3) separate from both of the first ground (1) and the second
 15 ground (2), said set of output ports includes a third output port (12A(C)), said at least one set of audio transducers (14A(L),14A(R)) comprises a third audio transducer (14A(C)) and said set of conductors further comprises a fifth conductor (26A(C1)) and a sixth conductor (26A(C2)) connecting drive terminals of said third audio transducer (14A(C)) to ground and line terminals,
 20 respectively, of said third output port (30A(C)), the remote unit (34A') is connected also to the third ground (3), and the transferring means (44',76,80-96) is connected to the first, second and third grounds (1,2,3) for transference of power to the remote unit (34A') via one pair of the three grounds and transference of said data signals via a different pair of the three grounds
 25 (1,2,3).

10. Apparatus according to claim 4, wherein there is provided at least a second set of audio transducers (14B(L), 14B(R)) connected to the audio distribution unit (10) by a second set of conductors and a second remote unit
 30 (34B) associated therewith, and the audio distribution unit (10) has a third ground (3) separate from the first ground (1) and second ground (2) and a second set of said output ports (12B(L), 12B(R)) each having a ground terminal and a line terminal, the second set of output ports included one port (12B(L))

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with its ground terminal connected to the ground (1) and another port (12B(R)) with its terminal connected to the third ground (3), drive terminals of one of the audio transducers (14B(L)) of the second set of audio transducers being connected to said one port (12B(L)) and drive terminals of a second audio
 5 transducer (14B(R)) of said second set being connected to said another output port (12B(R)), and wherein the transferring means (44,44/2,76,80-96) is connected also between said first ground (1) and said third ground (3) for transferring either or both of power and data signals also between said audio distribution unit (10) and said second remote unit (34B), the apparatus further
 10 comprising means (40) for determining the set of audio transducers to which a particular one of said data signals relates.

11. Apparatus according to claim 10, wherein the transferring means (44,44/2) is coupled to the first and second sets of audio transducers,
 15 respectively, and to the first, second and third grounds (1,2,3) the determining means (40) determining the set of audio transducers according to the interface unit whereby the data signal is communicated.

12. Apparatus according to claim 4, wherein the transferring means
 20 comprises, at the or each remote unit (34A-34D), transmitter means (84,90,94) coupled to the grounds for transmitting said data signals to the audio distribution unit (10) and, at the audio distribution unit (10), receiver means (72) coupled to said grounds for receiving said data signals and means (40) responsive to said data signals for controlling operation of said audio
 25 distribution unit (10) in dependence thereupon.

13. Apparatus according to claim 4, wherein the transferring means (44,76,80-96) comprises, at the audio distribution unit (10) transmitter means (50,66) for transmitting data signals via said grounds and the or each remote
 30 unit (34A...34D) further comprises corresponding receiver means (82) connected between the said grounds for receiving said data signals.

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14. Apparatus according to claim 4, wherein said transferring means (44,76,80-96) comprises means (50,54) for establishing a potential difference between the pair of grounds used for power transfer and the remote unit (34A...34D) comprises means (76) coupled to the grounds for extracting power
5 therefrom.

15. Apparatus according to claim 4, wherein the audio distribution unit (10) comprises a preamplifier unit (16) and one or more power amplifier units (22A...22D), each connected to a corresponding set of audio transducers by
10 way of the corresponding set of output ports 12A...12D, the or each audio power amplifier unit 22A...22D comprising a first power amplifier (24A(L)...24D(L)), a second power amplifier (24A(R)...24D(R)) and a coupling device (32A...32D), the first power amplifier (24A(L)...24D(L)) having a ground terminal and an output terminal connected to, respectively, the ground terminal
15 and line terminal of one (12A(L)...12D(L)) of the set of output ports, the ground terminal being connected also to said first ground (1), and an input terminal (21A(L)...21D(L)) connected to a first audio output port (20A(L)...20D(L)) of the preamplifier unit (16), the second power amplifier (24A(R)...24D(R)) having a ground terminal and an output terminal connected to, respectively, the
20 ground terminal and line terminal of another (12A(R)...12D(R)) of the set of output ports, said coupling device (32A...32D) connecting a second audio output port (20A(R)...20D(R)) of the preamplifier (16) to an input (21A(R)...21D(R)) of the second power amplifier (24A(R)...24D(R)).

25 16. Apparatus according to claim 15, wherein the or each coupling device (32A...32D) comprises a differential amplifier having a signal output terminal and an output ground terminal connected to, respectively, an input (21A(R)...21D(R)) of the second power amplifier (24A(R)...24D(R)) and said second ground (2), a first input terminal connected to said first ground (1), and
30 a second input terminal connected to said second audio output port (20A(R)...20D(R)) of the preamplifier unit (16).

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17. Apparatus according to claim 15, wherein the or each coupling device (32A') comprises an isolation amplifier having an input connected to said second audio output port (20A(R)) of the preamplifier unit (16), and an output connected to an input (21A(R)) of the second power amplifier (24A(R)).

5

18. Apparatus according to claim 15, wherein the coupling device (32A) comprises a capacitor connected between said second audio output port (20A(R)) of the preamplifier unit (16) and an input of the second power amplifier (24A(R)).

10

19. Apparatus according to claim 4, wherein the remote unit (34A...34D) comprises infrared receiver means (106) for receiving an infrared signal from a separate control unit (107) and transmitter means (84,90,94) for transmitting corresponding signals via said grounds, and the audio distribution unit
15 comprises means for detecting said corresponding signals.

20. Apparatus according to claim 4, wherein the remote unit (34A...34D) comprises voice circuitry (102) for converting signals from a microphone (96A') into voice signals for transmission to said audio distribution unit (10) and the
20 audio distribution unit (10) further comprises means (16) for relaying the voice signals to one or more of said sets of audio transducer units.

21. Apparatus according to claim 20, wherein the remote unit (34A) includes means (84,90,94) for associating with the voice signal an address of said
25 remote unit (34A) and the audio distribution unit (10) comprises means (40,16) for detecting said address and broadcasting the voice signals to audio transducer sets other than that from which the voice signals originated.

22. Apparatus according to claim 20, wherein the remote unit further
30 comprises means (84,90,94) for associating with the voice signals an address of a recipient set of audio transducers to which the voice signals are to be sent, and the audio distribution unit (10) comprises means (40,16) for

detecting the address and controlling the audio distribution unit to transmit the voice signal to the recipient set of audio transducers.

23. Apparatus according to claim 14, wherein the means (76) for extracting
5 power comprises means response to a voice control signal for modulating at least a part of the power extracted for the remote unit, the power transfer means at the audio distribution unit (10) comprises a constant current source and the audio distribution unit further comprises means for detecting the modulation as a corresponding fluctuation of a potential difference between
10 said grounds to recover the voice signals and relaying the voice signals to one or more of said sets of audio transducer units, the remote unit further comprising voice circuitry (102) for converting signals from a microphone (96A) to produce said voice control signal.

15 24. Apparatus according to claim 23, wherein the remote unit has a backlight supplied with current by the power extraction unit and the power modulating means modulates the backlight current with the voice control signal.

20 25. Apparatus comprising an audio distribution unit (10) and a plurality of sets each of at least one audio transducer (14A(L), 14A(R)) connected by a signal transmission path to the audio distribution unit, at least one of the sets of at least one audio transducer being remote from the audio distribution unit and having a remote unit (34A) associated therewith, the remote unit
25 comprising voice circuitry (102) for transmitting voice signals from a microphone (96A') via a transmission channel interconnecting the remote unit and the audio distribution unit, and means (84,90,94) for adding to the signals at least one address corresponding to one of the sets of audio transducers, and the audio distribution unit (10) comprises means (104) for extracting the voice
30 signals from the transmission channel and applying the extracted voice signals to an input of the audio distribution unit (10), and means (16) for detecting the address and, in dependence thereon, directing the voice signals to one or more audio transducer sets other than that from which the voice signals originated.

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26. Apparatus according to claim 25, wherein the means (84,90,94) for adding at least one address is arranged to add the address of at least one of the other audio transducer sets .

5 27. Apparatus according to claim 25, wherein the means (84,90,94) for adding at least one address adds only the address of the remote unit from which the voice signal is being transmitted, and the audio distribution unit comprises means (16,40) for detecting said address and distributing the voice
10 remote unit transmitting the voice signal.

28. Apparatus according to claim 25, wherein said transmission path and said transmission channel comprise the same medium.

15 29. Apparatus according to claim 28, wherein said transmission channel comprises a pair of conductors interconnecting one or more of the audio transducer units and the audio distribution unit.

30. Apparatus according to claim 4, further comprising an infrared
20 transmitter coupled to the audio distribution unit for transmitting infrared control signals for controlling one of the audio sources, and wherein the remote unit comprises a receiver for receiving, from an infrared remote control device, infrared signals corresponding to said infrared control signals, generating in
25 response thereto corresponding source control signals, and transmitting the source control signals to the audio distribution unit, the audio distribution unit having means for detecting said source control signals and controlling the infrared transmitter in dependence thereupon to generate said infrared control signals.

30 31. Apparatus according to claim 30, wherein said means for transmitting said data signals and said means for transmitting the source control signals transmit their respective signals via the same transmission path but each using a different form of modulation.

40

32. Apparatus according to claim 31, wherein the means for transmitting the source control signals uses amplitude shift keying (ASK) and the means for transmitting the data signals uses frequency shift keying (FSK).

5 33. Apparatus comprising an audio distribution unit (10) having means (11) for providing audio signals from at least one audio source controllable by infrared control signals, at least one audio transducer at a location remote from the audio distribution unit and connected thereto by a signal transmission path, and a remote unit associated with the audio transducer and connected thereto
10 by a transmission channel, the apparatus further comprising an infrared transmitter for transmitting said infrared control signals for controlling said audio source, and wherein the remote unit comprises a receiver for receiving, from an infrared remote control device, infrared signals corresponding to said infrared control signals, generating in response thereto corresponding source
15 control signals, and transmitting the source control signals to the audio distribution unit by way said transmission channel, the audio distribution unit having means for detecting said source control signals and controlling the infrared transmitter in dependence thereupon to generate said infrared control signals.

20

34. Apparatus according to claim 33, wherein the means for transmitting said data signals and said means for transmitting the source control signals transmit their respective signals via the same transmission path but each using a different form of modulation.

25

35. Apparatus according to claim 30, wherein the means for transmitting the source control signals uses amplitude shift keying (ASK) and the means for transmitting the data signals uses frequency shift keying (FSK).

30 36. Apparatus according to claim 33, wherein said transmission path and said transmission channel comprise the same medium.

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37. Apparatus according to claim 36, wherein said transmission channel comprises a pair of conductors interconnecting one or more of the audio transducer units and the audio distribution unit.

5 38. An audio distribution unit (10) for the apparatus of claim 4 and having means (11) for providing audio signals from audio sources, a first ground (1), a separate second ground (2), at least one set of output ports (12A...12D) each having a line terminal and a ground terminal, said set of output ports including a first output port (12A(L)) and a second output port (12A(R)), each
10 having a line terminal and a ground terminal, having their respective ground terminals connected to the first ground (1) and second ground (2), respectively, and their line terminals for connection to a first and a second, respectively, of a corresponding set of audio transducers, and means (44) for transferring by way of said first ground (1) and said second ground (2) and said first and
15 second output ports at least one of power and data signals between said audio distribution unit (10) and a remote unit (34A) associated with said set of audio transducers.

39. An audio distribution unit according to claim 38, for use with a plurality
20 of said sets of at least two audio transducers (14A(L), 14A(R), ... 14D(L), 14D(R)) connected to a corresponding plurality of sets of said output ports, and a corresponding plurality of remote units (34A...34D), each remote unit being connected to said first ground (1) and second ground (2) and associated with a respective one of the sets of audio transducers, the transferring means
25 (76,80-96) at each of said remote units (34A-34D) comprising means (84,90,94) for transmitting to said audio distribution unit (10) data signals including an identifier specific to the set of audio transducers with which that remote unit is associated, the audio distribution unit (10) further comprising means (40,44) for receiving said data signals and using said identifier to
30 identify the remote unit whereby the data signal was transmitted.

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40. An audio distribution unit according to claim 39, for use in apparatus according to claim 8, further comprising means (40,44) for receiving said data signals and using said identifier to identify the transmitting remote unit.

5 41. An audio distribution unit according to claim 38, for use in the apparatus according to claim 9, further comprising a third ground (3) separate from both of the first ground (1) and the second ground (2), and wherein said set of output ports includes a third output port (12A(C)), and the transferring means (44') is connected to the first, second and third grounds (1,2,3) for
10 transference of power to the remote unit (34A') via one pair of the three grounds and transference of said data signals via a different pair of the three grounds (1,2,3).

42. An audio distribution unit according to claim 38, for use in apparatus
15 according to claim 10, further comprising a third ground (3) separate from the first ground (1) and second ground (2) and further comprising a second set of said output ports (12B(L), 12B(R)) each having a ground terminal and a line terminal, the second set of output ports included one port (12B(L)) with its ground terminal connected to the first ground (1) and another port (12B(R))
20 with its ground terminal connected to the third ground (3), drive terminals of one of the audio transducers (14B(L)) of the second set of audio transducers being connected to said one port (12B(L)) and drive terminals of a second audio transducer (14B(R)) of said second set being connected to said another output port (12B(R)), and wherein the transferring means (44,44/2) is connected also
25 between said first ground (1) and said third ground (3) for transferring either or both of power and data signals also between said audio distribution unit (10) and said second remote unit (34B), the audio distribution unit further comprising means (40) for determining the set of audio transducers to which a particular one of said data signals relates.

30

43. An audio distribution unit according to claim 42, for use in apparatus according to claim 10, wherein the transferring means (44,44/2) comprises a first interface unit and a second interface unit coupled to the first and second

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sets of output ports, respectively, and to different pairs of the first, second and third grounds (1,2,3), the determining means (40) determining the set of audio transducers according to the interface unit whereby the data signal is communicated.

5

44. An audio distribution unit according to claim 38, for use in apparatus according to claim 12, further comprising receiver means (72) coupled to said grounds for receiving said data signals and means (40) responsive to said data signals for controlling operation of said audio distribution unit (10) in
10 dependence thereupon.

45. An audio distribution unit according to claim 38, for use in apparatus according to claim 13, wherein the transferring means (44) comprises transmitter means (50,66) for transmitting data signals via said grounds.

15

46. An audio distribution unit according to claim 38, for use in apparatus according to claim 14, wherein said transferring means (44) comprises means (54) for establishing a potential difference between the pair of grounds used for power transfer .

20

47. An audio distribution unit according to claim 38, for use in apparatus according to claim 15, further comprising a preamplifier unit (16) and one or more power amplifier units (22A...22D), each connected to a corresponding set of audio transducers by way of the corresponding set of output ports
25 (12A...12D), the or each audio power amplifier unit (22A...22D) comprising a first power amplifier (24A(L)...24D(L)), a second power amplifier (24A(R)...24D(R)) and a coupling device (32A...32D), the first power amplifier (24A(L)...24D(L)) having a ground terminal and an output terminal connected to, respectively, the ground terminal and line terminal of one (12A(L)...12D(L))
30 of the set of output ports, the ground terminal being connected also to said first ground (1), and an input terminal (21A(L)...21D(L)) connected to a first audio output port (20A(L)...20D(L)) of the preamplifier unit (16), the second power amplifier (24A(R)...24D(R)) having a ground terminal and an output

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terminal connected to, respectively, the ground terminal and line terminal of another (12A(R)...12D(R)) of the set of output ports, said coupling device (32A...32D) connecting a second audio output port (20A(R)...20D(R)) of the preamplifier (16) to an input (22A(R)...22D(R)) of the second power amplifier
 5 (24A(R)...24D(R)).

48. An audio distribution unit according to claim 47, for use in apparatus according to claim 16, wherein the or each coupling device (32A...32D) comprises a differential amplifier having a signal output terminal and an output
 10 ground terminal connected to, respectively, an input (22A(R)...22D(R)) of the second power amplifier (24A(R)...24D(R)) and said second ground (2), a first input terminal connected to said first ground (1), and a second input terminal connected to said second audio output port (20A(R)...20D(R)) of the preamplifier unit (16).

15

49. An audio distribution unit according to claim 47, for use in apparatus according to claim 17, wherein the or each coupling device (32A') comprises an isolation amplifier having an input connected to said second audio output port (20A(R)) of the preamplifier unit (16), and an output connected to an input
 20 (22A(R)) of the second power amplifier (24A(R)).

50. An audio distribution unit according to claim 47, for use in apparatus according to claim 18, wherein the coupling device (32A) comprises a capacitor connected between said second audio output port (20A(R)) of the preamplifier
 25 unit (16) and an input of the second power amplifier (24A(R)).

51. An audio distribution unit according to claim 38, for use in apparatus according to claim 19, and comprising means for detecting, between said grounds, signals corresponding to infrared signals received by the remote unit.
 30

52. An audio distribution unit according to claim 39, for use in apparatus according to claim 20, further comprises means (16) for relaying the voice

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signals to one or more of said sets of output ports for said audio transducer units.

53. An audio distribution unit according to claim 52, for use in apparatus
5 according to claim 21, further comprising means for detecting said address and broadcasting the voice signals to audio transducer sets other than that from which the voice signals originated.

54. An audio distribution unit according to claim 52, for use in apparatus
10 according to claim 22, further comprising means (40,16) for detecting the address and controlling the audio distribution unit to transmit the voice signal to the recipient set of audio transducers.

55. An audio distribution unit according to claim 46, for use in apparatus
15 according to claim 23, wherein the power transfer means (10) comprises a constant current source and the audio distribution unit further comprises means (40,16) for detecting the modulation as a corresponding fluctuation of a potential difference between said grounds to recover the voice signals and relaying the voice signals to one or more of said sets of output ports for the
20 audio transducer units.

56. An audio distribution unit (10) for use with a plurality of sets each of at least one audio transducer (14A(L), 14A(R)) connected by a signal transmission path to the audio distribution unit, at least one of the sets of at least one audio
25 transducer being remote from the audio distribution unit and having a remote unit (34A) associated therewith, the remote unit comprising voice circuitry (102) for transmitting voice signals from a microphone (96A') via a transmission channel interconnecting the remote unit and the audio distribution unit, and means (84,90,94) for adding to the signals at least one address
30 corresponding to one of the sets of audio transducers, said audio distribution unit (10) comprising means (104) for extracting the voice signals from the transmission channel and means (16) for detecting the address and, in

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dependence thereon, directing the voice signals to one or more audio transducer sets other than that from which the voice signals originated.

57. An audio distribution unit according to claim 56, wherein the detecting
5 means (16,40) is arranged to detect said address of at least one of the other audio transducer sets and direct said voice signals thereto.

58. An audio distribution unit according to claim 57, wherein the means
(16,40) for detecting said address is arranged to distribute the voice signal to
10 all audio transducer sets other than that set associated with the remote unit transmitting the voice signal.

59. An audio distribution unit according to claim 38, for use with a remote unit comprising a receiver for receiving, from an infrared remote control device,
15 infrared signals corresponding to said infrared control signals, generating in response thereto corresponding source control signals, and transmitting the source control signals to the audio distribution unit, the audio distribution unit having an infrared transmitter coupled to the audio distribution unit for transmitting infrared control signals for controlling one of the audio sources,
20 and means for detecting said source control signals and controlling the infrared transmitter in dependence thereupon to generate said infrared control signals.

60. An audio distribution unit according to claim 59, wherein said receiver is arranged to receive and demodulate both said data signals and said source
25 control signals.

61. An audio distribution unit according to claim 60, wherein the receiver comprises means for demodulating said source control signals modulated using amplitude shift keying (ASK) and said data signals modulated using frequency
30 shift keying (FSK).

62. An audio distribution unit (10) having means (11) for providing audio signals from at least one audio source controllable by infrared control signals,

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and configured for use with at least one audio transducer at a location remote from the audio distribution unit and connected thereto by a signal transmission path, and a remote unit associated with the audio transducer and connected thereto by a transmission channel, the remote unit comprising a receiver for
5 receiving, from an infrared remote control device, infrared signals corresponding to said infrared control signals, generating in response thereto corresponding source control signals, and transmitting the source control signals to the audio distribution unit by way of said transmission channel, the audio distribution unit having an infrared transmitter for transmitting said infrared control signals for
10 controlling said audio source, and means for detecting said source control signals and controlling the infrared transmitter in dependence thereupon to generate said infrared control signals.

63. An audio distribution unit according to claim 62, wherein the receiver
15 comprises means for receiving both said data signals and said source control signals via the same transmission path.

64. An audio distribution unit according to claim 62, wherein the receiver comprises means for demodulating said source control signals modulated using
20 amplitude shift keying (ASK) and said data signals modulated using frequency shift keying (FSK).

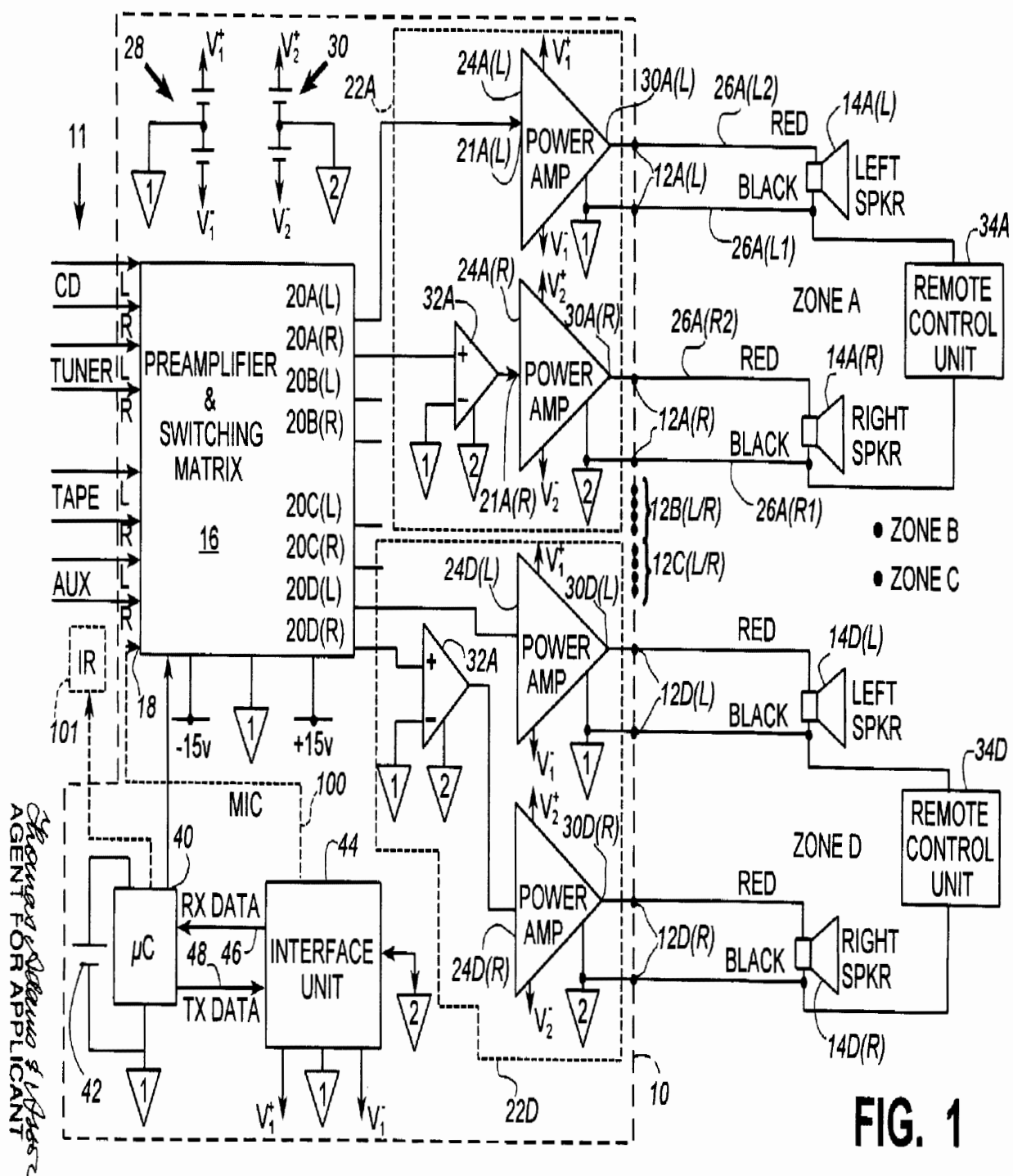
65. An audio distribution unit according to claim 38, in an assemblage with one or more of said remote units.
25

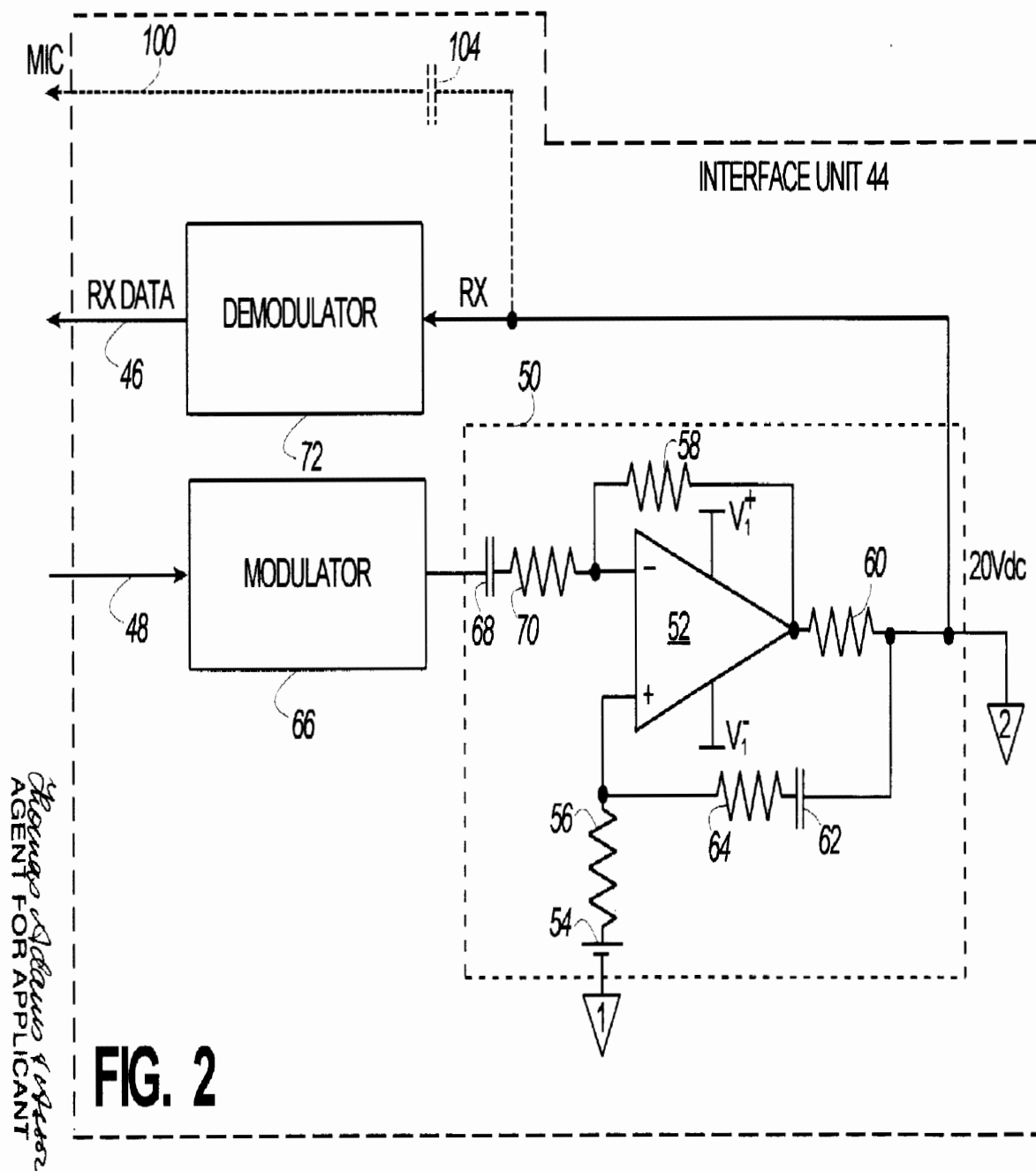
66. Apparatus according to claim 1, wherein the remote unit (34A...34D) comprises infrared receiver means (106) for receiving an infrared signal from a separate control unit (107) and transmitter means (84,90,94) for transmitting corresponding signals to the audio distribution unit (10), and the audio
30 distribution unit comprises means for detecting said corresponding signals.

67. Apparatus according to claim 1, wherein the remote unit (34A...34D) comprises voice circuitry (102) for converting signals from a microphone (96A')

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into voice signals for transmission to said audio distribution unit (10) and the audio distribution unit (10) further comprises means (16) for relaying the voice signals to one or more of said sets of audio transducer units.





Theresa Adams, Attorney
for Applicant

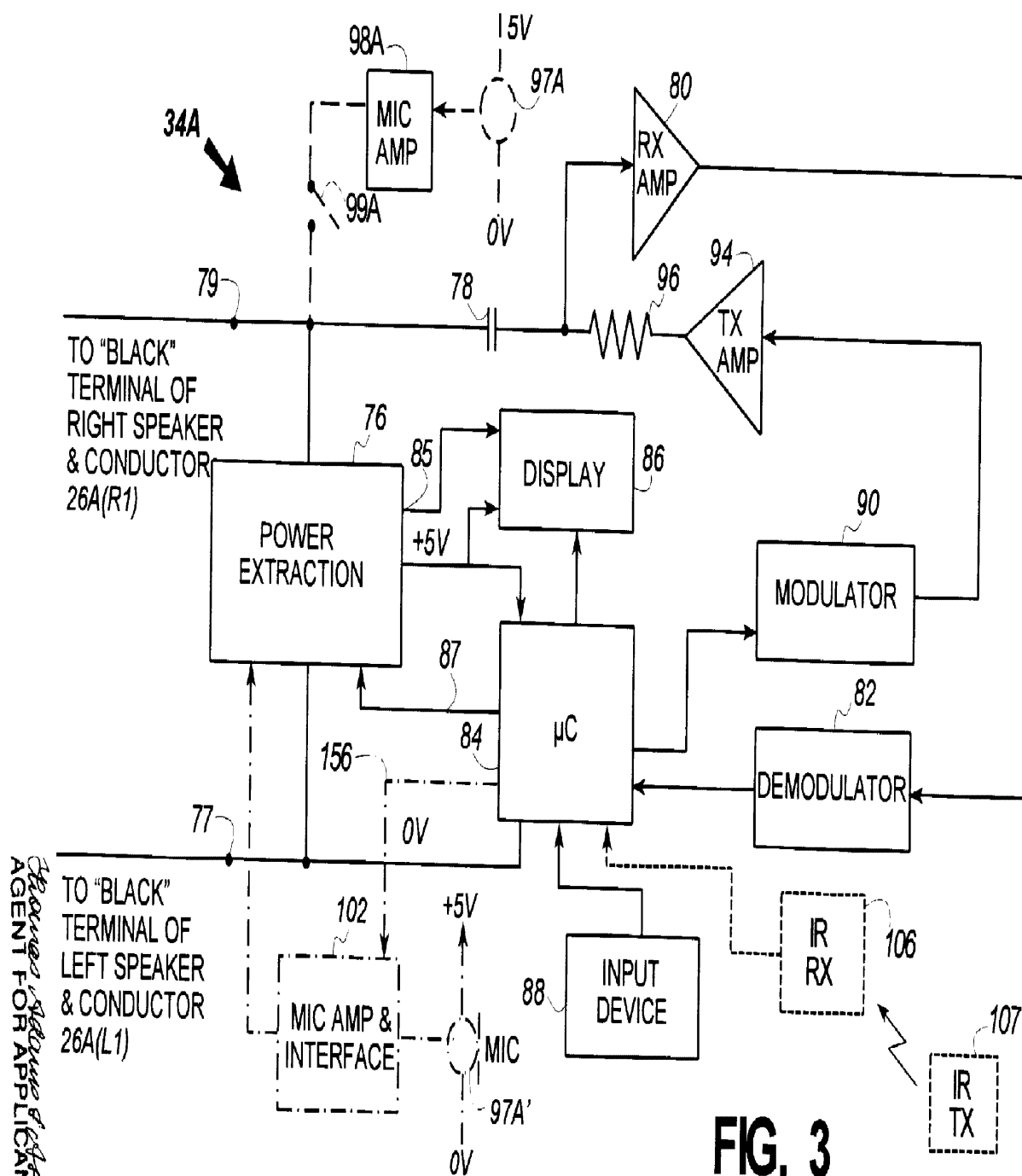


FIG. 3

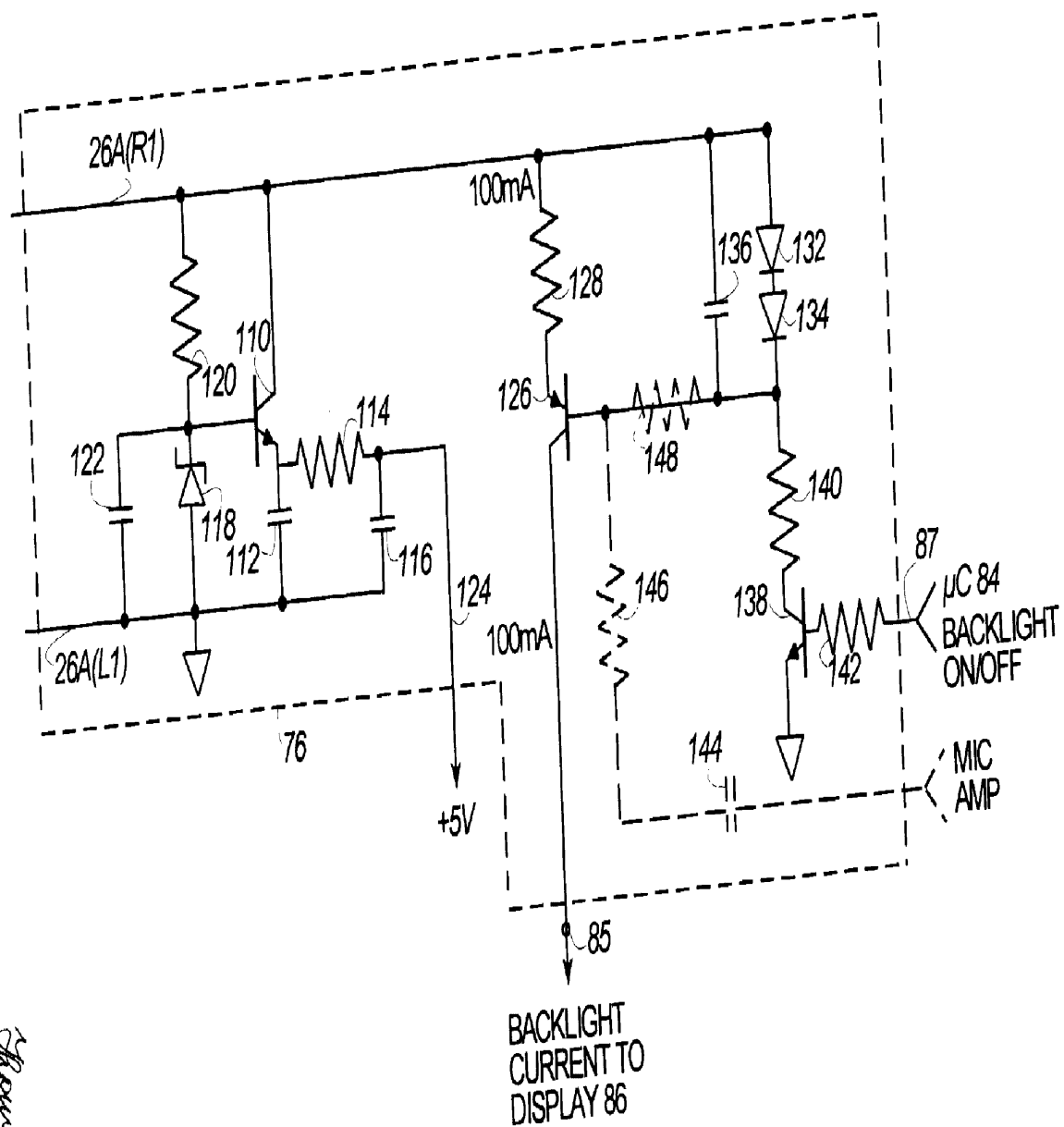


FIG. 4

Shawna Adams & Associates
AGENT FOR APPLICANT

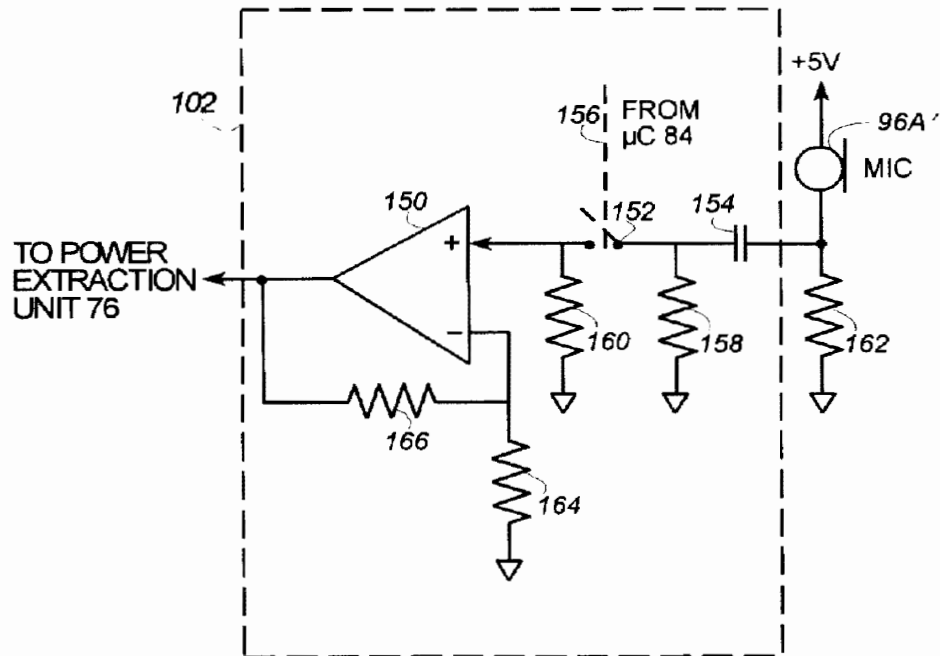


FIG. 5

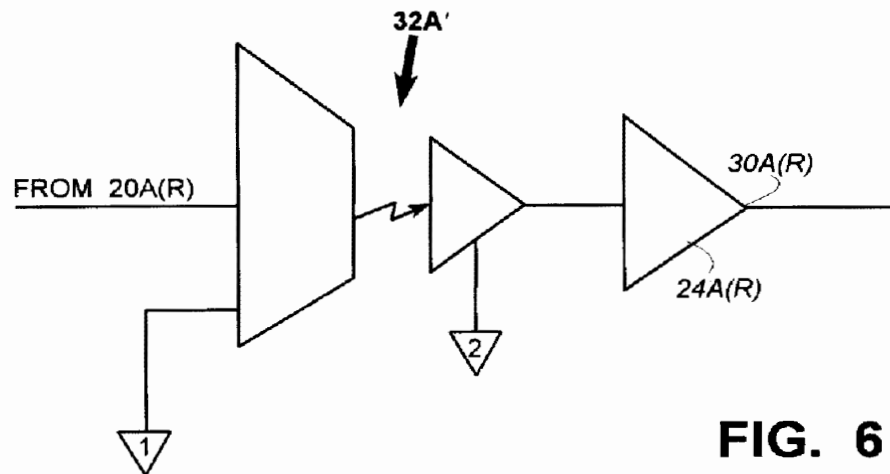
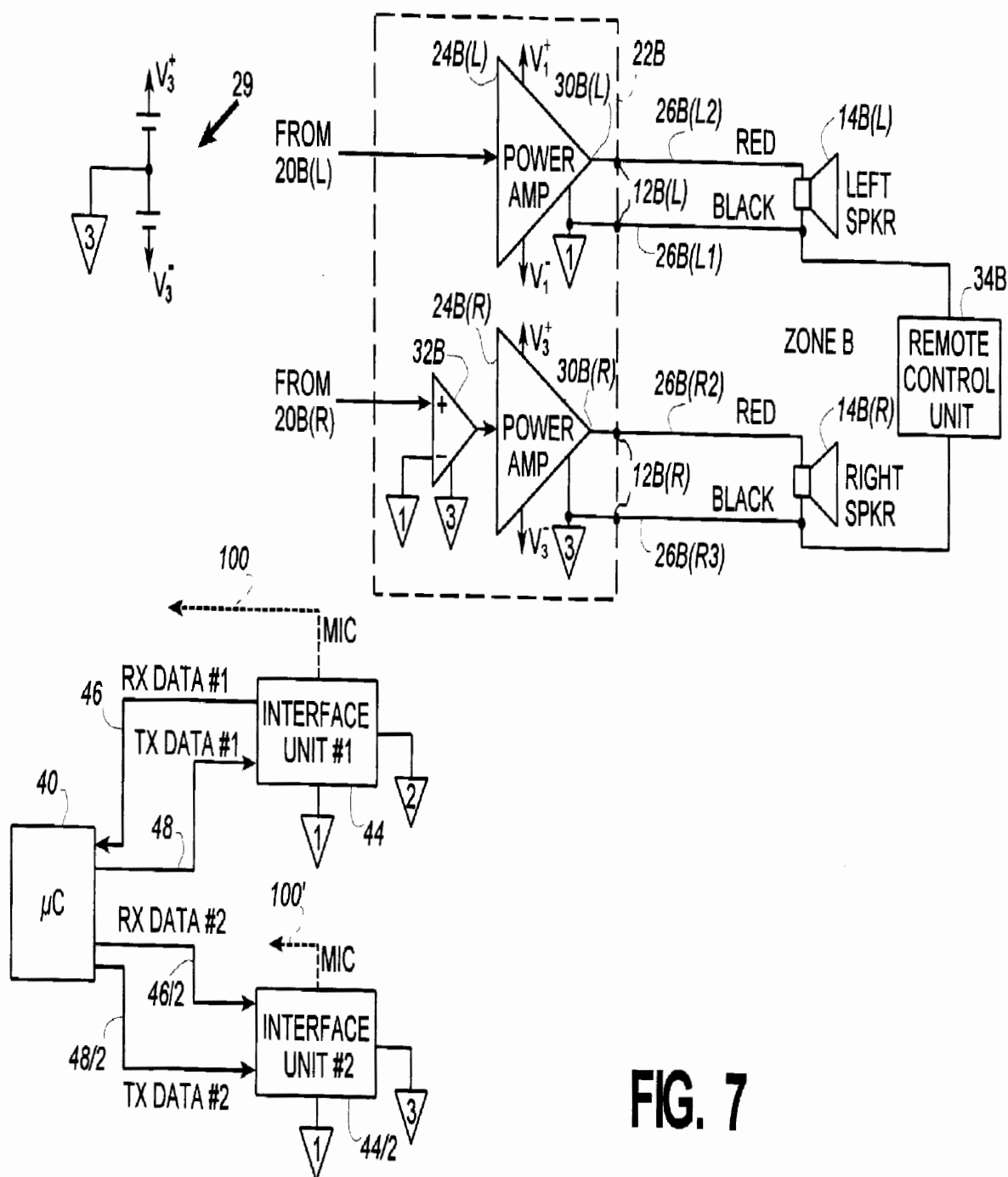


FIG. 6

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AGENT FOR APPLICANT



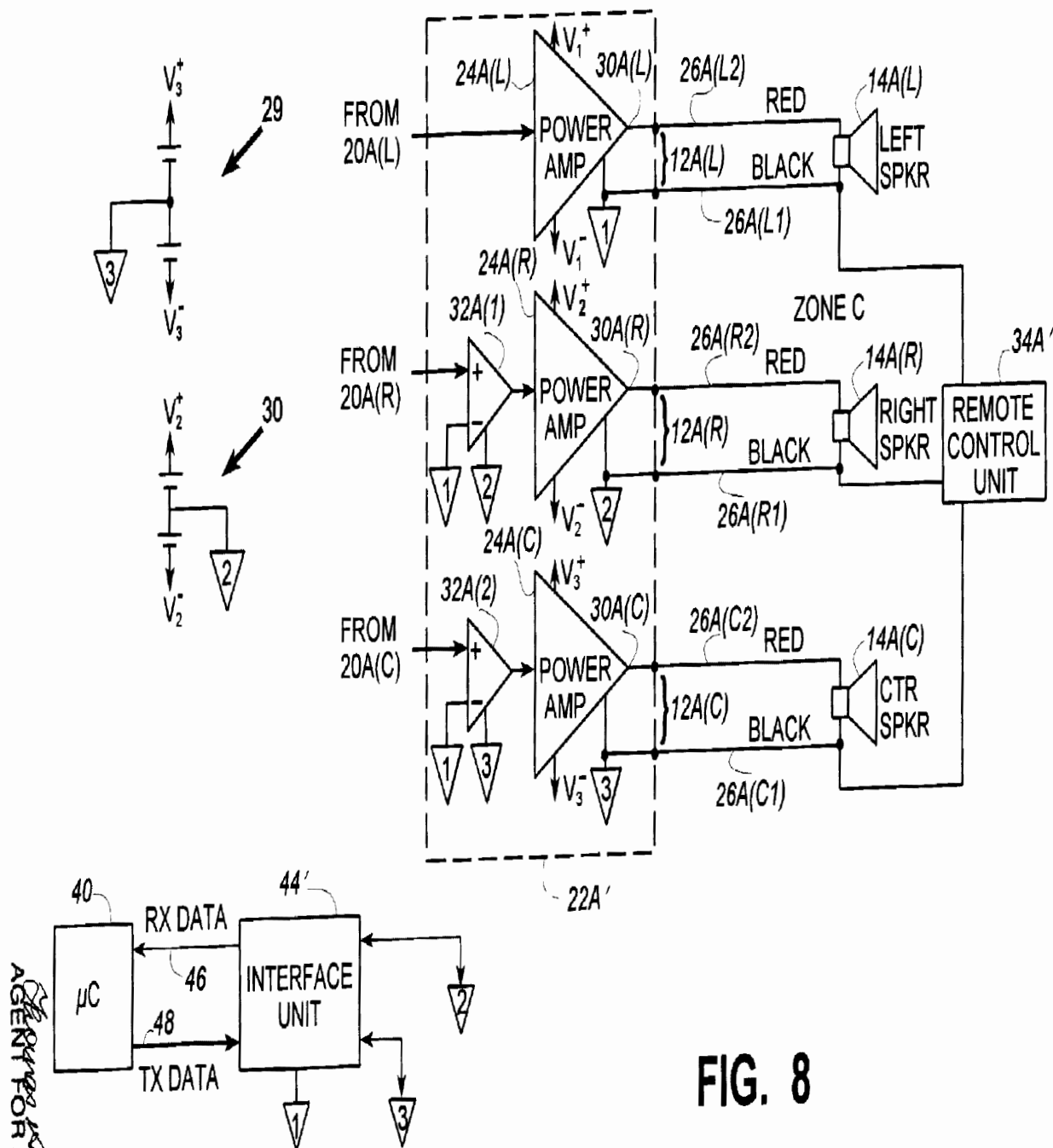
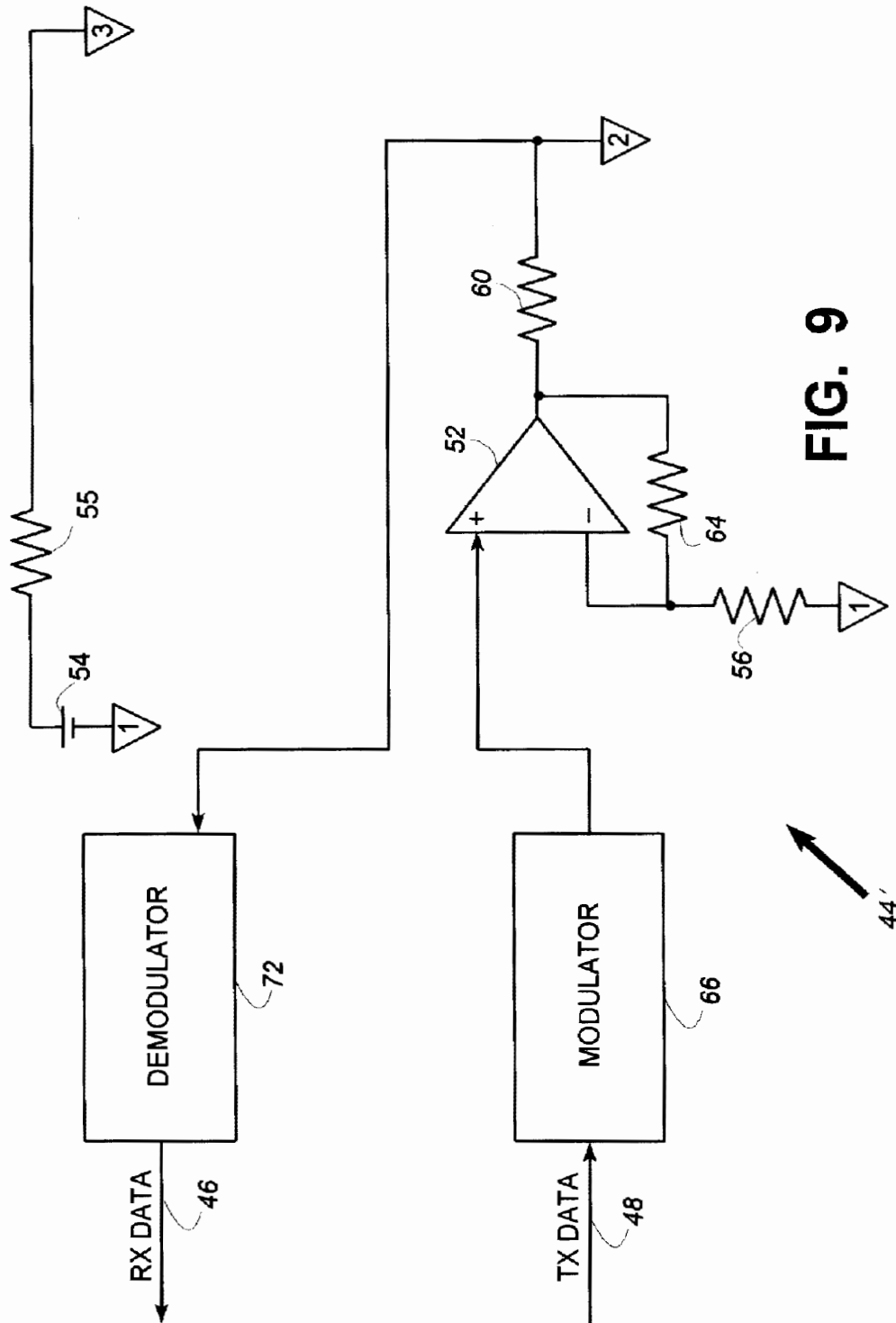
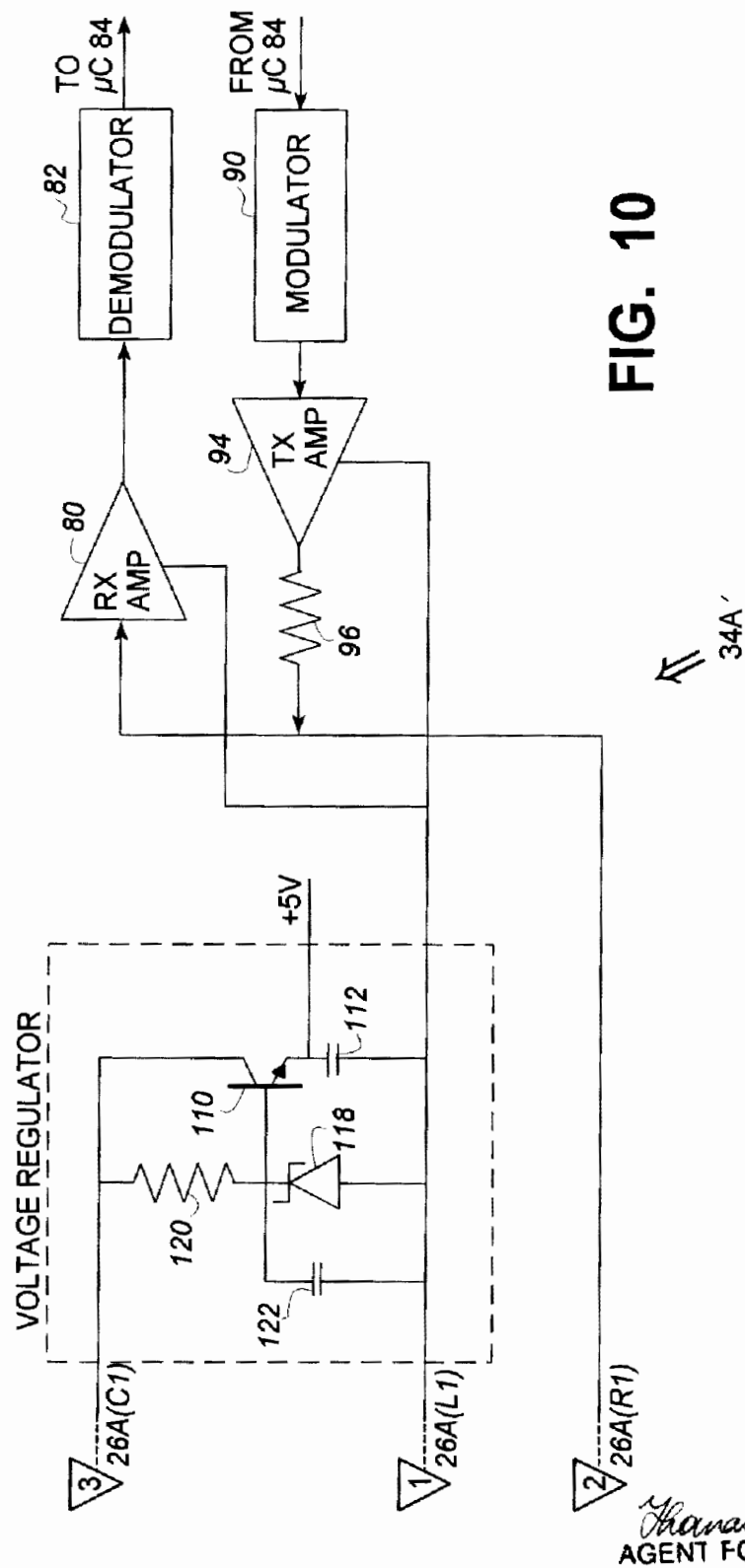


FIG. 8



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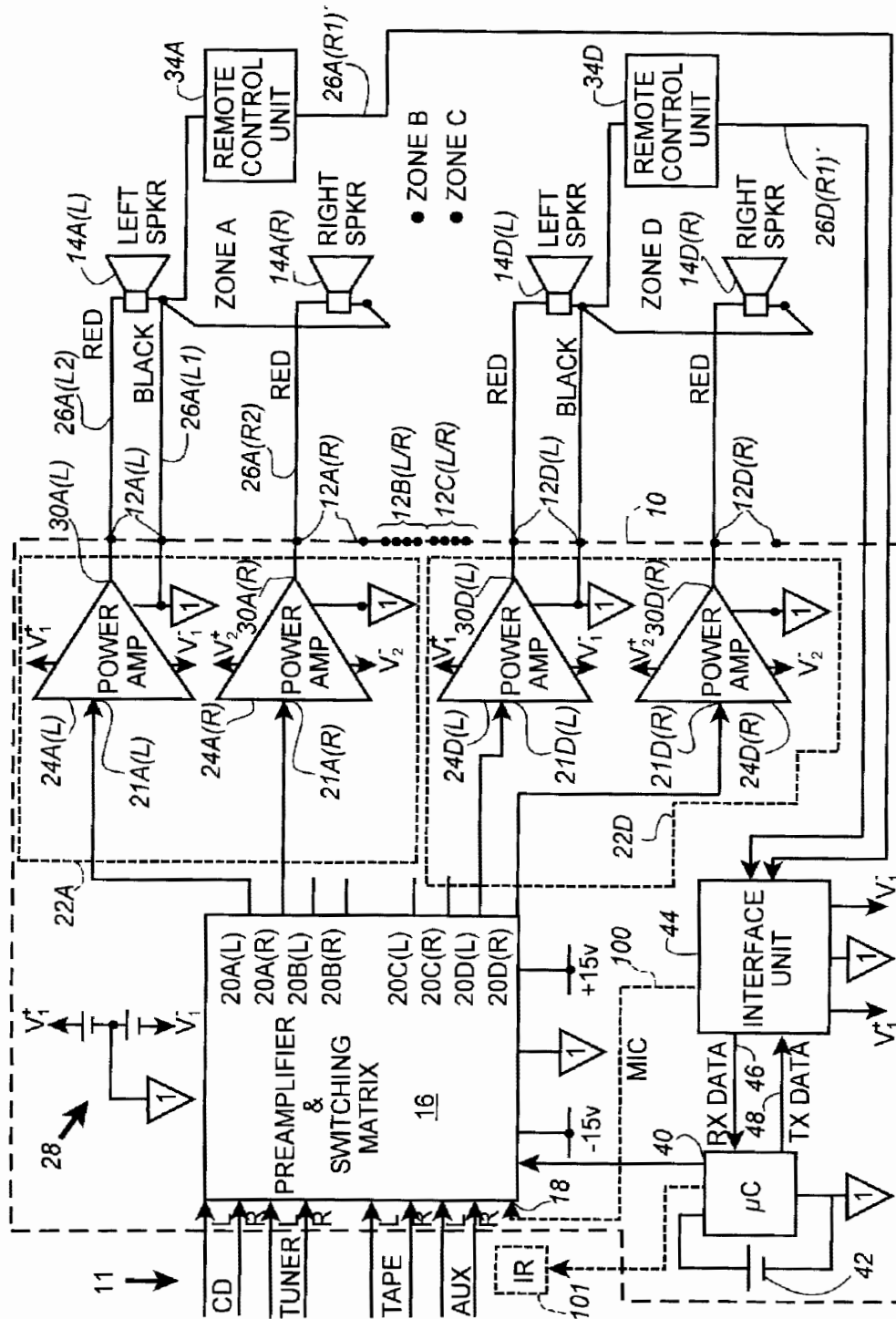


FIG. 11

Thomas Adams & Assoc.
AGENT FOR APPLICANT



Espacenet

Bibliographic data: CN1598767 (A) — 2005-03-23**Method and system for synchronous playing audio-video at BREW platform**

Inventor(s): HUANG XIAOPENG [CN]; DING LI [CN]; XIAO XUEWEN [CN] ±
(XIAOPENG HUANG, ; LI DING, ; XUEWEN XIAO, ;
XIAOPENG,DING HUANG)

Applicant(s): XIAO XUEWEN [CN] ± (XIAO XUEWEN)

Classification: - **international:** *G06F17/00; G06F9/44; H04N7/24;*
(IPC1-7): G06F17/00; G06F9/44; H04N7/24
- **cooperative:**

Application number: CN2004109566 20040915

Priority number(s): CN2004109566 20040915

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Abstract of CN1598767 (A)

The invention relates to an approach of synchronous playing audio and video in BREW platform. It includes: setting a data receiving module to connect with network; setting triggering incident of data receiving to receive data from network and storage into data zone; setting an audio playing module to check data in audio data zone, playing audio data and update time stamp of audio playing by using periodic incident to trigger in process of playing audio data; setting a video playing module to check data in video data zone, decode video data, convert video data format to local format, adjust video and audio synchronization, display video data, and what's more, to adjust interval among video frames to realize synchronistically play audio and video data and guarantee its fluency and timely responsibility.

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[71] 申请人 萧学文

地址 100083 北京市海淀区泉宗路 2 号万柳
光大家园 9 号楼 0938

[72] 发明人 黄小鹏 丁 力 萧学文

[74] 专利代理机构 北京律诚同业知识产权代理有限公司

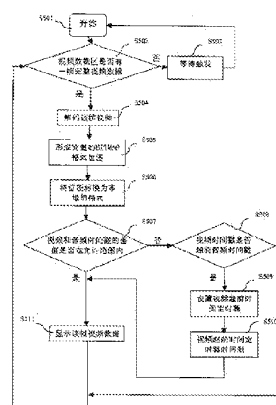
代理人 梁 挥 祁建国

权利要求书 3 页 说明书 9 页 附图 4 页

[54] 发明名称 在 BREW 平台上同步播放音频视频的方法及系统

[57] 摘要

本发明涉及一种在 BREW 平台上实现同步播放音频视频的方法，包括：设置一数据接收模块，用于与网络建立连接，设置数据接收触发事件，接收来自网络的数据，并存储到数据区；设置一音频播放模块，用于检查音频数据区的数据，播放音频数据，并且在播放音频数据的过程中，利用周期性事件触发定时更新音频播放的时间戳；设置一视频播放模块，用于检查视频数据区的数据、解码视频数据、将视频数据的格式转换为本地格式、调节视频和音频的同步和显示视频数据，还可以进一步调节视频帧间隔，从而实现在 BREW 环境中实现同步播放音频视频数据，并保证其流畅性和及时响应性。



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1. 一种在 BREW 平台上实现同步播放音频视频的方法，其特征在于，所述方法包括：
- 5 设置一数据接收模块，用于与网络建立连接，设置数据接收触发事件，接收来自网络的数据，并存储到数据区；
- 设置一音频播放模块，用于检查音频数据区的数据，播放音频数据，并且在播放音频数据的过程中，利用周期性事件触发定时更新音频播放的时间戳；
- 设置一视频播放模块，用于检查视频数据区的数据、解码视频数据、将视频数据
- 10 的格式转换为本地格式、调节视频和音频的同步、显示视频数据。
2. 根据权利要求 1 所述的在 BREW 平台上实现同步播放音频视频的方法，其特征在于，所述设置一音频播放模块中，当检查音频区，发现音频数据不够播放时，系统转到等待触发。
3. 根据权利要求 2 所述的在 BREW 平台上实现同步播放音频视频的方法，其特征
- 15 在于，所述等待触发的方式是事件触发或定时器触发。
4. 根据权利要求 3 所述的在 BREW 平台上实现同步播放音频视频的方法，其特征在于，所述事件触发中的事件是数据流接收模块中的数据接收触发事件。
5. 根据权利要求 3 所述的在 BREW 平台上实现同步播放音频视频的方法，其特征在于，所述定时器触发的步骤，包括：
- 20 设置音频等待定时器；
- 音频等待定时器时间到。
6. 根据权利要求 1 所述的方法，其特征在于，所述音频播放的步骤包括：
- 步骤 61、检查音频数据是否足够播放；音频数据足够时，就转到步骤 63；
- 步骤 62、设置等待触发的条件，在触发条件满足时，返回步骤 61；
- 25 步骤 63、播放音频数据，并且在播放音频数据时，周期性事件触发更新音频播放时间戳。
- 步骤 64、音频播放完成返回步骤 61。
7. 根据权利要求 6 所述的方法，其特征在于，所述的音频播放步骤还包括，在音频播放部分的开始设置音频播放时间的周期性事件触发，在播放音频数据的过程中，
- 30 周期性事件触发定时更新音频播放的时间戳。

8. 根据权利要求1所述的方法, 其特征在于, 所述视频播放步骤包括:
- 步骤801、检查是否接收到一帧完整视频数据; 接收到一帧完整视频数据时, 转到步骤803;
- 步骤802、设置等待触发的条件, 条件满足时, 转到步骤801;
- 5 步骤803、解码接收到的视频数据;
- 步骤804、形成完整的BITMAP格式位图;
- 步骤805、将位图转换成为手机本地的格式;
- 步骤806、比较视频时间戳和音频时间戳的差值, 如果差值在允许范围内, 转到步骤810;
- 10 步骤807、比较视频时间戳是否超前音频时间戳, 如果是, 转到步骤809;
- 步骤808、返回步骤801;
- 步骤809、设置视频超前定时器, 定时器时间到转到步骤806;
- 步骤810、显示该帧数据, 返回步骤801。
9. 根据权利要求8所述的方法, 其特征在于, 在步骤806 视频时间戳和音频时间戳的差值在范围内, 还可以进一步调节视频数据的帧间隔。
- 15 10. 根据权利要求9所述的方法, 其特征在于, 所述调节视频数据的帧间隔的步骤包括:
- 步骤101、比较视频准确播放时间和当前时间, 视频准确播放时间在后, 执行步骤102, 视频准确播放时间在前, 跳转到步骤104;
- 20 步骤102、设置差值定时器;
- 步骤103、差值定时器时间到;
- 步骤104、帧间隔调节完毕。
11. 根据权利要求1所述的方法, 其特征在于, 所述调节视频和音频的同步包括有:
- 25 视频时间戳和音频时间戳的差值在允许的范围之内, 调节完成;
- 视频时间戳和音频时间戳的差值在允许的范围之内, 调节视频数据的帧间隔, 调节完成;
- 视频时间戳和音频时间戳的差值不在允许的范围之内, 并且视频时间戳超前音频时间戳, 设置视频超前定时器, 定时器时间到, 调节完成;
- 30 视频时间戳和音频时间戳的差值不在允许的范围之内, 并且视频时间戳落后音频

时间戳，调节完成；

12. 根据权利要求8所述的方法，其特征在于，在所述步骤803、所述步骤804、步骤805、步骤806之后都可以增加短时定时器，在短时定时器时间到后再执行下一步骤。

- 5 13. 根据权利要求8所述的方法，其特征在于，步骤802所述等待触发的方式是事件触发或定时器触发。

14. 一种在 BREW 平台上实现同步播放音频视频的系统，其特征在于，所述系统包括数据接收模块、音频播放模块和视频播放模块；其中：

- 10 数据接收模块，用于实现与网络建立连接，设置数据接收触发事件，接收来自网络的数据，并存储到数据区；

 音频播放模块，用于实现检查音频数据区的数据，播放音频数据，在播放音频数据的过程中，利用周期性事件触发定时更新音频播放的时间戳；

 视频播放模块，用于实现检查视频数据区的数据、解码视频数据、转换视频数据的格式、调节视频和音频的同步和显示视频数据。

- 15 15. 根据权利要求14所述的在 BREW 平台上实现同步播放音频视频的系统，其特征在于，所述的触发条件包括事件触发或者定时器触发。

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在 BREW 平台上同步播放音频视频的方法及系统

5 技术领域

本发明涉及多媒体通讯技术领域，尤其涉及在 BREW 平台手机上实现同步播放音频视频的方法。

背景技术

10 BREW 是 Binary Runtime Environment for Wireless 的缩写，是为无线数据应用程序开发和执行提供的通用平台。

BREW 运行时环境是一个较薄的软件接口层，位于芯片系统软件 and 应用程序间的无线设备上，使最终用户可以无线下载程序并在启用 BREW 的设备上运行。BREW 具有较高的闪存和 RAM 使用效率（即，它所占闪存较少），而且在应用程序运行时动态分配 RAM。BREW 提供一个高效、低成本、可扩展和熟悉的应用程序执行环境（AEE），一套高度集成的芯片集功能、系统软件以及用于实现高级因特网功能和多媒体功能的插件，着重开发可无缝植入任何实际手持设备的应用程序。BREW 提供的功能环境就如同 PC 机上的操作系统一样，可以通过服务提供商下载指定类型的应用程序或游戏来使用。同时，通过 BREW 接口功能，供应商可以提供成套完整的资讯、商务、娱乐功能。BREW 主要应用在移动通信领域，BREW 类似一个开放免费的 PC 操作系统，厂商可以在这个平台上设计各项应用。

25 由于 BREW 环境不支持多线程技术和本身平台的特殊性，在 windows 或 Linux 环境下实现多媒体音频视频数据同步播放的方式无法在 BREW 环境中实现，为了实现多媒体的同步播放，必须使用事件技术。

发明内容

本发明所要解决的技术问题在于提供一种在 BREW 平台上同步播放音频视频的方法及系统，实现在 BREW 环境中实现同步播放音频视频数据，并保证其流畅性和及时响应性。

30 为了实现上述目的，本发明提供了一种在 BREW 平台上实现同步播放音频视频的

方法，包括：

设置一数据接收模块，用于与网络建立连接，设置数据接收触发事件，接收来自网络的数据，并存储到数据区；

5 设置一音频播放模块，用于检查音频数据区的数据，播放音频数据，并且在播放音频数据的过程中，利用周期性事件触发定时更新音频播放的时间戳；

设置一视频播放模块，用于检查视频数据区的数据、解码视频数据、将视频数据的格式转换为本地格式、调节视频和音频的同步、显示视频数据。

所述设置一音频播放模块中，当检查音频区，发现音频数据不够播放时，系统转到等待触发。

10 所述等待触发的方式是事件触发或定时器触发。

所述事件触发中的事件是数据流接收模块中的数据接收触发事件。

所述定时器触发的步骤，包括：

设置音频等待定时器；

音频等待定时器时间到。

15 所述音频播放的步骤包括：

步骤 61、检查音频数据是否足够播放；音频数据足够时，就转到步骤 63；

步骤 61、设置等待触发的条件，在触发条件满足时，返回步骤 61；

步骤 63、播放音频数据，并且在播放音频数据时，周期性事件触发更新音频播放时间戳。

20 步骤 64、音频播放完成返回步骤 61。

所述的音频播放步骤还包括，在音频播放部分的开始设置音频播放时间的周期性事件触发，在播放音频数据的过程中，周期性事件触发定时更新音频播放的时间戳。

所述视频播放步骤包括：

25 步骤 801、检查是否接收到一帧完整视频数据；接收到一帧完整视频数据时，转到步骤 803；

步骤 802、设置等待触发的条件，条件满足时，转到步骤 801；

步骤 803、解码接收到的视频数据；

步骤 804、形成完整的 BITMAP 格式位图；

步骤 805、将位图转换成为手机本地的格式；

30 步骤 806、比较视频时间戳和音频时间戳的差值，如果差值在允许范围内，转到

步骤 810;

步骤807、比较视频时间戳是否超前音频时间戳, 如果是, 转到步骤809 ;

步骤 808、返回步骤 801;

步骤 809、设置视频超前定时器, 定时器时间到转到步骤 806;

5 步骤 810、显示该帧数据, 返回步骤 801。

在步骤 806 视频时间戳和音频时间戳的差值在范围内, 还可以进一步调节视频数据的帧间隔。

所述调节视频数据的帧间隔的步骤包括:

步骤 101、比较视频准确播放时间和当前时间, 视频准确播放时间在后, 执行步

10 骤 102, 视频准确播放时间在前, 跳转到步骤 104;

步骤 102、设置差值定时器;

步骤 103、差值定时器时间到;

步骤 104、帧间隔调节完毕。

所述调节视频和音频的同步包括有:

15 视频时间戳和音频时间戳的差值在允许的范围之内, 调节完成;

视频时间戳和音频时间戳的差值在允许的范围之内, 调节视频数据的帧间隔, 调节完成;

视频时间戳和音频时间戳的差值不在允许的范围之内, 并且视频时间戳超前音频时间戳, 设置视频超前定时器, 定时器时间到, 调节完成;

20 视频时间戳和音频时间戳的差值不在允许的范围之内, 并且视频时间戳落后音频时间戳, 调节完成;

在所述步骤 803、所述步骤 804、步骤 805、步骤 806 之后都可以增加短时定时器, 在短时定时器时间到后再执行下一步骤。

步骤 802 所述等待触发的方式是事件触发或定时器触发。

25 本发明还公开了一种在 BREW 平台上实现同步播放音频视频的系统, 所述系统包括数据接收模块、音频播放模块和视频播放模块; 其中:

数据接收模块, 用于实现与网络建立连接, 设置数据接收触发事件, 接收来自网络的数据, 并存储到数据区;

30 音频播放模块, 用于实现检查音频数据区的数据, 播放音频数据, 在播放音频数据的过程中, 利用周期性事件触发定时更新音频播放的时间戳;

视频播放模块,用于实现检查视频数据区的数据、解码视频数据、转换视频数据的格式、调节视频和音频的同步和显示视频数据。

所述的触发条件包括事件触发或者定时器触发。

本发明实现在 BREW 环境中实现同步播放音频视频数据,并保证其流畅性和及时
5 响应性。

附图说明

图 1 所示为设置网络数据接收触发事件的流程图;

图 2 所示为数据流接收的流程图;

10 图 3 所示为音频数据播放的流程图;

图 4 所示为音频定时器触发的流程图;

图 5 所示为视频数据播放的流程图;

图 6 所示为视频定时器触发的流程图;

图 7 所示为调节视频显示帧间隔的流程图。

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具体实施方式

以下结合附图和具体实施例对本发明进行详细描述,但不作为对本发明的限定。

由于 BREW 平台不支持多线程技术,在 BREW 平台要实现音视频的同步性要使用基于事件和时间触发的技术,实现如下三大模块的功能:数据流接收模块、音频播放模
20 块和视频播放模块。

数据流接收模块,主要负责接收来自网络的数据。其功能包括:设置网络数据接收触发事件;和在有数据到达后,系统产生数据接收触发事件,随后开始接收数据。

图 1 为数据流接收模块设置网络数据接收触发事件的流程图。

在步骤 S101 中,通过 BREW 平台与服务器相连接。

25 在步骤 S102 中,设置数据接收触发事件。这可以通过调用函数 `ISOCKET_Readable` 来设置。

图 2 为在有数据到达 BREW 平台后,接收数据流的流程图。

在步骤 S201 中,有数据到达 BREW 平台。

在步骤 S202 中,系统产生数据接收触发事件。

30 在步骤 S203 中,开始从网络接收流媒体数据。

在步骤 S204 中，将接收的数据流中的音频数据放到音频数据区中，视频数据放到视频数据区中。

步骤 S205 为音频播放和视频播放的开始点。

当有新的数据到达时，BREW 平台会自动产生新的数据接收触发事件，重复执行如图 2 所示的数据接收步骤。

音频播放模块，负责播放音频数据。

图 3 为音频数据播放的流程图。

在图 3 所示的流程图中步骤 S301，即音频数据播放流程的开始点，可以是图 2 所示数据接收流程图步骤 S205 的位置 A。

在步骤 S302 中，检查音频数据区的数据是否够播放。是否足够播放的条件是判断音频数据区的音频数据有多少，或音频数据区的数据能够播放多长时间。这个值用户可以自行设定，如够播放 3 秒钟。如果音频数据区的音频数据不够播放，那么跳转到步骤 S303；如果数据够播放跳转到步骤 S304。

在步骤 S303 中，系统等待触发事件的到达。当触发条件满足后，跳转到步骤 S302，继续检查音频数据区的数据是否够播放。触发的条件可以是事件触发，如数据流接收模块中的数据接收触发事件，也可以是定时器触发。图 4 所示为音频定时器触发的流程图。

在步骤 S304 中，更新音频时间戳。

在步骤 S306 中，播放音频数据。

在步骤 S307 中，播放完音频数据后，跳转到步骤 S302 继续检查音频数据是否够播放。

在图 3 音频播放流程图的开始位置，可以设置音频播放时间的周期性事件触发，在播放音频数据的过程中，周期性触发事件在触发条件满足后 (S305)，更新音频播放的时间戳 (S304)。这个时间戳被视频播放模块用作播放时间的参考。例如可以设置音频播放中每播放 500 毫秒的音频数据就触发一个事件。如果没有设置音频播放时间的周期性事件触发，则系统缺省每播放 1000 毫秒的音频数据就触发一个事件。在音频播放的过程中，音频播放的时间戳由于周期性的事件触发被定期的更新。例如，当前时间戳为 1000 毫秒，过 500 毫秒的音频数据就触发一个事件，事件触发后时间戳更新为 1500 毫秒。

图 4 为音频定时器触发的流程图。在图 4 中，设置音频等待定时器 (S401)，在

音频等待定时器时间到(S402)后,再去检查音频数据区中是否有足够的数据(S302)。

视频播放模块,负责播放视频数据。

图5为视频数据播放的流程图。

在图5所示的流程图中步骤S501,即视频数据播放流程的开始点,可以是图2所示数据接收流程图步骤S205的位置A。

在步骤S502,检查视频数据区的数据是否够一帧完整的视频数据,如果视频区的数据不够一帧完整的视频数据,跳转到步骤S503等待触发;如果视频区的数据够一帧完整的视频数据,执行步骤S504。

在步骤S503,系统等待触发事件的到达。当触发条件满足后,转到S502继续检查视频数据区的数据是否够一帧完整的视频数据。在步骤S503中触发的条件可以是事件触发,如数据流接收模块中的数据接收触发事件,也可以是定时器触发。如图6所示为视频定时器触发的流程图。

在步骤S504中,视频数据区的数据够一帧完整的视频数据,解码这一帧视频数据。

在步骤S505中,将解码出来的数据形成一幅完整的BITMAP格式位图。

在步骤S506中,将步骤S505中形成的BITMAP格式位图转换为手机本地的格式。

在步骤S507中,计算视频时间戳和音频时间戳的差值,判断音频当前播放的时间戳和该视频帧的时间戳的差值(也称上述差值为同步误差)是否在允许的范围内。例如视频时间戳和音频时间戳差值的允许范围可以设置为正负100毫秒之间。

当视频时间戳和音频时间戳的差值不在允许的范围内时,转到步骤S508,判断视频时间戳是否超前音频时间戳。视频时间戳超前音频时间戳时,根据视频时间戳和音频时间戳的差值设置视频超前定时器(S509),视频超前定时器时间到时(S510),从而保证同步误差在允许的范围内;视频时间戳落后音频时间戳时,显示该帧视频数据已经没有意义,丢弃该视频数据,然后转到S502继续检查视频数据区是否有一帧完整的视频数据。

当视频时间戳和音频时间戳的差值在允许的范围内时,显示该帧视频数据(S511)然后转到步骤S502继续检查视频区是否有一帧完整的视频数据。

在如图6中,首先设置视频等待定时器(S601),在视频等待定时器时间到(S602)后,检查视频数据区中是否有一帧完整的视频数据(S502)。

在步骤S504、步骤S505、步骤S506、步骤S507之后可以增加短时定时器,在短

时定时器时间到后再执行下面的步骤。例如在步骤 S504, 即解码该帧视频后, 设置 10 毫秒的短时定时器, 在 10 毫秒时间到后再执行步骤 S505 形成完整的 BITMAP 格式位图。这样可以使手机在播放视频音频流的时候依然可以及时相应用户的按键请求等。

- 5 为了使视频播放更加流畅, 不仅要求音频数据和视频数据的同步误差在允许的范围内, 还要求任何两帧视频之间的时间间隔基本一致。为了达到上述目的, 当同步误差在允许范围内时, 还可以进一步调节帧间隔显示该帧视频数据。

在音频视频同步的情况下, 视频准确显示的时间等于上一帧显示的时间加上帧间隔时间。帧间隔时间等于帧率的倒数。例如每秒显示 5 帧视频数据, 帧间隔时间是 200
10 毫秒。在实际播放中, 很能难保证两帧显示之间的时间差绝对准确地等于帧间隔时间, 因此, 可以设定视频帧间隔时间误差 (称为帧间隔误差) 允许范围。当两帧之间的帧间隔误差在允许的范围内时, 认为两帧视频之间的时间间隔基本一致, 即两帧视频的播放是流畅的。例如设置帧间隔误差允许范围为 $\pm 10\%$, 即当两帧显示的时间差在 200 毫秒时, 允许的误差范围为正负 20 毫秒。

- 15 图 7 为调节视频显示帧间隔的流程图。

在图 7 中, 首先视频时间戳和音频时间戳的差值在允许范围内 (S701); 判断当前时间是否超前于视频准确播放时间 (S702), 当前时间落后于视频准确播放时间时, 直接显示此帧视频数据 (S705); 当前时间超前视频准确播放时间时, 设置差值定时器 (S703), 差值定时器时间到 (S704), 显示此帧视频数据 (S705)。显示此帧视
20 频数据 (S705) 后, 记录播放时间戳 (S706)。在图 7 中, 步骤 S702、S703、S704、S705、S706 共同构成帧间隔调节模块 B。

在步骤 S703 中设置的差值定时器的差值满足如下两个条件:

条件一、同步误差在允许范围内; 和

条件二、帧间隔误差在允许范围内。

- 25 条件一可以保证视频数据和音频数据同步; 条件二可以保证视频数据帧间隔基本一致, 从而得到流畅的视频效果。

在视频播放的过程中, 若选择调节帧间隔, 那么可以用图 7 中的帧间隔调节模块 B 取代图 5 中的步骤 S511, 并且当视频时间戳落后音频时间戳时, 转到步骤 S706 之前, 记录视频播放时间戳, 从而在 BREW 环境中实现同步播放音频视频数据。

- 30 下面是在以 BREW 平台为基础的手机上, 应用本发明的方法播放音频和视频的另

一个实施例。

首先手机与服务器相连接，然后调用系统函数，设置网络数据接收触发事件。

当来自服务器的数据流到达手机的 BREW 终端后，系统会产生数据接收触发事件，随后开始接收数据。系统分配两个数据区，一个为音频数据区，一个为视频数据区。

- 5 对到达 BREW 终端的数据流进行分析，将视频的数据放到视频数据区，音频数据放到音频数据区。

例如设置足够播放的条件为音频数据区的数据能连续播放 5 秒钟。检查音频数据区的数据是否足够播放。检查结果发现音频数据区的数据仅仅能播放 3 秒钟，设置音频等待定时器为 2 秒钟。在 2 秒钟音频定时器时间到后，再次检查音频数据区的数据
10 是否足够。检查结果音频数据区的数据能够播放 5 秒。

播放音频数据区的数据。播放完成，则继续检查音频数据区是否有足够的音频数据。

- 在音频播放的开始设置音频播放时间的周期性事件触发，例如每播放 500 毫秒的音频数据就触发一个事件。在播放音频数据的过程中，音频播放的时间戳由于周期性的事件触发被定期的更新，如现在的时间戳为 1000，经过 500 毫秒的音频数据就触发
15 一个事件，触发后，时间戳更新为 1500，再过 500 毫秒，时间戳更新为 2000。

视频播放时，先检查视频数据区的数据是否够一帧完整的视频数据，如果不够，等待触发。触发条件在此选用定时器触发。例如设置视频等待定时器 1 秒钟。在 1 秒钟后，再去检查数据是否足够。

- 20 视频数据区的数据够一帧完整的视频数据后，就调用解码器解码这一帧视频。解码出来的数据通过色彩空间转换变为 RGB24 位的色彩空间，在内存区组织形成一幅完整的 BITMAP 格式位图内容，再将该位图转换为手机本地的格式。

- 随后要比较同步误差是否在允许的范围之内。例如同步误差的允许范围设置为正负 100 毫秒内。若音频的时间戳是 2000，视频的时间戳是 2050，那么就可以直接显示
25 该帧视频数据。

如果同步误差不在允许的范围之内，就要判断视频时间戳是否超前音频时间戳。当视频时间戳超前音频时间戳时，例如，音频的时间戳是 2000，视频的时间戳是 2150，即视频超前 $(2150 - 2000) = 150$ 毫秒，此时设置 150 毫秒的定时器，在该定时器时间到后，再显示该帧视频。

- 30 当视频时间戳落后音频时间戳时，例如，音频的时间戳是 2000，视频的时间戳是

1850, 在这种情况下, 显示该帧视频数据已经没有意义, 因此丢弃该帧数据。

为了使视频播放更加地流畅, 需要进一步调节帧间隔。

例如, 在BREW终端上设置每秒5帧视频图像, 帧间隔时间是200毫秒, 设定帧间隔误差范围 $\pm 10\%$, 即允许两帧显示的时间差在200毫秒正负20毫秒之间。

- 5 当第N帧视频播放的时间是2050, 则第(N+1)帧视频的准确播放时间应该是2250。
若当前视频的时间戳是2230, 那么视频帧超前20毫秒, 帧间隔误差在误差允许的范围(± 20 毫秒)之内, 因此可以直接显示当前帧的视频数据。或当前视频的时间戳是2220, 那么视频帧超前30毫秒, 帧间隔误差超过了误差允许的范围, 则需要设置差值定时器的值为30毫秒, 以调节帧间隔, 从而保证播放的视频图像流畅。在差值定
10 时器时间到后, 显示该帧视频。若当前视频的时间戳是2260, 表示已经落后于准确显示时间, 则直接显示该帧视频。显示完该帧视频数据后, 记录视频时间戳的值。

- 在解码该帧视频、形成完整的BITMAP格式位图、将位图转换为手机本地的格式和显示该帧视频这四个步骤之后增加10毫秒的定时器, 在10毫秒到后再继续执行下面的步骤。这样可以使手机在播放视频音频流的时候依然可以及时响应用户的按键请求
15 等。

在处理完一帧数据之后, 继续检查视频数据区的情况。

当然, 本发明还可有其他多种实施例, 在不背离本发明精神及其实质的情况下, 熟悉本领域的技术人员当可根据本发明作出各种相应的改变和变形, 但这些相应的改变和变形都应属于本发明所附的权利要求的保护范围。

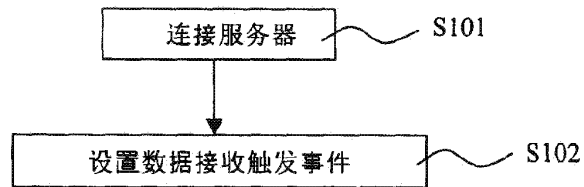


图 1

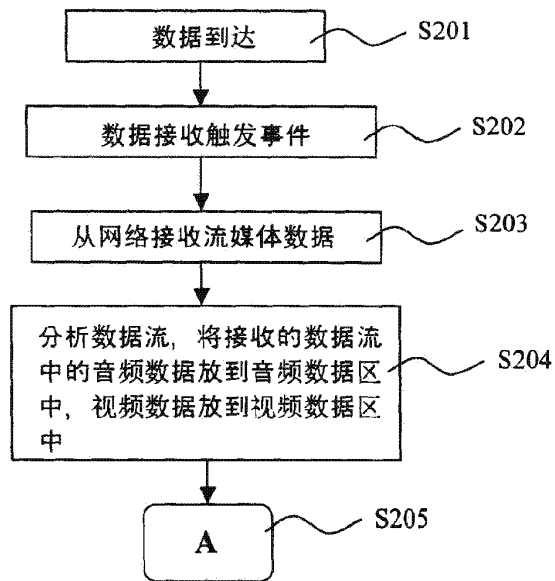


图 2

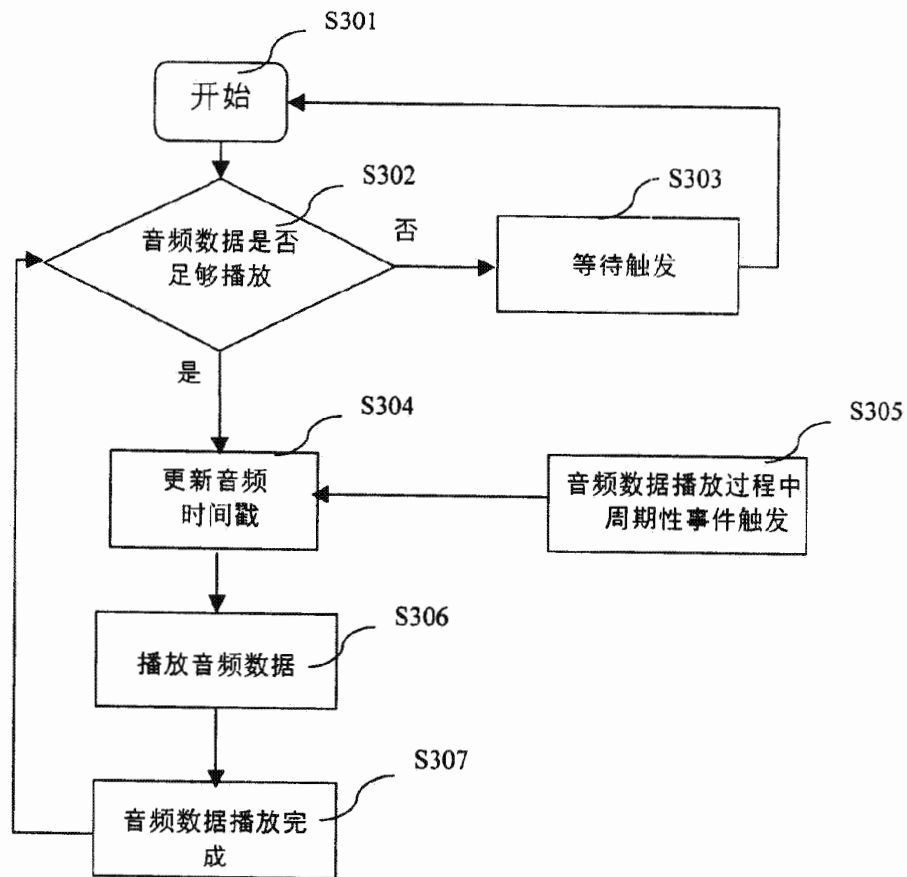


图 3

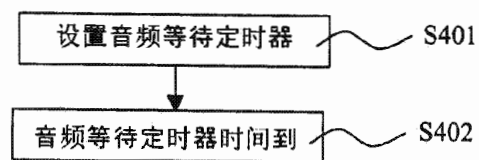


图 4

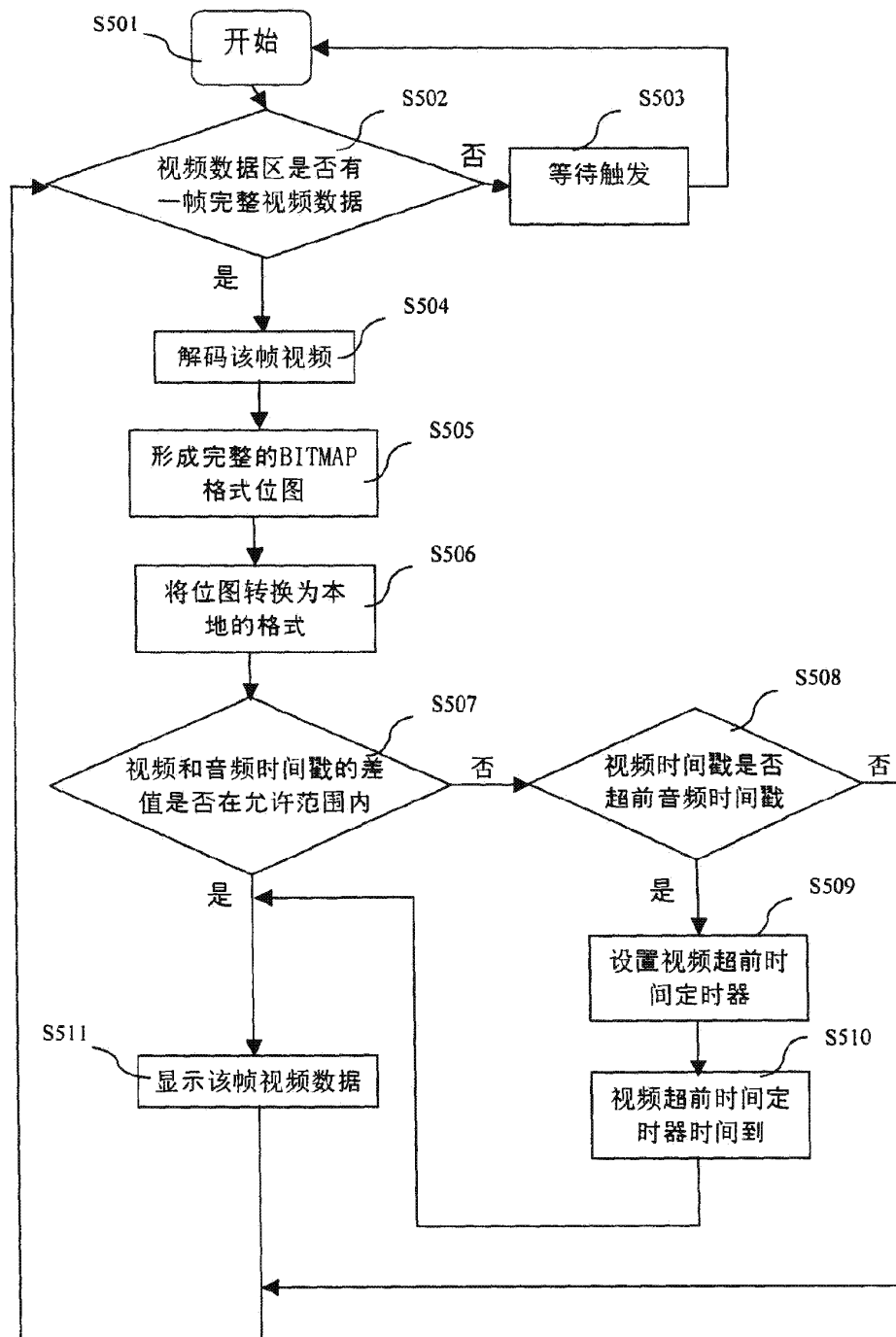


图 5

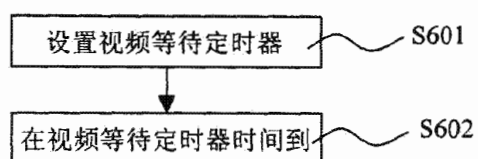


图 6

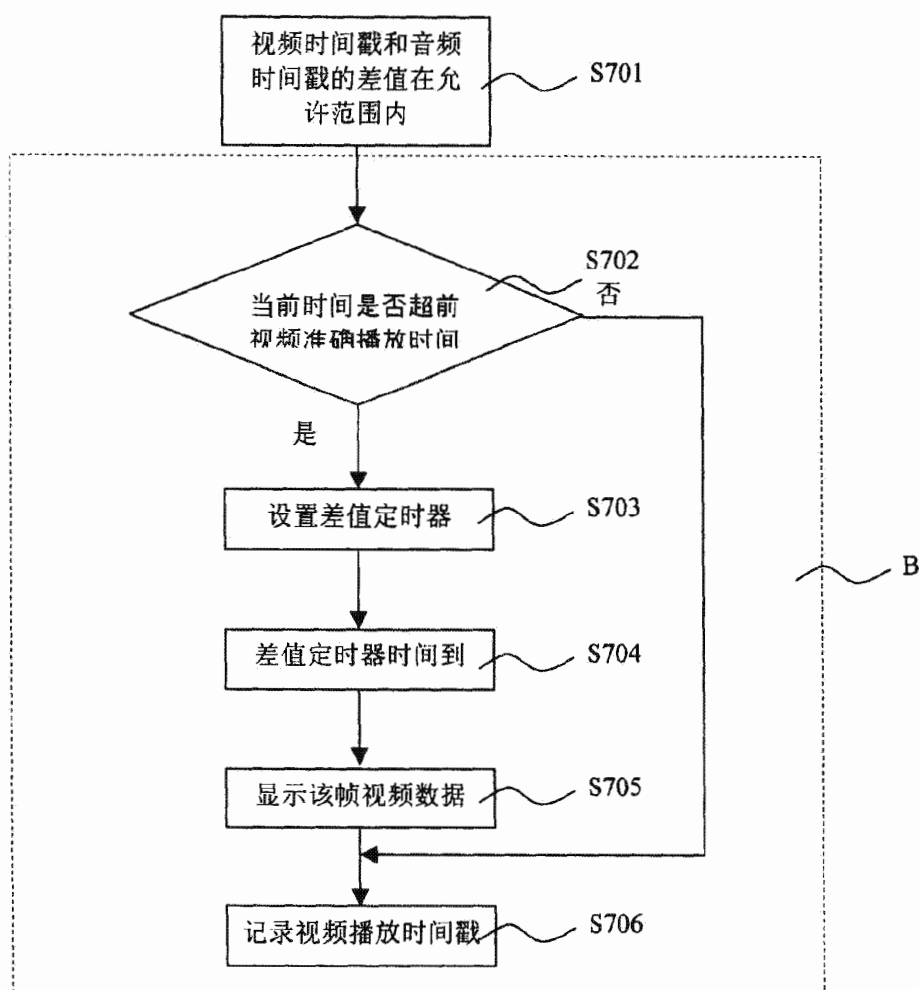


图 7



(11) **EP 0 742 674 B1**

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(54) **Video encoding method and system using a rate-quantizer model**

Methode und System zur Videokodierung mit einem Ratenquantifizierermodell

Méthode et système de codage vidéo employant un modèle débit-quantificateur

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(73) Proprietor: **Kabushiki Kaisha Toshiba**
Kawasaki-shi,
Kanagawa-ken 210-8572 (JP)

(72) Inventors:
• **Azadegan, Faramarz**
Waltham,
Massachusetts 02154 (US)
• **Unno, Hiroaki**
Ichikawa-shi,
Chiba 272 (JP)
• **Mimura, Hideki**
Yokohama 236 (JP)
• **Kitamura, Tetsuya**
Komea-shi,
Tokyo 201 (JP)

(74) Representative: **Ahner, Philippe et al**
BREVALEX
95 rue d'Amsterdam
75378 Paris Cedex 8 (FR)

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- "MPEG DIGITAL VIDEO-CODING STANDARDS - DELIVERING PICTURE-PERFECT COMPRESSION FOR STORAGE, TRANSMISSION, AND MULTIMEDIA APPLICATIONS", IEEE SIGNAL PROCESSING MAGAZINE, IEEE SERVICE CENTER, PISCATAWAY, NJ, US, vol. 14, no. 5, 1 September 1997 (1997-09-01), pages 82-100, XP011089789, ISSN: 1053-5888, DOI: 10.1109/79.618010

DescriptionCROSS-REFERENCES TO RELATED APPLICATIONS

5 **[0001]** This application is related to U.S. Patent Application S/N 08/438,014, entitled "METHOD AND SYSTEM TO REPLACE SECTIONS OF AN ENCODED VIDEO BITSTREAM", filed May 8, 1995 (WO 97 03 443 A). This application is also related to and being concurrently filed with four other patent applications: These applications are U.S. Patent Application S/N 08/XXX,XXX, entitled "AUDIOVISUAL ENCODING SYSTEM WITH A REDUCED NUMBER OF AUDIO ENCODERS" (WO 96 36 173 A), U.S. Patent Application S/N 08/XXX,XXX, entitled "METHOD AND SYSTEM FOR A

10 USER TO MANUALLY ALTER THE QUALITY OF PREVIOUSLY ENCODED VIDEO FRAMES" (WO 96 35 999 A), U.S. Patent Application S/N 08/XXX,XXX, entitled "METHOD AND SYSTEM FOR A USER TO MANUALLY ALTER THE QUALITY OF A PREVIOUSLY ENCODED VIDEO SEQUENCE" (WO 96 36 170 A), and U.S. Patent Application S/N 08/XXX,XXX, entitled "A RATE-CONTROLLED DIGITAL VIDEO EDITING METHOD AND SYSTEM WHICH CONTROLS BIT ALLOCATION OF A VIDEO ENCODER BY VARYING QUANTIZATION LEVELS" WO 96 35 998 A, each

15 filed on June 6, 1995.

BACKGROUND OF THE INVENTIONField of the Invention

20 **[0002]** The present invention relates to video encoding systems, such as MPEG and MPEG-2 video encoders, and more particularly, to video pre-encoders which analyze the bit-rate of the pre-encoded digital video on a frame by frame basis to determine an average bit-rate for each quantization level used in the pre-encoding process. The present invention is used to encode digital video to fit within a fixed capacity storage medium based on bit-rates of future frames as

25 determined in the pre-encoding process.

Discussion of the Background

30 **[0003]** Digital video encoding systems encode raw digital video signals representing the intensity of portions of a frame into compressed, encoded digital video for use in one of two different playback modes; these modes are a constant bit-rate (CBR) mode and a variable bit-rate (VBR) mode. The constant bit-rate mode is suitable for use in transmission technologies such as cable television and direct satellite broadcast which will often use a fixed bandwidth to deliver a signal from a transmitter to a remote receiver. The variable bit-rate mode is suitable for playing back digital video from digital storage media such as compact discs.

35 **[0004]** In CBR mode, the transmitter must send, typically, 30 frames per second to the receiver while avoiding video buffer underflow and overflow. Underflow occurs in constant bit-rate mode when at the end of a 1/30th of a second time interval, there is insufficient data in the video decoding buffer to construct a picture. Overflow in CBR mode occurs when there is too much video data placed in the buffer for the decoder to decode within the 1/30th of a second interval.

40 **[0005]** To control the bit-rate in the CBR mode, video encoders use a technique called adaptive quantization to vary the instantaneous bit-rate to meet the target bit-rate. Adaptive quantization involves estimating a number of bits to be generated by a series of macroblocks based on the quantization levels of the macroblocks, encoding the series of macroblocks, determining the number of bits of the encoded macroblocks, comparing the number of bits resulting from the encoding with the estimated number of bits and changing the quantization levels of a succeeding set of macroblocks based on the accuracy of the previous estimate.

45 **[0006]** In order to accurately estimate the number of bits that will be generated by a series of macroblocks, a quantizer level versus bit-rate model indicates what the average number of bits to be generated by a particular quantization level will be. In a paper entitled "Joint Multi-Channel Bit-Rate Control for Multiplexed MPEG Video Using Rate-Distortion Curves", by Reiningger et al. and presented at Princeton Conference 1994, a constant bit-rate encoder is disclosed which splits the encoding process into two phases. In the first phase, a first frame is pre-encoded with parts of frames being pre-encoded using different quantization levels. This gives a quantization level versus bit-rate curve that describes how the first frame responds to the first phase. The first frame then enters a second phase where the quantization level versus bit-rate curve generated in the first phase is used to regulate the encoding of the first frame into part of the final encoded digital video stream. By using the curve generated in the first phase, the second phase may more accurately decide on the quantization levels to be assigned to each macroblock such that the frame is encoded to avoid underflow and overflow. In parallel with the second phase encoding of the first frame, the Reiningger method begins the first phase on the second frame and gathers statistics. The process is repeated for all the frames in a movie. This technique, however, provides only a single frame look ahead which is limited in its ability to control rate.

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[0007] In the VBR mode, overflow can be avoided by only reading data from the digital storage medium when the

buffer can accept new data. In the VBR mode, underflow is still possible and results from too much rather than too little, data being used to represent video frames. When too much data represents a frame, the video decoder cannot perform all the necessary decoding operations in 1/30th of a second. An error results because a frame needs to be displayed before it has been totally decoded.

[0008] In order to provide encoded digital video representing a full-length movie on a single compact disc, various parts of the raw video may be encoded using different quantization levels to produce various image qualities. To determine the amount of space that a particular section of raw video will consume when encoded, the raw video can be pre-encoded at a particular quantization level and the resulting bit lengths of each frame recorded to aid in a later encoding process. To determine the resultant bit-rates for different quantization levels, this process can be repeated for the whole raw video multiple times, and the results recorded after encoding each frame. However, even though this technique provides the greatest accuracy with which the quantization model can be generated, it is very time consuming. As a typical movie will last between an hour and a half and two hours, even pre-encoding a movie twice using a different quantization level each time would take at least three hours.

[0009] EP 0 597 647 A discloses a method for processing video signals included in video frames to determine characteristics of the video frames, comprising the steps of:

- a) obtaining video signals included in plural video frames;
- b') assigning a quantization level to one of said plural video frames;
- c') pre-encoding said one of said plural video frames using the quantization level to obtain a bit-rate;
- d') determining the bit-rate for the quantization level resulting from said pre-encoding step;
- e) repeating steps b', c' and d' sequentially for plural succeeding video frames; and
- f) encoding a plurality of said plural video frames after determining the bit-rates for each of said plural video frames;
- g) determining a target number of bits for said plurality of said plural video frames;
- h) assigning a first set of quantization levels for each frame in said plurality of said plural video frames using said target number of bits and the plural determined bit-rates; and
- i) encoding one video frame of said plurality of said plural video frames into an encoded frame with a resultant number of bits.

SUMMARY OF THE INVENTION

[0010] Accordingly, one object of the present invention is to overcome the problems of video pre-encoding as outlined above.

[0011] A further object of this invention is to pre-encode raw video with at least two quantization levels such that a bit-rate versus quantization level model can be generated for plural frames of the raw video.

[0012] It is a further object of this invention that two quantization levels be uniformly assigned to the plural frames during the pre-encoding process so as to generate two quantization level versus bit-rate points in a line.

[0013] It is a further object of this invention that the two quantization levels be assigned to different regions of the plural frames in a checkerboard style fashion.

[0014] It is a further object of this invention that four quantization levels be assigned to different regions of the plural frames during the pre-encoding process.

[0015] It is a further object of this invention that four quantization levels be assigned uniformly to the plural frames such that four points on a quantization level versus bit-rate curve can be generated.

[0016] These and other objects are accomplished by a method and system according to the independent claims. The method and system generate and record bit-rates for each quantization level for the plural frames during a pre-encoding process, such that these quantization level versus bit-rate statistics can be used in a later encoding of the raw video. Bit-rates for quantization levels not used in the pre-encoding process can be extrapolated using known quantization levels from the pre-encoding phase. Furthermore, because statistics have been gathered in advance of the encoding process, the encoder can be directed to use greater or fewer bits in a particular section of raw video based on future needs as indicated by quantization level versus bit-rate curves for later frames.

BRIEF DESCRIPTION OF THE DRAWINGS

[0017] A more complete appreciation of the invention and many of the attendant advantages thereof will be readily obtained as the same becomes better understood by reference to the following detailed description when considered in connection with the accompanying drawings, wherein:

- 5
- Figure 1A illustrates a block diagram of the encoder system architecture;
- Figure 1B illustrates devices for obtaining the audio and video to be encoded, and devices for writing and transmitting the encoded audiovisual data generated by the encoding system;
- 10 Figure 2 is a flowchart illustrating the general operation of the system of Figure 1;
- Figure 3 is a flowchart illustrating a process performed in the set-up state;
- Figure 4 illustrates a process performed in the compression state;
- Figure 5 illustrates the process performed in the edit state;
- Figure 6 illustrates a process performed for the start, completion, and exit states;
- 15 Figure 7 is a flowchart showing how to determine the number of audio encoders needed to encode a predetermined number of audio tracks while using a multipass video encoder, and the process of simultaneously performing audio and multipass video encoding;
- Figures 8A - 8D illustrate the Sequence Layer log file format;
- Figures 9A - 9C illustrate the Picture Layer log file format;
- 20 Figures 10A and 10B illustrate the full and short forms of the Macroblock Layer log file format;
- Figure 11 illustrates a process for utilizing the information within the log files;
- Figure 12 illustrates the general process performed during the manual adjustment of picture quality for different time periods;
- Figure 13 illustrates a user interface which is used to input the parameters for changing the picture quality over various time periods;
- 25 Figure 14 illustrates the general procedure for calculating the number of bits associated with the desired picture quality;
- Figure 15 illustrates the process used for processing sections having maximum or minimum user selected priorities;
- Figures 16A and 16B illustrate a flowchart for calculating the number of bits corresponding to desired picture qualities which are not maximum or minimum priorities;
- 30 Figure 17 illustrates a flowchart of the process used to check if underflow results from the changing of the picture quality;
- Figure 18 illustrates a flowchart for determining if buffer underflow will occur in frames after the edit segment due to changes made within the edit segment;
- 35 Figures 19A and 19B illustrate two different examples of how the bit rates of the encoded video are changed by the user selected quality;
- Figure 20 illustrates the general process performed during the manual adjustment of picture quality for different regions within a frame;
- Figure 21 illustrates an example of different regions having user selected priorities in a frame of video;
- 40 Figure 22 illustrates a graph of the user selected priorities versus the fraction of the original quantizer values used for the selected priority;
- Figure 23 illustrates a process for determining if the difference between the number of bits resulting from the user selected priorities and the number of bits resulting from the original encoding of a frame is acceptable or whether a correction procedure must be performed;
- 45 Figures 24A-24C illustrate a process for correcting a frame when it is determined to have too many bits and therefore, the quantizer levels of the macroblocks are incremented in order to reduce the number of bits;
- Figures 25A-25C illustrate a process for correcting for too few bits in a frame by reducing the quantizer levels of macroblocks within a frame in order to increase the number of bits;
- Figure 26 is a flowchart illustrating the general process of loading from memory the expected number of bits to be used in each of the sets of n macroblocks, and then for each set of n macroblocks re-encoding, counting the resulting number of bits, calculating the accuracy of the estimation based on the expected number and generated number of bits and adjusting the pre-assigned quantization levels of the next n macroblocks to remain close to the target size of the series of frames;
- 50 Figure 27A is a picture of frame illustrating the pre-assigned quantization levels;
- 55 Figure 27B is a picture of the frame of Figure 27A when each set of macroblocks contains one slice and after the first set of macroblocks has been re-encoded, a first correction factor has been calculated and the first correction factor has been added to the second set of macroblocks;
- Figure 27C is a picture of the frame of Figure 27B after the second set of macroblocks has been re-encoded, a

second correction factor has been calculated and the second correction factor has been added to the third set of macroblocks;

Figure 27D is a picture of the frame of Figure 27A when each set of macroblocks contains two slices and after the first set of macroblocks has been re-encoded, a first correction factor has been calculated and the first correction factor has been added to the second set of macroblocks;

Figure 28A is a frame containing two different regions whose quantization levels are changed;

Figure 28B is a picture of two possible encodings of the frame of Figure 28A illustrating the change in the number of bits in a video stream used to represent the two different regions whose quantization levels were changed;

Figure 29 is a video stream illustrating the change in the number of bits used to represent a series of images where the total length of the re-encoded series equals the series original length;

Figure 30 is a flowchart illustrating how a correction factor index is determined from a ratio lookup table using an estimation ratio;

Figure 31 is a flowchart illustrating how a correction factor is chosen from a correction factor table using a correction factor index;

Figure 32 is a flowchart illustrating how a correction factor is calculated using both an estimation ratio and the number of remaining unre-encoded sets of macroblocks;

Figure 33A illustrates a section of original video in an encoded format;

Figure 33B illustrates a section of new video in an unencoded format which is to be substituted into the original encoded video;

Figure 33C illustrates the unencoded new section of video with unencoded sections of the original video at the ends thereof;

Figure 33D illustrates the video in Figure 33C in an encoded format which is to be substituted into the original encoded video;

Figure 33E illustrates the original encoded video having the substituted encoded video included therein;

Figure 34 illustrates a process for substituting a section of video into the encoded video data stream which results in video with reduced decoding artifacts;

Figure 35A illustrates an unencoded new section of video with an unencoded section of the original video at the trailing end;

Figure 35B illustrates a section of encoded video to be substituted into the originally encoded video illustrated in Figure 33A;

Figure 35C illustrates the originally encoded video of Figure 33A containing the substitute encoded video of Figure 35B;

Figure 36 illustrates a process for reducing decoding artifacts when editing an encoded video bitstream by encoding the substitute video using a closed group of pictures mode;

Figure 37A is a frame of raw video that has two quantization levels assigned to it to generate quantization level versus bit-rate models in the pre-encoding phase;

Figure 37B is a frame with four quantization levels assigned in a rotating pattern;

Figure 37C is a frame with four quantization levels distributed in block formation;

Figure 38 is a flowchart which illustrates the process of obtaining video signals and pre-encoding them to determine a bit-rate for each quantization level specified in the pre-encoding phase;

Figure 39 is two recorded frames and their associated quantization levels versus bit-rates;

Figure 40 illustrates an outline of the final format of the encoded data which is ultimately included on an optical disc;

Figure 41 illustrates the disc information file 908 of Figure 40;

Figure 42 illustrates the data file management information contents of the data files illustrated in Figure 40;

Figure 43 illustrates the details of the data stored in the data files of Figure 40;

Figure 44A illustrates a data pack in which the data thereof does not require a stuffing packet and Figure 44B illustrates a data pack which requires a packet for padding;

Figure 45 illustrates a playback information pack;

Figure 46 illustrates the information contained within the data search information 996 of Figure 45;

Figure 47 illustrates the general information of Figure 46;

Figure 48 illustrates the synchronous reproduction information illustrated in Figure 46;

Figure 49 illustrates a video pack;

Figure 50 illustrates the relation between a Group of Pictures (GOP) and a series of video packs;

Figure 51A illustrates an audio pack encoded according to the MPEG audio encoding specification;

Figure 51B illustrates an audio pack constructed according to AC-3 or linear PCM audio encoding;

Figure 52 illustrates the relationship between the encoded audio stream and the audio packs;

Figure 53 illustrates the structure of a sub-picture unit;

Figure 54 illustrates the structure of a sub-picture pack;

Figure 55 illustrates the structure of a sub-picture unit in relation to sub-picture packs; and
Figure 56 illustrates a transition between displaying sub-picture unit n and sub-picture unit n+1.

DESCRIPTION OF THE PREFERRED EMBODIMENTS

[0018] The present invention is a digital video encoding system which includes a plurality of components and functional aspects. In order to more easily identify the various components of the system, this writing is organized into the following sections:

I. INTRODUCTION

II. SYSTEM COMPONENTS

III. GENERAL SYSTEM OPERATION

IV. SYSTEM ALTERNATIVES

V. REDUCING THE NUMBER OF AUDIO ENCODERS WHILE MAINTAINING OPTIMUM ENCODING EFFICIENCY

VI. LOG FILES FOR STORING INFORMATION OF THE ENCODED VIDEO, AND LOG INTERPRETER UTILITIES TO ACCESS THE LOG FILES AND THE ENCODED VIDEO

VII. CHANGING THE QUALITY OF THE ENCODED VIDEO

A. Changing the Quality Over Periods of Time

B. Changing the Quality Within Regions of Frames

VIII. RE-ENCODING THE VIDEO AFTER CHANGING THE QUALITY

A. Bit Allocation Control When the Quality Has Been Changed Within Single Frames

B. Re-Encoding While Avoiding Decoding Artifacts At Edit Points

1. Restoring the Encoder to the State Used When Originally Encoding

2. Re-Encoding in a Closed Group of Pictures Mode

IX. RATE-QUANTIZER MODELING USED FOR ENCODING

X. COMBINING AUDIO, VIDEO, AND SUB-PICTURE DIGITAL SIGNALS USING DATA STRUCTURES

I. INTRODUCTION

[0019] The disclosed system relates to a digital audiovisual encoding system, also known as a digital compression system. The ideas disclosed herein are usable with any type of encoding scheme and are particularly applicable to MPEG-2 (Moving Picture Experts Group) formally known as ISO/IEC 13818-2, ISO/IEC 13818-1 which specifies MPEG-2 system information, and the MPEG-2 predecessor, MPEG-1 (ISO/IEC 11172-video). MPEG-1 and MPEG-2 will collectively be referred to herein as "MPEG". The MPEG standard specifies only the syntax of the compressed or encoded video stream but not how to perform compression. It is to be noted that as an MPEG encoded data stream is a compressed data stream, the encoding process may be referred to as a compression process and vice versa. The system also relates to the encoding of audio which may be performed using Dolby AC-3, or the MPEG-2 standard as defined in ISO/IEC 13818-3.

[0020] MPEG video compression is a temporal causal predictive compression. As the MPEG encoder receives a new video picture, it compares the new video picture to a predicted picture that the encoder has generated based on previously encoded pictures. The term "causal" predictive coding is used because the prediction is made based on previously transmitted information, and the term "temporal" causal predictive coding is used because the prediction is performed in terms of temporal picture units. The prediction procedure involves estimating the motion of areas within a video picture. Therefore, the temporal prediction is sometimes called motion-compensated prediction.

[0021] The result of the comparison between old and new pictures represents new information contained in the later video picture. The new information, also called "residual information," is then subjected to a series of information compaction processes. First, a linear mathematical transformation known as the discrete cosine transformation (DCT) is performed. The DCT operation transforms the pixel residual information into a series of coefficients. In MPEG compression, the DCT is performed over blocks of 8x8 pixels. Similarly, the DCT coefficients are also in 8x8 arrays of numbers. These coefficients are then individually quantized (or rounded) to a set degrees of precision, based upon a quantization step size or q-level. The quantization process typically produces many coefficients having the value of zero. It is this quantizing process which is mostly responsible for introducing coding artifacts, otherwise referred to as quantization noise.

[0022] The quantized coefficients are then coded using a combination of run-length coding on the zero-valued coef-

ficients and variable-length coding of the resultant pairs of two numbers, each representing the run-length value and non-zero coefficient value, respectively. The codes for these run-length non-zero value pairs, together with other overhead information, form the compressed digital data stream corresponding to the video picture. Due to physical limitations such as output data rate or storage capacity, it is necessary to vary the degree of quantization so that an appropriate amount of compression is accomplished. This is referred to as buffer feedback.

[0023] An MPEG-compliant decoder reads the encoded digital data stream and performs operations opposite to the encoding process.

[0024] Typically, the total number of bits in the digital data stream is substantially less than the total number of bits needed to represent the video pictures, and hence compression is achieved. It is important, however, to recognize that such compression as exemplified by MPEG video coding is by no means free of distortion or artifacts. As mentioned, the principal source of compression artifacts is the quantization process.

[0025] The disclosed system is aimed at dynamically adjusting the degree of quantization during the compression or encoding process. In particular, adjustment is made in accordance with instructions provided by a human operator. The adjustments may apply to parts of a larger video sequence, or possibly to the entire video sequence.

II. SYSTEM COMPONENTS

[0026] Referring now to the drawings, wherein like reference numerals designate identical or corresponding parts throughout the several views and more particularly to Figure 1A thereof, there is illustrated a block diagram of the encoder system architecture. It is to be noted that the video encoding process is a complex process which requires a number of different devices which operate synchronously in order to achieve the encoding and editing without interruption or errors. However, the audio and video encoding can be separately performed as long as the appropriate time codes are used. While four workstations have been illustrated in Figure 1A, ordinarily, the user or operator of the encoding system will interact with only one keyboard and workstation. The workstations contain conventional general purpose computer components such as a processor, RAM, ROM, a network interface, a hard disk, a keyboard, monitor, and other conventional components.

[0027] A workstation 30 functions as the system control station which, in the preferred embodiment, is a Sun SPARC20 Unix workstation. Workstation 30 is equipped with a graphical console display monitor 31 as well as conventional workstation input devices such as a keyboard 32 and a pointing device 33 which is a mouse or any other pointing device. The user visually interacts with the workstation 30 primarily through a graphical user interface (GUI) which is displayed on the console display 31.

[0028] The workstation 30 is connected to and controls the other workstations 40, 10 and 20 which are described below. Workstation 30 also communicates with the audio interface unit 72, and the digital video capture and storage device 60 which are also discussed below. The workstation 30 is connected to the workstation 40 through a serial interface as defined by the Electronic Industries Association (EIA) standard RS-232. Similarly, the connection between workstation 30 and the audio interface unit 72 is also by means of the serial RS-232 standard. The connection between the workstation 30 and the digital video capture and storage device 60 is through an industry-standard S-bus interface.

[0029] A frame jogger 41 is connected to the workstation 40 to allow the operator to position the video of either of the VTRs 51 or 52 to where editing is to be performed. The system is capable of using time codes input by an operator to control the position of the video. However the jog controller enhances the user-friendliness of the system by allowing the use of a jog knob to position the video. Workstation 30 controls whether the displayed video on the video display monitor 61 is from the digital video capture and storage device 60 and/or from the record VTR 52.

[0030] Video encoding device 50 is a digital video encoder which performs compression according to the MPEG-1, MPEG-2, or other video compression standard. A commercially available MPEG-1 encoder is the Sony RTE-3000. An MPEG-2 encoder can be constructed according to the teachings set forth in Japanese Patent Application 6-326435, filed December 27, 1994 (JP 8 186 821 A). Workstation 10 directly controls the video encoding device 50 through the industry standard S-bus interface and the workstation 10 receives commands from workstation 30 through standard networking equipment such as an Ethernet network. During the encoding process, workstation 10 supports real-time execution of the video encoding control program of the video encoding device 50. In the preferred embodiment, the workstation 10 is a Sun SPARC20 Unix workstation.

[0031] The workstation 10 responds to commands from the central workstation 30 to begin the compression process. During the compression process, workstation 10 dynamically controls the degree of quantization applied to particular areas of the video signal. The particular areas for which the quantization is varied extend over the spatial and/or temporal domains and the exact manner in which the quantization is controlled is explained below.

[0032] The input video to the video encoding device 50 is from a tape within the playback VTR 51. The playback VTR 51 is a professional digital video standard, ITU-R 601 (formerly known as CCIR 601 video standard) video tape player. This standard applies to both NTSC as well as PAL systems. It is noted that the tape within the playback VTR 51 contains electromagnetic information representing various images and scenes which have been captured through a video or film

camera, or generated by a computer. The playback VTR 51 operates under the control of the workstation 40. The encoded output from the video encoding device 50 is transferred to a workstation 20 for storage on one or more hard disks 21.

[0033] In order to view video data which has been encoded and output by the video encoding device 50, a digital video decoder such as an MPEG-2 decoder is necessary. However, at the time of this writing, the implemented system does not incorporate a separate MPEG-2 decoder as such a decoder was not readily available during the early development stages of the system. Therefore, the video encoding device 50 not only outputs an MPEG-2 video stream to workstation 20, but also a decoded video image equivalent of the MPEG-2 encoded data is output from the video encoding device 50 to the record VTR 52. The record VTR 52 is connected to the video encoding device 50 by the industry-standard digital video signal connector known as the D1 interface and controlled by the workstation 40. Both the playback VTR 51 and the record VTR 52 are suitably implemented using Ampex DCT professional video tape recorders.

[0034] The audio is encoded and compressed, for example, into the Dolby AC-3 format, or alternatively an MPEG-2 compliant format such as described in the ISO/IEC 13818-3 standard. The audio sources for the encoding system are four digital audio tape (DAT) players 71a, 71b, 71c, and 71d. Studio quality DAT players are commercially available from Sony. The DAT player 71a-71d are connected and controlled via the industry-standard protocol RS-422 by the workstation 40. The audio signals output by the DAT players 71a-71d are input into audio encoders 70a, 70b, 70c, and 70d respectively. These audio encoders are implemented using the commercially available DOLBY AC-3 model DP 525 encoders and are controlled by the audio interface unit 72. Typically, the audio encoding will begin and end at specified time codes. The audio interface unit 72 receives instructions such as encoding start and ending time codes from the workstation 30 over an RS-232 connection therebetween. Additionally, the audio interface unit 72 multiplexes the various digitally compressed audio data into a single digital stream transmitted to workstation 20. Workstation 20 demultiplexes the input digital audio stream into multiple files corresponding to the compressed digital data output by the audio encoding devices 70. The video encoding device 50 provides a synchronization signal based on its video synchronization signal to the audio interface unit 72. The sync signal allows the audio bitstream to be synchronized to the video bitstream output by the video encoding device 50.

[0035] Workstation 30, has the capability of instructing the audio interface unit 72 to provide a duplicate of the audio data from any one of the various audio encoding devices 70a-70d to the audio decoder 73. The audio decoder 73 decodes the compressed audio so that a user may listen to the audio via a speaker 75. The choice of which audio encoding device to monitor is made by the human operator through interaction with the graphical user interface on the console display 31 via either the keyboard 32 or pointing device 33. The construction of the audio interface unit 72 is hardware which is capable of performing the above-described functions. This hardware is one or more programmed microprocessors and/or conventional circuit elements arranged to achieve the disclosed functions.

[0036] Workstation 40 is the studio equipment control station and issues appropriate commands to control both the playback video tape recorder 51 and the digital audio tape players 71a-71d along with the recording video tape recorder 52. The workstation 40 also instructs the digital video capture and storage device 60 to begin capturing video at the appropriate time. The connections between workstation 40 and the various video and audio tape decks is by means of the industry standard RS-422 protocol. This protocol allows each tape deck to inform the workstation 40 of its current tape location in standard time code format. The workstation 40 uses such information to properly synchronize the various tape decks so that the video and audio information is played out and recorded correctly. In the present embodiment, workstation 40 is a standard IBM compatible personal computer running DOS and Ensemble Pro, a commercially available software program from the Editing Technologies Corp., Moorpark, California. It is noted that a minor modification was made to the Ensemble Pro program so that workstation 30 can communicate with Ensemble Pro running on the workstation 40 through an RS-232 communication port. This modification is either commercially available through Editing Technologies Corp., or may be performed by one of ordinary skill in the art without undue experimentation.

[0037] A digital video capture and storage device 60 serves two functions within the encoding system; first, this device uses its capture and storage capability to provide a convenient "before-and-after" comparison of manual editing that has been applied to the video signal, and second, provides a graphical overlay feature to the video display. This graphic overlay feature is used, for example, during the spatial manual editing phase, to allow the human operator to define the regions of interest in which the quantization and consequently the number of bits allocated is varied.

[0038] In the present invention, the digital video capture and storage device is implemented as a commercially available DRAM-based video product, model VS6000, from Viewgraphics, Mountain View, California. The digital video input to this device is by means of the industry standard D1 having ITU-R-601 (formerly CCIR 601) image resolution. The output of the digital video capture and storage device 60 is connected to the video display monitor 61 through three separate analog signals representing the red, green and blue components of the video signal. The graphics overlay information is processed by the digital video and capture storage device 60 prior to creation of the final analog signal output by the digital video capture and storage device 60 and therefore will be visible to the human operator along with the other video data on the video display monitor 61.

[0039] The workstation 30 running under the Unix environment maps graphical windows thereof onto the video display

monitor 61 using the digital video capture and storage device 60. This allows the human operator to perform graphical operations (e.g., drawing rectangles) on the graphical overlay plane of the display shown in monitor 61. The device 60 provides an overlay function. The overlay is performed in video memory prior to the conversion of the information in the memory to an analog video signal which drives the monitor. In the VS6000 used by the system, there is an 8-bit memory plane on top of the video memory. On this 8-bit plane, the computer can draw any color graphics and text. One of the "colors" on this 8 bit plane is the transparent color. Any pixel on this overlay plane that has the transparent color will take on the value of the video memory underneath it. In general, most of the overlay plane will be filled with the transparent color, with graphics (e.g. lines, text, etc) using non-transparent colors. Thus, a display memory is formed which consists of both the video memory and the overlay graphics memory. It is the display memory that is ultimately converted to an analog signal for display on the monitor 61. The human operator typically uses the pointing device 33 associated with the keyboard 32 to perform such graphical operations and manipulations of graphical objects thus created.

[0040] A workstation 20 receives digital data streams from the video encoding device 50 as well as the audio interface unit 72. The workstation 20 is connected to the workstation 30 via an Ethernet connection, and the workstation 20 is also connected to the video encoding device 50 and the audio interface unit 72 through the industry-standard S-bus interface. The received digital data streams are stored in one or more hard disks 21 as separate data files. Separate Direct-Memory-Access (DMA) cards are used in transferring the captured digital stream data to the hard disks 21. For long video sequences that result in large quantities of digital data, the hard disks 21 may be implemented using commercially available hard disks such as the SPARCstorage Array X655A/G5 from Sun Microsystems. Workstation 20, in this embodiment, is a commercially available SPARCserver 1000 from Sun Microsystems.

[0041] Workstation 20 also formats the data files corresponding to the video and audio compressed data into a single formatted file suitable for transport or transmission to decoding devices. The final medium is preferably disc-based, in which case the formatting will be in accordance to the pertinent disc format. Typically, the formatted file is transported by some intermediate digital storage means, such as a digital tape, to a facility where the transfer to the final transmission medium or media takes place. The digital storage devices 22, shown in the figures, serve such a purpose. In the present embodiment, digital storage devices 22 include a commercially available 8mm Exabyte tape drive. The interface of the digital storage device 22 is the industry-standard Small Computer System Interface (SCSI).

[0042] It is known that multiple SCSI devices can be attached to the same SCSI bus. Therefore, the digital storage devices 22 can include other SCSI devices such as the commercially available digital linear tape (DLT) drive, a magneto-optic (MO) disc drive, or a floppy disk drive. These configurations are easily accommodated, and would serve to provide a flexible input mechanism for any auxiliary data that may be needed prior to or during the formatting stage. For example, in movie production, subtitle information is often available in a medium that requires one of the SCSI devices mentioned above. In such cases, the auxiliary data is read by software programs executing in workstation 20 and manipulated so that the auxiliary data is included into the formatted files. The auxiliary data may include subtitle data which is in the U.S. Close-Caption format. The auxiliary data can include any desired data such as program codes or postscript data.

[0043] Figure 1B illustrates an audio/visual encoding system 98, which corresponds to and functions in the same manner as the system of Figure 1A. In Figure 1B, video input to the encoding system is supplied from a camera 80 and audio is supplied from a microphone 82. Movie film containing visual and possibly audio information is transferred to the encoding system 98 through the transfer device 88 which contains a light to illuminate the film and a light detector to capture images from the film. The information from the input devices 80, 82, and 88 is stored in an electro-magnetic format before encoded by the encoding system 98, such as by an audio or video tape recorder. The audio and video tapes are encoded by the system of Figure 1A.

[0044] The final digital bitstream produced by the encoding system is sent to optical disc mastering equipment 90 which presses optical discs, an optical disc writer 84 which writes optical discs, or may be transmitted to a television receiver or set-top box decoder via the television transmitter 86. The pressing and creation of optical discs is performed in a known and commercially available manner using a master disc, for example. The encoded information is also usable with a video on demand system. The encoded bitstream is ultimately decoded by decoding processes which correspond to the encoding processes and the audiovisual information is displayed to a user on a television or monitor. The decoding devices include digital video disk players which decode the encoded disks for display on a television or monitor.

[0045] It is evident from Figure 1B that the encoding system inputs electrical signals representing audio and images, transforms the electrical signals to a new format, and ultimately, the signals are decoded to recreate the original information which was encoded.

III. SYSTEM OPERATION

[0046] Figure 2 is a flowchart illustrating the general operation of the system illustrated in Figure 1A. As seen in Figure 2, the system goes through several "states" during its cycle of operation. The dark solid lines show the normal operational path, the solid lines illustrate an interrupt exit path, and the dashed lines illustrate a by-pass path.

[0047] The first operation by the system occurs in the set-up state 100. This step allows input parameters to be specified

before the compression state 102 is entered. In the compression state 102, a video image stored on a tape in the playback VTR 51 is encoded into a digitally compressed format by the video encoding device 50.

[0048] After the compression state 102, the user is permitted to manually edit the compression process based on the picture quality of the previously compressed data in the edit state 104. This allows the user to increase or decrease the picture quality within regions of a frame of video or over a time period. It is to be noted that the term "edit" does not mean that the scenes themselves are added or removed from the picture stream. In the present application, the term "edit" means to change the quantization and subsequently the quality of the video. After the image is edited in the edit state 104, it is then necessary to perform the compression operation again in the compression state 102 in order to encode the data according to the new user-specified parameters.

[0049] After the user is satisfied with the edited form of the video, all information including the audio, video, and other information such as subtitles are combined together into a desired format in the format state 106. The completion state 108 erases any temporary files and the process terminates in the exit state 110. More detailed information with respect to the individual states illustrated in Figure 2 will now be explained with respect to Figures 3-6.

[0050] Figure 3 illustrates the steps performed in the set-up state 100. Step 120 begins the set-up state and step 122 initialized the system. This step will begin the execution of system control software contained within workstation 30 and start up other programs such as a graphical user interface (GUI) program which is responsible for the user interface displayed on the console display 31 and having parameters input through the keyboard 32 and pointing device 33. Also during step 122, the system controller software in workstation 30 queries other devices to determine the system condition including the available devices of the system.

[0051] Step 124 displays information to the user through the GUI. A welcome message, information pertaining to the system, parameters used for the encoding process, and the existence of any previously encoded files are displayed. The displayed system information includes the devices connected to the system, including audio encoders, VTRs, audio tape decks, and their configuration. In the event that the system contains previously encoded files, information is displayed to the user describing the audio and video encoding parameters used to generate the files.

[0052] Step 126 then inputs information from the user such as a command to begin encoding using standard or user specified system parameters, any changes to the system parameters, or a confirmation that the selected parameters are acceptable. The user also enters the beginning and ending time codes for the encoding operation.

[0053] Step 128 then processes the input data by transferring the input data from the graphical user interface to the system controller software in order to prepare for the encoding and compression operation. The set-up state ends in step 130.

[0054] Figure 4 illustrates the process performed in the compression state 102. According to the normal operation flow, the user has already specified a video and any accompanying audio sequence to be compressed in the set-up state. The user must then instruct the system to perform an automatic or auto-compression procedure. The video compression during the auto-compression procedure does not permit the user to control the dynamic quantization process over a time period or within a frame or video. It is intended as an initial compression run in which the coding decisions are made based on objective measures as computed by the video encoding device 50. If the auto-compression has already been performed, the user may manually adjust the quantization in an edit-compress mode either spatially (over a frame or video) or temporally (over a video sequence).

[0055] In the compression state, after starting in step 140, step 142 prepares for the compression operation. In this operation, parameters for the compression process such as the time codes of the video and any accompanying audio to be compressed along with the compression parameters are downloaded to the video encoding device 50 from the workstation 30 through the workstation 10.

[0056] Step 144 then determines the type of compression which is desired. The first time information is compressed, the compression process is performed automatically and without user intervention. Step 146 prepares for the auto-compression process. During this preparation, the workstation 30 determines which of the digital audio tape players 71a-71d and the associated audio encoding devices 70a-70d are to be used. Also, the workstation 30 commands the workstation 20 to prepare for bitstream capture from the output of the video encoding device 50 and the audio encoders 70a-70d through the audio interface unit 72. Further, the playback VTR 51, the record VTR 52 and the DAT players 71a-71d are rolled to the appropriate starting position of the tapes. Workstation 40 then signals the playback video tape recorder 51, the record tape recorder 52, and the DAT players 71a-71d to begin rolling in step 148. Step 150 then encodes the video data according to the desired format such as an MPEG-1 or MPEG-2 format using the video encoding device 50. Also, the audio data is encoded according to the desired format such the Dolby AC-3 format.

[0057] It is necessary to create log files to assist later in the manual editing process. The log files indicate parameters of the encoded video data which are necessary in order to perform later editing of the video data. The encoding process is stopped in step 160 when the appropriate time code is reached. This time code may be the end of the information to be encoded or may be at a user specified point which is not at the end of the video sequence.

[0058] If step 144 determines that the auto-compression process has already been performed and a user desires to manually change the picture quality of one or more frames of video using the parameters specified during the edit state

as described with respect to Figure 5 below, step 144 will determine that the edit-compression operation is to be performed. Step 152 prepares for the edit compression operation which includes cueing of the playback VTR 51 to the desired starting point which is the point at which the manual editing begins. Also, the audio equipment is disabled as there is no need to change the audio information once it is encoded in the auto-compression operation. Step 154 then rolls the

5 playback VTR 51 and the record VTR 52.
[0059] Step 156 subsequently begins encoding the video data according to the parameters which have been specified by the user in the edit state described in Figure 5. The edit compression is terminated in step 160 when the ending time code is reached. Step 162 displays a message to the user that the compression and encoding is complete and the process ends in step 164.

10 **[0060]** Figure 5 illustrates the process performed in the edit state 104. As previously explained, the editing referred to is not the conventional editing of video where scenes are removed, shortened, or moved within a video sequence; the edit state in the present writing refers to changing the quality of a portion of or time sequence of the video by manually setting the quantization of the video.

15 **[0061]** After starting the edit state in step 170, step 172 asks a user if he wishes to manually edit the video by changing the bit allocation of the encoded video. If the user does not desire to change the bit allocation, the process ends at step 188. If the user desires to change the bit allocation, the user defines the video segment to be edited in step 174. This is performed by selecting a time period of the video over which the picture quality is to be changed. Step 176 then determines if the user desires spatial or temporal editing. Spatial editing is where the quantization or allocation of bits is changed within a frame of video whereas temporal editing is where the quantization or allocation of bits is changed over a time period. If the user selects spatial editing, flow proceeds to step 178 which inputs regions within a frame to be edited. Also inputted is the relative degree of change to be applied. For the present invention, an integer scale of -5 through +5 including zero is used to indicate the relative amount of change. A user will mark a region of video with the keyboard 32 and/or pointing device 33 and assign one of the integers between and including -5 through +5. It is also possible for the user to indicate that the previously set quantization level for a region is not to be changed. For example, 20 if the user desires an increased bit allocation for a certain region, it is necessary that the bit allocation for other regions must be decreased. If the user sets a region to be non-changeable, also referred to as "protected", the needed bits for the improved quality area will not be taken from the protected area.

25 **[0062]** If step 176 determines that the user desires temporal editing, flow proceeds to step 180 which inputs how the time periods are to be edited. Similar to the spatial editing, for the temporal editing, the user assigns integer values between -5 and +5 indicating the relative degree of change to be applied to a particular sequence of video. The change will be effective for the entire time period which has been selected.

30 **[0063]** After either the spatial or temporal editing is performed in steps 178 and 180, step 182 re-encodes and recompresses the video scene to an MPEG-2 format, for example, according to the parameters which were manually entered. The previously encoded data is displayed next to the newly edited data on the video display monitor 61 in order for the user to closely compare how the quality has been changed.

35 **[0064]** When the operator indicates the segment to be edited, the video from the VTR 52 is transferred to the storage device 60, taking up at most half of the total available memory in the storage device 60. That segment corresponds to the "before" segment. During the edit-compression process, the bitstream is captured by workstation 20, and stored as a separate disk file from the master video bitstream file. The master video bitstream file is the compressed bitstream for the entire movie. The edit-compressed video is recorded in VTR 52. During the encoding process, the operator will be viewing the picture at the same time it is being recorded on VTR 52. Should the operator wish to view the picture after the encoding, the VTR can be used as a playback machine. Should the operator wish to do a "before-and-after" comparison, then the corresponding segment from the VTR 52 is transferred to the storage device 60, consuming up to the remaining half of the total memory of the device 60.

40 **[0065]** After this encoding, the operator makes a decision as to whether the changed video is to be kept. Should the operator decide to disregard the "after" segment, the master bitstream is left intact, and the edit-compressed bitstream is deleted. To allow the VTR 52 to reflect the master bitstream, an additional auto-compression on the edited segment is performed solely for the purpose of recording the "before" segment. The alternative of using the device 60 to transfer the "before" segment back to the VTR 52 is also possible. Should the operator decide to keep the "after" segment, then the master bitstream file must be updated to include the edit-compressed bitstream file, as discussed below.

45 **[0066]** If the user determines that the video is acceptable in step 184, the previously encoded data is replaced with the newly encoded data. This step is explained in further detail in a separate section below. After the newly encoded data replaces the previously encoded data, step 192 determines if the user desires more editing. If more editing is desired, flow proceeds back to step 174. If the user has finished editing the video, flow proceeds to step 188 which terminates the edit state.

50 **[0067]** If the user determines that the video is not acceptable in step 184, the user may exit the editing operation in step 186 so that the editing changes are not recorded or the user may re-edit the video. The video is re-edited by the users selecting new parameters in the editing. It is also possible, if the user desires, to define a new video segment to

be edited.

[0068] When the user is satisfied with the compression (either auto-compression alone, or auto-compression in conjunction with edit-compression), the individual data files including the compressed video, compressed audio, and any other files are ready for formatting into the final format and the format state 106 is entered. The auxiliary data may include information such as subtitles to be included into the final formatted data file. In the flowchart illustrated in Figure 6, after starting the format state in step 200, the workstation 20 reads the auxiliary data, should the user decide to include the auxiliary data, from the digital storage devices 22 which include, for example, a floppy disk within a floppy disk drive. Step 204 then combines the auxiliary data with the encoded audio and video to generate the final formatted file. The final digital bitstream generated by the system may be constructed as a 'Program Stream' as defined in ISO/IEC 13818-1. Alternatively, the final bitstream may be any type of format such as the direct satellite DSS format, a VBR (variable bit rate) format suitable for use on an optical disc, or any other type of format. As the encoded video and audio may be stored separately, the system allows the same encoded video and audio to be encoded into different final formats. This is accomplished either using one formatter having the capability of generating different formats or by separate formatters. Step 206 stores the formatted file onto the disk 21.

[0069] The completion state is entered in step 208 which writes the formatted file to a tape within the digital storage devices 22 in step 210. Subsequently, the exit state is entered in step 212 which performs various "house cleaning" functions which delete temporary files which are no longer needed. In the event of an abnormal exit request such as before the compression process is finished, an abort procedure will be carried out within the workstation 30 which stops the audio and video tape players and also cleans up potentially corrupted data files. The process then ends in step 216.

IV. SYSTEM ALTERNATIVES

[0070] The preferred system components and their operation have been described above and of course, alternative hardware may be substituted for the hardware disclosed in Figures 1A and 1B. The video encoding device 50 is preferably an MPEG-2 video encoder, if available. However, a commercially available ISO/IEC11172-Video standard encoder otherwise known as an MPEG-1 video encoder may be used. Necessary modifications to the MPEG-1 encoder will need to be made to allow the MPEG-1 buffer management module to be controlled from an external source such as the workstation 10. Additionally, the input video support must also be augmented to include ITU-R 601 (formally CCIR 601) resolution video. Such modifications can be implemented by those skilled in the art of digital video encoding without undue experimentation.

[0071] While four audio tape players and four audio encoders have been illustrated in Figure 1A, other configurations can be easily adopted for the audio system. For example, a commercially available 8mm digital audio player may be used instead of a multiplicity of DAT players. Using a single TASCAM digital audio player, only a single RS-422 control signal is needed from the workstation 40 while up to eight separate audio channels can be simultaneously supported. The audio encoders must of course be able to accept eight audio inputs simultaneously and the audio interface unit 72 must also be modified to accommodate the increase in audio data streams from four to eight.

[0072] As an alternative to simultaneous audio and video encoding, the audio and video encoding can be done at separate times and/or locations and combined into the final format at a later time. This requires the use of time codes, which are known, such as the SMPTE (Society of Motion Picture and Television Engineers) time code format. The time codes are generated by a video tape player, a workstation, or a separate stand-alone time code generator.

[0073] The record VTR 52 may be eliminated if a video decoder is available and connected to the workstation 20. In that case, the reconstructed video would be generated from the disk files in disks 21 instead of being recorded from the video encoding device 50 during the compression phase. Eliminating the record VTR 52 will have a significant savings in cost of the system, both in terms of equipment as well as tape cost.

[0074] Furthermore, the ability to perform graphical operations onto the graphic overlay plane of the display video can be supported on the console display using the commercially available X-terminal option. For example, the console display 31 may be replaced with an X-terminal that is capable of displaying video with an overlay graphic window. A commercially available multimedia display terminal such as the HDSV View Station from Human Designed Systems of King of Prussia, Pennsylvania, provides the necessary features. The video from the video decoder can then be connected to the X-terminal for display and for manual editing region definition operations. The professional quality video from the video decoder should, however, be displayed on a professional monitor such as monitor 61 shown in Figure 1A to allow the human operators to assess the quality of the reconstructed video signal.

V. REDUCING THE NUMBER OF AUDIO ENCODERS WHILE MAINTAINING OPTIMUM ENCODING EFFICIENCY

[0075] As explained above, Figure 1A illustrates an encoding system having a multipass video encoding device 50 and four one-pass audio encoders 70a, 70b, 70c, and 70d. The playback VTR 51 supplies the video signal to be encoded to the video encoding device 50 and the digital audio tape (DAT) players 71a-71d supply the audio signals to be encoded

to the one-pass audio encoders 70a-70d, respectively.

[0076] The preferred embodiment of the invention encodes eight audio tracks in the final formatted audiovisual data stream. Each audio track includes one or more audio channels. As an example, an audio track may include left and right channels, although the audio tracks can include audio in other arrangements. The manner of determining the number of audio encoders in order to perform the most efficient audiovisual encoding and the manner of performing this audiovisual encoding is explained in Figure 7. In Figure 7, after starting, step 240 determines P, the number of passes required for video encoding. In order to encode the video into an MPEG format, typically two or more passes are necessary to encode the video. For the present invention, the preferred number of passes is two but three or more passes are also possible. Step 242 then determines T, the number of audio tracks to be encoded. In the present invention, the number of audio tracks is preferably eight although different numbers of audio tracks are also possible. Next, step 244 calculates AE, the number of one-pass audio encoders which are needed to perform the most efficient audiovisual encoding. The number of audio encoders necessary is equal to the number of audio tracks to be encoded divided by the number of passes required for the video encoding. In the preferred embodiment, eight audio tracks divided by two passes for the encoder is equal to four audio encoders necessary. If the calculated number of audio encoders is not an integer, it is necessary to round up the calculated number of audio encoders to be an integer. For example, if only seven audio tracks are to be encoded for a two-pass video encoder, $7/2 = 3.5$, and 3.5 rounded up to the next highest integer is 4. During the encoding process performed by this system, one video pass uses only three of the audio encoders while the other video encoding pass uses all four audio encoders. It is to be noted that steps 240-244 may only need to be performed upon the initial designing of the system, and it is not necessary to calculate the number of one-pass audio encoders needed every time an audiovisual work is to be encoded. Further, steps 240, 242 and 244 are either computer/machine implemented by a computer such as workstation 30, or may be determined by a user or designer of the audiovisual encoding system.

[0077] After the number of audio encoders has been determined and the audiovisual encoding system is set up, the audiovisual encoding process may begin. Step 246 sets COUNTER equal to one. COUNTER is a variable used to count each pass of the video encoding system. Step 248 then performs the first video encoding pass while performing one-pass audio encoding on some, but not all, of the audio tracks. For example, if there are eight audio tracks and a two-pass video encoder, four audio encoders may be used during the first pass. Step 250 then increments the COUNTER by one. Step 252 performs the next video encoding pass while performing the one-pass audio encoding of unencoded audio tracks using the same audio encoders which were used in step 248. For example, during the second video encoding pass, the four audio encoders used in step 248 can encode audio tracks five through eight. Step 254 then determines if COUNTER is equal to P, the number of required video encoding passes. In the preferred embodiment, there are only two video encoding passes and therefore, the process would end. Otherwise, flow returns back to perform the loop of steps 250, 252, and 254.

[0078] Of course, other embodiments are possible besides using a two-pass video encoder for an audiovisual work which has eight audio tracks. For example, a three-pass video encoder may be used with six audio tracks. In this situation, the most efficient number of audio encoders is three ($6/2 = 3$). If a two-pass video encoder were used and a total of six audio tracks were to be encoded, the most efficient number of audio encoders would be three. As an alternative, a three-pass video encoder may be used in a system in which there are five audio tracks to encode. This alternative system requires two one-pass audio encoders with any one of the video encoding passes operating while only one of the audio encoders is used and the other two video encoding passes occurring while both of the audio encoders are being used. Of course, other combinations of video encoders and numbers of audio tracks are possible as will become apparent to one of ordinary skill, in light of the teachings contained herein.

[0079] An important aspect of the present invention in being as efficient as possible is to always perform audio encoding when a pass of the video encoding is occurring and to always perform a pass of the video encoding when the audio encoding is occurring. In this arrangement, at least some of the audio encoders will always be used during each pass of the video encoding operation. To be as efficient as possible, when the number of audio tracks is not evenly divisible by the number of audio encoders (i.e., there is a remainder when the number of audio tracks to be encoded is divided by the number of video passes), during the entire encoding process, there will be a number of audio encoders idle which is equal to the number of video passes minus the remainder. For example, for a three pass video encoder and four audio tracks, two audio encoders are needed and there is a remainder of 1 (e.g., $4/3 = 1$ with a remainder of 1). Therefore, to be as efficient as possible and complete the audio encoding at the same time the three pass video encoding is completed, there can be no more than two (e.g., 3 video passes minus a remainder of 1 is equal to 2) times that the audio encoders are idle. The two times that the encoders are idle may occur in the same pass (there are two different audio encoders idle in the same video encoding pass) which would mean that the other two video encoding passes in the above example each utilize all of the audio encoders, or there may be two video encoding passes in which there is one idle audio encoder. In these two passes, the same or different audio encoders may be idle.

VI. LOG FILES FOR STORING INFORMATION OF THE ENCODED VIDEO, AND LOG INTERPRETER UTILITIES TO ACCESS THE LOG FILES AND THE ENCODED VIDEO

[0080] The disclosed encoding system initially encodes video into a compressed digital format such as MPEG-2, MPEG-1, or other digital video encoding format. This encoding is described above in section II pertaining to the system components and section III pertaining to the general system operation. The encoded video data is then stored in a digital format on a hard disk, for example.

[0081] During the digital video encoding process, statistics and other information of the video are generated and stored in log files. One of ordinary skill in the art who is familiar with MPEG video encoding is familiar with the type of statistics and information which describe the video and how to generate these statistics and information. In a conventional encoding system, information and data of this type is generated but ordinarily discarded after the video encoding. However, for the present invention, this information which has been generated such as the total bits used to code a picture, the mean squared error encoding a picture, the mean quantizer scale for each picture, etc., is dumped from system RAM to log files after the encoding process. Examples of log files used by the present invention are illustrated in Figures 8A-10B. These log files are particularly useful in a VBR system and are generated during an initial pass (passes) of the video encoder. The log files are then used during the final encoding pass and during later editing and re-encoding of the video.

[0082] There are three different types of log files which may be used by the present invention; a sequence layer log file, a picture layer log file, and a macroblock layer log file. Each of these log files describes statistics and information of its respective layer. There are two different types of macroblock layer log file formats; the full format illustrated in Figure 10A and a short format illustrated in Figure 10B. The full format is used when detailed information of the macroblock layer is desired and a short format is used when it is not necessary to keep track of all of the details of the macroblocks. Due to the large number of macroblocks in a two hour movie, the macroblock log files will consume a large amount of storage space. Therefore, unless a large storage space is available, it may not be possible to store the full or short format macroblock files. However, if it is necessary to obtain the macroblock information, it can be reconstructed by decoding the encoding video or by other means.

[0083] It is to be noted that many of the details of the log file formats illustrated in Figures 8A-10B are unimportant and these file formats have been provided simply as illustrations of the data which may be generated during the encoding process. However, some of the information in the log files is important as it is used during the process of changing the quality of the video. As will be explained below, in order to determine an edit point of the encoded video, it is necessary to sum the number of bits consumed by earlier pictures. The information which is summed up is illustrated in Figures 9B and 9C and pertains to the number of bits for each picture (generated_bit_picture), the number of bits for the sequence header (bits_sequence_header), the number of bits for the group of pictures (GOP) header (bits_GOP_header), and the number of bits used as stuffing or bits used to fill space (stuffing_size).

[0084] Figure 11 illustrates a flowchart of different functions which are performed using the log files. After the initial encoding of a video segment, step 270 reads in the picture and sequence layer log files generated from the automatic run of the video encoder which are illustrated in Figures 8A-9C. Step 272 then creates an index table indicating the byte offsets of the log file records for every picture recorded in the encoded video. Two tables are created; one indexed by frame number and the other by time code. The time code index table contains the byte offset into the log file of each picture and is sorted in the increasing order of time codes. Similarly, the picture frame index table contains the byte offsets into the log file which are sorted in the increasing order of input picture numbers (as seen by the encoder).

[0085] The generated index tables help to quickly pick out the log information of any desired picture or macroblock. Although fixed size records are used to store the sequence, picture or macroblock data within the log file, the encoder re-orders the incoming pictures prior to coding them. In addition, when dealing with a movie (24 frames per second) source that has been 3:2 pulled down to 30 frames per second, certain time codes are skipped. The index tables provide a means to quickly locate the appropriate data despite the reordering of pictures and the skipping of time codes.

[0086] It is to be noted that the present encoding system does not maintain a directory or index of the number of encoded bits used in coding each picture as the encoded video will frequently change as various frames and time periods of the video have their quality manually altered. However, the log files describing the information of the encoded video are fixed size records and therefore, it is easy to maintain a directory or index of the information within the log files, even when the quality of the pictures or time periods is altered.

[0087] After the creation of the index tables, step 274 allows a user to quickly obtain various information from the log files using various log file utilities illustrated in blocks 276, 278, and 280. When records in the log files are needed, the created index is used to determine the desired record in the log file and the complete record is pulled out of the log file. This record is then parsed for retrieving the individual parameters which are desired. The process of parsing individual records with the help of index tables is the basis for all of the other log interpreter utilities. Parsing processes are known and one of ordinary skill in the art knows how to retrieve desired information from an organized file.

[0088] Step 276 is the log file cut and paste utility. After manually changing the quantization (and picture quality) within a frame or over time periods, the original set of log files needs to be updated to correspond to the changes which occurred

during editing. The log file cut and paste utility locates offsets in the log files corresponding to the edited segment time codes and replaces the information in the log files with new information corresponding to the re-encoded video. The process of re-encoding is explained below in section VIII.

[0089] Step 278 illustrates a utility to obtain parameters to enable editing of the digitally encoded bitstream. After the video is initially encoded and a user desires to alter the quality of frames or a time period of the video, it is necessary to determine the appropriate parts in the encoded video which are to be removed and replaced with the newly encoded video having the desired quality.

[0090] The byte offsets for the edit points in the bitstream of the encoded video are determined by calculating a cumulative tally of the bytes used to encode each picture up to the pictures corresponding to the edit point. In order to determine the total number of bytes or bits from the beginning of the file at which a specific picture is located, the number of bits used to encode each of the previous pictures is summed. The information which is obtained from the picture log file which is summed are the number of bits B_i used to encode the i^{th} picture, the number of bits S_i comprising the sequence header, the number of bits G_i used to encode the Group Of Pictures (GOP) header, and the number of bits T_i used as stuffing. The bit offset for the N^{th} picture is calculated as

$$\sum_{i=1}^{N-1} (B_i + S_i + G_i + T_i) \quad (1)$$

The byte offset is simply calculated by dividing the bit offset by eight. As an alternative to the use of the log files to determine the address offset in the encoded video bitstream, a bitstream decoder could be used to create a directory of picture offsets. However, this approach is cumbersome as the decoder has to run through the entire bitstream after each edit in order to update the directory.

[0091] Finally, step 280 illustrates a utility for rounding time codes to the start or end of the nearest Group Of Pictures (GOP). All cutting of the encoded bitstream or edits are performed at the boundaries of a Group Of Pictures (GOP). However, the GOP boundaries are unknown to the operator performing edits on the video. Once the operator specifies the edit segments start and end points, the system computes the GOP boundaries. The GOP boundary is established by looking at the picture layer parameter called GOP number stored within each picture record. By tracing through successive records of the picture layer log file, when the GOP number (GOP_number) in Figure 9B changes, there is determined to be a GOP boundary.

[0092] As the log files and log file utilities are closely related to video encoder, the log files are stored in a hard disk in the workstation 10 illustrated in Figure 1A and the log file utilities are executed by the workstation 10. Alternatively, workstation 30 or any other processing system may contain the log files and execute the log file utilities.

VII. CHANGING THE QUALITY OF THE ENCODED VIDEO

A. Changing the Quality Over Periods of Time

[0093] The encoded video produced by the encoding system is intended to be written to a final digital storage medium such as an optical disc. The encoded video on the disc is encoded at a Variable Bit Rate (VBR) which affords the possibility of large changes in bit rates among the different sections of video, as compared to Constant Bit Rate (CBR) operation. As the VBR coding allows a significantly different number of bits to be assigned to different pictures, the picture quality can be varied over time. Further, VBR affords the possibility of reassigning data rates to some user-defined sections of video in a scale not possible with CBR operation. Extremely large buffers would be required at the decoder to accomplish the same with CBR.

[0094] The encoding system allows what shall be referred to as manual editing but the manual editing is not the shortening or lengthening of scenes or substituting different scenes in a movie but is the changing of the picture quality. The picture quality may be changed over time and is referred to as temporal editing or temporal manual editing as will be discussed in this section. Another section will discuss the changing of picture quality within frames of data by shifting bits from one section of a frame to another section of a frame and will be referred to as spatial editing or spatial manual editing.

[0095] Three major constraints need to be satisfied in temporal editing or the re-allocation of bits among the various sections of an edit segment. These are:

1. The total capacity for the movie/video that is edited does not exceed the allocated capacity;
2. There is no buffer underflow within the edit segment; and

3. There is no buffer underflow outside of the edit segment caused by the temporal editing.

[0096] Constraint 1 is handled by re-encoding the edit segment such that resulting edited segment consumes the same number of bits as the originally encoded segment. It is to be noted that an edit segment is the entire time period under consideration. Different periods within the edit segment are assigned qualities by a user and these periods of the edit segment are referred to simply as sections or time periods.

[0097] In order to determine if an underflow condition will exist, it is necessary to check the target bits using a model of the decoder. In VBR operation of the encoder, a modified version of the Video Buffer Verifier (VBV) model is used. This model is referred to as the simple modified model for VBR and may be used for a Digital Storage Medium (DSM). The VBV for VBR mode is known and defined in the MPEG-2 standard and may be specifically found at ISO/IEC 13818-2, Annex C, section C.3.2.

[0098] In the VBV model for the VBR, the buffer of the decoder may remain full but is restricted from underflowing. In addition, the buffer fills up at a constant rate, $R_{\max}$, until the buffer is full and then no bits are transmitted to the buffer until the buffer begins to empty. At every frame interval, the bits required to decode one frame are removed from the buffer. With this in mind, it can be seen that in a VBR mode, buffer underflow is caused by allocating too many bits to a series of pictures. As the bits of the pictures are removed from the buffer, the buffer becomes empty and underflows if large numbers of bits of high quality pictures are removed at a rate faster than the buffer can be filled.

[0099] To prevent underflow, the number of bits used to encode a picture is reduced. Reducing the number of bits for a picture reduces the rate at which bits are removed from the buffer while keeping the fill rate the same. Buffer overflow cannot occur for VBR because if the VBV occupancy is full, then the bitstream transmission to the VBV buffer is immediately stopped. This means that there is no buffer overflow but simply a buffer full situation. The simple modified VBV model for VBR, as explained in ISO/IEC 13818-2, Annex C, Section C.3.2 is as follows:

1. bit_rate field at sequence_header represents the max_bit_rate.
2. The VBV is initially empty. After filling the VBV buffer full (i.e., VBV buffer size) at the rate of max_bit_rate, the decoding process begins. The picture data is immediately removed and decoded like CBR's VBV model defined in the MPEG-2 International Standard.
3. The bitstream is placed into the VBV buffer at the rate of max_bit_rate until the VBV buffer is full. If the VBV buffer occupancy is full, then the bitstream transmission to the VBV is immediately stopped. After the data for the next picture is removed, the bitstream transmission immediately starts.
4. The VBV buffer shall not underflow.
5. Every vbv_delay field is set to FFFF.

[0100] Changing the quality of the video over periods of time requires adding or removing bits from frames of video. An explanation into the number of bits that may be added or taken away from a section of video will now be provided. The number of bits which may be added or taken away from a section of video depends on the effective bit rate of the section of video under consideration. The effective bit rate, R , is calculated as:

$$R = (\text{Total Allocated Bits} / \text{Number of Frames}) * \text{Frame_Rate} \quad (2)$$

Because the total number of allocated bits is directly proportional to the bit rate, referring to the number of allocated bits conveys the same meaning as the bit rate, except with different units. Assuming that a guaranteed minimum rate is applied to any section of video, say $R_{\min}$, then the maximum number of bits that may be taken away from section 'i' is:

$$T_i = (R_i - R_{\min}) * N_i / \text{Frame_Rate} \quad (3)$$

where N_i is the number of frames in section 'i', and R_i is the effective rate of the same section. The guaranteed minimum is required in order to assure some minimum quality level for the section. Similarly, the maximum number of bits that may be added to a given section 'j' is calculated as:

$$T_j = (R_{\max} - R_j) * N_j / \text{Frame_Rate} \quad (4)$$

where $R_{\max}$ is the maximum rate, or the rate at which the buffer fills up when not full and N_j is the number of frames of section j .

[0101] A description of the changing of the picture quality over time periods (temporal editing) will now be given with respect to Figure 12 and the Figures referenced therein. After starting, step 281 performs an automatic video encoding without any user intervention, as described above in sections II and III. It is to be noted that for the automatic encoding, there is a maximum and minimum rate of the generated data stream. The maximum bit rate is determined by the capabilities of the video decoder and $R_{\max}$ is, for example, 8 Mbits/s. The minimum bit rate may be specified by the user, depending on the minimum picture quality desired. A typical minimum bit rate $R_{\min}$ is 3 Mbits/s. It is desirable to set $R_{\min}$ to be different from and somewhat below the average encoding rate in order to allow portions of the video to have a lower bit rate and portions of the video to have a higher bit rate. If $R_{\min}$ is set slightly below the average encoding rate of the video, the resulting encoded video will have a fairly constant quality. Typically, if $R_{\max} = 8$ Mbits/s and $R_{\min}$ is approximately 3 Mbits/s, the average value of the video should be around $3\frac{1}{2}$ to 5 Mbits/s.

[0102] After the automatic video encoding is performed in step 281, it is necessary to determine a rate-quantizer function. The process of the invention determines how many bits are necessary to obtain a desired quality but when re-encoded, a quantizer value must be set in order to result in the desired bit rate. The rate-quantizer function may be calculated as in a conventional manner such as described in the articles "Bit Allocation for Dependent Quantization with Applications to Multiresolution and MPEG Video Disclosures" by Ramchandran et al from IEEE Transactions on Image Processing, Vol. 3, No. 5, September, 1994, pp. 533-545, "Bit Allocation for Dependent Quantization with Applications to MPEG Video Coders" by Ramchandran et al, IEEE ICASSP, 1993, pp. V-381-V-384, and "Joint Multi-channel Bit-Rate Control for Multiplexed MPEG Video using Rate-Distortion Curves," by Reininger, presented at a Princeton video conference, 1994.

[0103] Next, step 284 inputs the user quality settings for selected time periods. The user inputs priorities for different sections which are ultimately used to control the quantization of the pictures. The quantization controls the picture quality, also referred to as resolution, for lossy MPEG encodings. Figure 13 illustrates a user interface which is used in order to gather the information to change the quality of the video over periods of time. Initially, a user will enter the time period for the entire edit segment over which the quality is to be changed. The input of this information is not illustrated in the user interface of Figure 13 but is inputted before the user interface of Figure 13 is displayed. For the example illustrated in Figure 13, the time period for the edit segment which is to be changed may be, for example, time 0 minutes to time 5 minutes. After the user has described the total time period of the edit segment, then smaller time periods within the edit segment are assigned different priorities in the range of -5 through +5 including a priority P , indicating that the quality of this time period is protected and should not be altered. Priority "none" is synonymous with a priority of P . Of course, the encoding system is not limited to the above input scheme and other ranges or symbols may be used to input the user's desired priorities.

[0104] In addition to assigning priorities, the user will also enter the minimum bit rate. This bit rate may be different than the minimum bit rate used to encode the video during the automatic video encoding of step 281. Another feature of the user interface is that it allows the user to set a "background priority" which means that time periods within the selected time period which are not assigned a priority by the user have this background priority. For example, in Figure 13, if the background priority were assigned "1" the unspecified time period 00:03:20:01 through 00:03:59:99 would automatically be assigned a priority of "1" without the user having to manually assign the priority for this particular section.

[0105] On the bottom of the interface illustrated in Figure 13, the "insert" block is used to create a new time period within the edit segment, the "recall" block is used to alter a previously created time period, "update" is used to save the changes made to a time period, and "delete" is used to delete a time period.

[0106] It is important to remember that the ultimate digital storage medium for the encoded video such as an optical disc has a limited storage capacity which ideally will be filled to capacity if the automatically encoded data were written to the disc. Therefore, whenever the picture quality is increased and the bit rate for a particular time period is increased, there must be a section of the video which has its quality reduced in order to supply the necessary extra storage capacity to the time period which has its quality increased. Accordingly, it is not possible in the preferred embodiment of the invention to assign only one period a high picture quality without assigning another time period with a different picture quality because the bits necessary to increase the quality must be taken from another section of the video. The segment of video which has been edited therefore should have at least one section which has a priority which is different from the background section and not marked as a "protected" section in order for the temporal manual editing to properly operate. Additionally, the user selected priorities do not result in the same quality and bit rate every time they are selected but the quality of a particular section also depends on the assigned priorities for other sections and the lengths of the other sections, as will become apparent upon understanding the process to determine the modified number of bits for the pictures within the edit segment.

[0107] A feature utilized by the present encoding system is that during the automatic encoding process or at a later time, a viewer may review the encoded video. As the encoded video is being displayed to the viewer, the viewer may perform a function such as pressing the return key or "clicking" a graphical user interface on a "button" displayed on the

screen in order to store points or regions of interest of the video without the need to stop the video or write down the time code of interest. These points of interest can be later called-up by the user and studied closely without the need to review the entire movie. This provides a convenient and efficient manner of determining points within the encoded video which a user may desire to change. This function is accomplished by storing a time code corresponding to the position within the encoded video of when the point or region of interest occurred.

[0108] After the user inputs the desired quality settings in step 284, step 286 computes the modified bits for each frame based on the user's input. This process is described in detail with respect to Figures 14-16B. The general procedure for changing the bit rates of the time periods within the edit segment of video is illustrated in the flowchart of Figure 14. In order to compute the bit rate based on the user's input, step 302 maps the user's input which was in the range between -5 and +5 to the range between 10 and 0. The initial input of -5 corresponds to the mapped setting of 10 and is the minimum quality and the input of +5 corresponds to the mapped setting of 0 and is the maximum quality. Similarly, an input quality selection of 0 corresponds to a mapped setting of 5, etc. The purpose of the mapping performed in step 302 is that the number of bits allocated is proportional to e (2.71828...) raised to a power of the quantity - 1 multiplied by the mapped setting. In this case, e^{-10} is less than e^0 so that the number of bits for the smaller priorities settings is less than for the larger priorities. The process of determining target bits corresponding to priorities of -4 through +4 is explained in detail in Figures 16A and 16B.

[0109] Step 304 determines if there are any "protected" sections. A protected section is a section in which the quality of the picture should not be changed from the previous automatic encoding. If there are protected sections, flow proceeds to step 306 in which the number of bits which is allocated for the re-encoding of the protected sections is set to be equal to the previous number of bits resulting from the initial automatic encoding. In this manner, the picture quality of the protected sections will not change.

[0110] Step 308 determines if there are any sections which are set to the maximum or minimum qualities. The maximum quality corresponds to a user input of +5 and the minimum quality corresponds to the user input of -5. For the maximum and minimum picture qualities, it is desired to obtain encoded video which has the highest and lowest quality video, respectively, while still satisfying the video buffer constraints and not exceeding the storage capacity of the medium containing the encoded video. If there are sections which are set to the maximum or minimum qualities, flow proceeds to step 310 which computes the modified number of bits for the maximum or minimum cases. This process is shown in detail in Figure 15.

[0111] In Figure 15, the process of determining the number of bits allocated for the maximum and minimum cases begins in step 320 by allocating the minimum and maximum rate to sections of the video assigned to mapped priorities of 10 and 0, respectively. Step 302 of Figure 14 performed this mapping and the remainder of the description of the temporal editing, unless specifically stated otherwise, will refer to the mapped priorities. The maximum data rate, as previously described, is limited by the ability of the decoder and may be 8 Mbits/s and the minimum data rate is input as illustrated by the user interface in Figure 13 and may go below or higher than the minimum bit rate used during the autoencoding mode which typically may be 3 Mbits/s.

[0112] Step 322 then determines if there are any sections having a mapped priority of 1 through 9. This means that in addition to there being a maximum or minimum quality assigned, there are other intermediate qualities assigned by the user. If there are other mapped priorities, flow proceeds to step 324 which computes the average bit rate for all sections with mapped priorities between 1 and 9. If this average is not in a permissible range (between the maximum and minimum bit rates), flow proceeds to step 328. In step 328, the maximum rate, the rate providing the highest picture quality, is reduced when the average is less than the minimum rate. By reducing the maximum rate, more bits will be available for the non-maximum and non-minimum situations (mapped priorities of 1-9), thereby placing the average within or closer to the desired range. Alternatively, in step 328, if the average is greater than the maximum rate, the minimum data rate is increased, thus reducing the number of bits for the non-maximum and non-minimum situations (e.g. priorities of 1-9). After step 328, flow returns to 324 where the loop of steps 324, 326 and 328 is repeated until the average is in range. Additionally, the maximum and minimum bit rates are adjusted to ensure that the "Protected" sections receive the same bits as before the adjustment of quality.

[0113] Once the average is determined to be in range in step 326, flow proceeds to step 330 which assigns the calculated minimum and maximum rates to the sections having the mapped priorities of 10 and 0, respectively. The remaining bits are then distributed in step 332 to sections which are not protected nor maximum or minimum situations. The sections of video having the mapped ranges of 1-9 and the manner of changing the data rates are described later with respect to Figures 16A and 16B.

[0114] If step 322 determines that there are not any sections having mapped priorities of 1-9, meaning there are only maximum, minimum or protected sections, step 340 adjusts the minimum and maximum rates, as necessary, until the final allocation of bits is equal to the initial allocation of bits from the automatic encoding. There are two alternative manners of performing step 340. First, the maximum bit rate $R_{\max}$ is fixed, for example, at 8 Mbits/s. The total bits for the overall edit segment is known and therefore, $R_{\min}$, the minimum data rate is simply adjusted in order to achieve the same total number of bits as were obtained during the automatic encoding. If it is not possible to adjust $R_{\min}$ to achieve

the previous number of total bits, then the maximum data rate, $R_{\max}$ is adjusted in order to achieve the total number of bits as were obtained during the automatic encoding.

[0115] As an alternative, step 340 assigns the maximum quality section with a priority of 1 and the minimum quality section with a priority 9 and perform the bit allocation algorithm illustrated in Figures 16A and 16B to determine the new bit rates for the time periods of the edit segment. Step 342 then assigns the data rates calculated according to Figures 16A and 16B, which are explained below, to the sections having the lowest and highest assigned qualities and the process of Figure 15 ends.

[0116] After step 308 and step 310, if necessary, are performed, step 312 determines if any sections have a mapped priority of 1-9. If this is the case, flow proceeds to step 314 which computes the modified number of bits for the mapped priorities of 1-9 as described in detail in Figures 16A and 16B.

[0117] The first step in Figure 16A is to determine the size of a common bit pool to be used. Initially, all sections with a mapped priority of 1-9 which are not protected have a certain number of bits removed. A factor k_1 , called the redistribution depth, is used for this purpose and for example, is set to 0.4 which means that 40% of the previously allocated bits (i.e. from the automatic encoding run) are to be removed from every section which is not protected nor marked with the maximum or minimum quality.

[0118] Next, step 362 determines how much the user selected quality will affect the quality. A variable k_3 , called the dynamic range factor, is set and determines the quality change which will result from the priorities. Depending on how k_3 is set, a priority of 2, for example, may impose different quality changes such as a relative small increase in quality or a relatively large increase in quality. Typically, the dynamic range factor k_3 is set to 1.0.

[0119] To use the user selected priority on the redistribution of bits, an exponential relationship of the form $\exp(-\text{priority}/k_3)$ is used. The dynamic range factor, k_3 , in the exponent determines the shape of the exponential curve. The curve gets steeper for larger values of the dynamic range factor, effectively increasing the disparity in bits reallocated among the sections with varying priorities. This negative exponential means that as the absolute value of the mapped priority increases (lower priority sections), the bits allocated to that section decrease, i.e., mapped priority of one has the highest priority and increasing numbers up to nine have a lower priority.

[0120] Step 364 normalizes the user selected priorities. This means that the user selected priorities are used to determine the relative priority of each section. First, E_i is calculated.

$$E_i = \exp(-p_i/k_3) \quad (5)$$

where p_i is the mapped user selected priority and i is the section time period number within the edit segment, excluding the time periods which are protected or have a maximum or minimum quality. Next, the priority normalization factor k_2 is calculated such that

$$k_2 = \sum_{i=1}^N E_i \quad (6)$$

where N is the number of section. The priority normalization factor is needed to ensure that the sum of all bits reassigned according to the priorities always equals the available common pool of bits.

[0121] The relative priority of each section is then calculated as

$$R_i = E_i/k_2 \quad (7)$$

As R_i is the relative priority fraction for the i^{th} section, the summation of all R_i will be unity.

[0122] Step 366 then normalizes the sizes of the sections. This is necessary as the time periods with the assigned priorities have different sizes and the bits should be distributed in proportion to the size of the different sections. First, the fraction T_i of the number of frames in each time period over the total number of edited frames (frames with a nonprotected non-max or non-min priority) is determined according to

$$T_i = N_i / N \quad (8)$$

5 where N_i is the number of frames in the i^{th} time period, and N is the total number of frames in the edit segment; that is all frames which are being altered and assigned a priority which is not protected and not a maximum nor minimum priority. Then, the size normalization factor k_4 is calculated according to the equation

$$10 \quad k_4 = \sum_{i=1}^N (T_i * R_i) \quad (9)$$

where N is the number of sections. From step 366 in Figure 16A, flow proceeds to step 368 in Figure 16B.

15 **[0123]** In Figure 16B, step 368 determines the common pool of bits according to the equation

$$20 \quad C = \sum_{i=1}^N (k_1 * B_i) \quad (10)$$

where B_i is the total number of bits which were used during the automatic encoding procedure of the i^{th} section plus any bits distributed in step 332 of Figure 15. The summation is over 'N' sections.

25 **[0124]** Step 370 then determines the target number of bits allocated for each section according to the equation

$$F_i = C * T_i * R_i / k_4 + (1 - k_1) * B_i \quad (11)$$

30 The target number of bits for each section F_i is equal to a percentage of the common pool of bits plus the original number of bits (B_i) reduced by the bits given to the common pool of bits. The percentage of bits from the common pool of bits is based on the fraction of the total number of frames of each section (T_i), the relative priority of the section (R_i) and the size normalization factor (k_4) which is dependent on the priority of a section multiplied by its corresponding fraction of frames summed for each of the sections.

35 **[0125]** It is possible that the target number of bits allocated for each section as computed in step 370 is not between the maximum and minimum data rates. If this is the case, step 372 reduces the redistribution depth k_1 by 0.01 and flow returns to recalculate the common pool of bits in step 368 and to recalculate the target number of bits in step 370. The loop of steps 368, 370, 372 and 374 is performed until the targeted number of bits calculated in step 370 is between the maximum and minimum data rates.

40 **[0126]** Step 376 then computes the bit scale factor S_i and the scale factors for intra and non-intra frames SI_i and SN_i , respectively. Intra and non-intra frames require different number of bits to obtain the desired quality. First, the bit scale factor S_i is calculated according to the equation

$$45 \quad S_i = F_i / B_i \quad (12)$$

Next, the scale factors for the intra and the non-intra pictures are calculated. The intra scale factor SI_i is calculated according to the equation

$$50 \quad SI_i = S_i - (I_f * S_i') \quad (13)$$

55 Large intra frames can potentially cause buffer underflow and the intra factor I_f provides a safety factor by limiting the variation in the bit assignment of the intra pictures to 20% when $I_f = 0.8$ (i.e., SI_i is limited to the range $0.8 < SI_i < 1.2$). Note that $S_i' = S_i - 1.0$. The non-intra bit scale factor SN_i is computed as

$$SN_i = S_i - (-I_f * S_i' * BI_i / BN_i) \quad (14)$$

5 where BI_i and BN_i are the total intra and non-intra bits summed over the i^{th} section by picture type.

[0127] Finally, the number of bits to be used for each frame is determined using the original number of bits multiplied by the appropriate scale factor (either the intra or non-intra scale factor). The process of Figure 16B (and also Figure 14) then ends.

10 [0128] At this time, the modified number of bits for each frame has been calculated and step 286 of Figure 12 is complete. However, it is possible that the changed number of bits for the frames may cause buffer underflow or the occupancy of the number of bits at the trailing edge of the edit segment has changed so that subsequent frames outside of the edit segment will underflow. These situations are examined and corrected, if necessary, in step 288 of Figure 12 which is detailed in Figure 17.

15 [0129] Figure 17 illustrates the process to check and correct for buffer underflow and to check and correct for possible problems due to bit reallocations at the trailing edge of the edit segment. In Figure 17, step 390 first scans the bit allocation for each of the frames of the edit segment. Step 392 then checks for underflow for the new bit distribution using the VBV model described above. A list of points is created where underflow problems exist (u_flow list) and the number of bits necessary to eliminate each underflow is determined and stored in u_val . It is important to keep in mind that underflow occurs due to a picture being represented by too many bits. When a high quality picture represented by a large number of bits is read out of the buffer by the decoder, a large number of bits is removed from the buffer when the picture is decoded and these bits cannot be replenished fast enough. Therefore, to eliminate underflow, bits are removed from pictures. Additionally, step 392 creates a list of points where the buffer is full and stores these points in o_flow list. Note that as explained above, there is no problem with a full buffer as when the buffer becomes full, data is no longer transmitted to the buffer.

25 [0130] Step 394 examines the buffer occupancy at the trailing edge of the edit segment. In the VBR mode of operation, it is possible for the buffer occupancy at the trailing edge of the edited video segment to be different from that of the run that was modified, even if the edited segment is replaced by exactly the same number of bits. This arises from the idiosyncrasy of the VBV model for VBR. Hence it is important to examine buffer occupancy constraints. It is fairly straightforward to show that no problem exists if the buffer occupancy at the trailing edge of an edit segment is exactly the same or higher as it was before the redistribution of bits. If the buffer occupancy is exactly the same as before, then that of subsequent frames also remains exactly the same as the previous run, which was already checked for VBV violations. If the occupancy is higher than before, there is the possibility of the buffer becoming full during subsequent frames, and that is a perfectly allowable condition as when the buffer is full, data will no longer be fed to the buffer and underflow does not occur. However, a third case arises when the buffer occupancy is lower at the trailing edge, compared to the previous run. This has the potential for creating buffer underflow in subsequent frames and needs to be examined closely.

40 [0131] There are two manners in which the last frame of the edit segment can be treated for preventing buffer underflow in frames outside of the buffer segment. The first approach is the easier of the two. If the number of bits of the last frame of the edit segment results in a buffer occupancy which is greater than or equal to the original encoding (stored in the log files), no underflow problem exists and therefore, no action needs to be taken. If the number of bits in the buffer for the last frame of the edit segment is less than the number of bits in the buffer for the originally encoded segment, this reduced number of bits may or may not cause underflow for frames outside of the edit segment, depending on the content of the frames after the edit segment. Instead of determining the number of bits needed to prevent underflow from occurring, the occupancy of the buffer for the last frame of the edit segment is simply adjusted to be the same as for the original encoding. For safety purposes, the buffer occupancy or fullness for the end of the edit segment may be raised to 10% above the occupancy of the original encoding to assure that there is no chance of underflow. If the buffer occupancy of the last frame of the edit segment needs to be raised to be at the level of the original encoding, the number of bits of the ending frames need to be reduced as performed in step 398. The last frame is treated as underflow point, is placed in u_flow list and the number of bits which need to be taken from the last frame to raise the buffer level up to the level of the original encoding are added to u_val .

50 [0132] A second approach for handling possible underflow caused by the re-encoding of the edit segment is an iterative process which estimates the minimum possible level of the buffer for the last frame of the edit segment which will not cause a buffer underflow condition outside of the edit frame. This is done by a process which estimates the greatest possible difference between the buffer occupancy of the original encoding and the buffer occupancy of the re-encoding for the last frame of the edit segment.

55 [0133] Turning to the flowchart illustrate in Figure 18, the buffer occupancy of the original encoding is assigned B_0 and has been stored in the log files of the original encoding. The buffer occupancy at frame 'i' beyond the boundary of the edit segment is B_i , where $i = 1...n$, where 'n' corresponds to the last frame in the movie to be encoded. How much

lower than the original encoding the buffer occupancy for re-encoding may go without creating underflow conditions for any of the successive frames must be determined as opposed to the first approach which simply sets the buffer level of the re-encoding to at least the point it was for the original encoding. Let the new lower buffer occupancy at the boundary that satisfies this condition be B_0' . This is the minimum allowable buffer occupancy at the edit segment boundary. The possibility of having a buffer occupancy lower than that of the prior run arises as a result of the fact that the difference may be absorbed in later frame intervals after the edit segment and therefore, underflow will not occur.

[0134] Beginning in step 420 of Figure 18, assume that the buffer occupancy, B_0' , of the re-encoded video of the last frame of the edit segment is zero without creating underflow in the future. This is the most liberal estimate for B_0' as it is possible that the buffer occupancy must be higher to prevent buffer underflow but the buffer occupancy cannot be lower than zero. However, the process of Figure 18 will check to see if the occupancy of zero for the last frame of the re-encoded edit segment creates an underflow and if it does, will increase the buffer occupancy at the end of the edit segment until a value of B_0' is found which will not create underflow.

[0135] The process of Figure 18 evaluates the estimated buffer occupancy difference, X' , between the original encoding and the re-encoding.

$$X' = B_0 - B_0' \quad (15)$$

The difference X' is called an estimate because the difference cannot be finally determined until it is assured that X' has been checked and adjusted as necessary until all chances of underflow have been eliminated. The procedure of Figure 18 steps through one frame at a time, starting with the frame immediately following the boundary frame, i.e. $i = 1$. For the first frame, with B_0' equal to zero, the estimated difference is simply the value of the buffer occupancy for the first frame of the original encoding, as calculated in step 422.

$$X' = B_0 \quad (16)$$

[0136] Next, step 424 determines the number of bits absorbed by the frames to the right of the boundary frame. The number of bits absorbed is a phrase used to refer to bits which could have been placed into the buffer by frames after the edit segment of the original encoding but in fact were not placed into the buffer during the original encoding because the buffer was full and therefore could not accept any more bits. The quantity Δt_i represents the time period of the i^{th} frame of the original encoding after the edit segment during which the buffer was full, and due to the reduced value of B_0' , can now accept additional bits. The time period Δt_i must be less than the time period of a frame as the buffer cannot be full for an entire frame and is greater than or equal to zero. The time period Δt_i is zero if the buffer was never full during the original encoding. Therefore,

$$1/P > \Delta t_i \geq 0 \quad (17)$$

where 'P' is the picture rate measured in frames per second and is typically 30 frames per second. The total number of bits absorbed to the right of the edit segment as a result of the buffer being full is calculated to be

$$a_i = \sum_{k=1}^i \Delta t_k R_{\max} \quad (18)$$

where $R_{\max}$ is the fill rate of the buffer.

[0137] Step 426 determines if the procedure for determining the buffer occupancy level of the last frame of the edit segment can end. If the sum of the bits absorbed by the i^{th} frame exceeds X' , i.e. if $a_i > X'$, then the procedure is stopped because the buffer is able to be filled by later frames so buffer underflow will not occur and step 428 is performed. Alternatively, if the last frame n is reached ($i=n$), the estimated value of X' will not cause an underflow problem up to the end of the video.

[0138] If the answer to step 426 is "No", step 430 calculates the number of bits, b_i , remaining to be absorbed

$$b_i = X' - a_i \quad (19)$$

5 Step 432 then determines if $b_i > B_i$, where B_i is the buffer occupancy for the i^{th} frame after the edit segment. If $b_i > B_i$, the estimated value of the difference, X' , is too big and will cause underflow because more bits remain to be absorbed than the current buffer occupancy ' B_i ' allows. Therefore, among the bits that remain to be absorbed, only ' B_i ' bits may be absorbed without causing underflow. Next, the estimate X' of the difference needs to be revised in step 434. In step 434, the revised estimate of X' is calculated as the bits remaining to be absorbed plus the bits absorbed

$$X' = B_i + a_i \quad (20)$$

15 If step 432 determines that $b_i \leq B_i$, there is no underflow problem because the buffer occupancy at the i^{th} frame is larger than the bits that remain to be absorbed, there is no need to change the estimate of X' , and flow proceeds to step 436. In step 436, the next frame is examined (i is incremented by 1) and the loop of steps 424, 426, 430, 432, 434, and 436 is repeated until the determination in step 426 has an affirmative answer. If step 426 has an affirmative answer, the estimate X' is acceptable and therefore in step 428, the minimum allowed buffer occupancy B_0' at the boundary of the edit segment is

$$B_0' = B_0 - X' \quad (21)$$

25 If the actual buffer occupancy of the re-encoded last frame of the edit segment is lower than B_0' , an underflow situation will occur due to the re-allocation of bits. In this case, the last frame is marked as an underflow point in u_flow list, and the number of bits necessary to be taken from the last frame of the edit segment to prevent the underflow condition (the difference between the number of bits for the changed video and the number of bits B_0' which result in the minimum allowable buffer occupancy) are placed in u_val in a similar manner as described for step 392.

30 **[0139]** It is to be noted that for the encoding system of the present invention, it is desirable to know the buffer occupancy for all points in the video and this information is stored in the log files. However, if the buffer occupancy at the last frame of the edit segment is reduced, the buffer occupancy for frames after the edit segment could be changed, making the buffer occupancy information in the log files inaccurate and therefore, corrective action would need to be taken which re-calculates the buffer occupancy for frames after the edit segment.

35 **[0140]** Referring back to the process in Figure 17, step 396 then determines if there are any entries in u_flow list. If there are none, there is no underflow problems and the process of Figure 17 ends. If there are entries in u_flow list, flow proceeds to step 398 which marks sections to the left of each position in u_flow list up to the nearest o_flow point as being protected and bits equal to the corresponding number of bits in u_val are removed from these sections. As buffer underflow is caused by too many bits in one or more frames, reducing the number of bits in the frames will solve the underflow problem. In order not to have a sharp reduction in picture quality due to the bits being removed to prevent underflow, the bits are taken not from just the frame at the underflow point but equally from all frames back to the last point that the buffer was full. This provides a manner for obtaining the highest quality video while still preventing underflow.

40 **[0141]** These bits which are removed in step 398 are then equally distributed to the sections which are not marked protected in step 400 and the process of Figure 17 begins again. The distributing of bits in step 400 does not guarantee that underflow will no longer be a problem and for this reason, it is necessary to again check for the underflow conditions using the entire process of Figure 17 in an iterative manner.

45 **[0142]** After the process of Figure 17 is complete, step 288 of Figure 12 is complete and step 290 in Figure 12 is executed. Step 290 determines the average quantizer value for each frame using the rate-quantizer function determined in step 282. The number of desired bits has previously been computed in step 286 and checked to see if underflow problems exist in step 288 but in order to re-encode the data so that the encoded data has the desired number of bits, the quantizer value must be determined. This is simply determined using the rate-quantizer function and determining the quantizer value by looking up the rate.

50 **[0143]** To determine the quantization level on a picture-by-picture basis, a rate-quantizer function (RQF) was determined in step 282. To estimate this function, two data points for each picture are required. The RQF is:

$$R = X * Q^{(-g)} + H \quad (22)$$

5 where R is the number of bits used to code one picture, Q is the quantizer scale, X, g, and H are the picture by picture basis coding characteristic parameters. H represents the bits which are independent of the quantizer scale so H is the bits consumed by the header, the motion vectors, the DC coefficients, etc. X and g are estimated from a pair of data points corresponding to the AC coefficient bits used to code the picture for a given quantizer scale.

10 **[0144]** The first run of the encoder may be used to generate the two data points for each coded picture in the video sequence. This may be done by a pair of alternating quantizer scale values used for each macroblock in the picture. These are processed to compute the rate-quantizer model parameters 'X' and 'g'. These two parameters, along with the value of H may be stored in a file, one per frame of coded video.

15 **[0145]** In order to determine the quantizer scale which will produce the target bits assigned to a particular coded frame of video, the above equation is used with 'R' representing the target bits, and the values of (X, g, H) are read from the stored data file which may be the log file described above. The value of 'Q' that best fits this equation is the desired quantizer scale.

[0146] Then, step 292 re-encodes the changed sections of video using the quantizer values determined in step 290. It is to be noted that re-encoding segments of previously encoded video may create encoding artifact problems unless special procedures are followed during the re-encoding. Details of the re-encoding process to avoid encoding and decoding artifacts are described in a separate section below. After the video is re-encoded, it is substituted into the bitstream for the previously encoded video, and the process ends.

20 **[0147]** As described with respect to Figure 5, the user may view the video after it is re-encoded to determine if it is acceptable. If it is not acceptable, the user does not have to accept the newly encoded video and may continue using the previously encoded video. If the user does find the newly encoded video desirable, the newly encoded video will replace the previously encoded video. The user may then edit a new segment of the video, either spatially or temporally, as illustrated in Figure 5.

25 **[0148]** Figures 19A and 19B illustrate two different examples of bit rates from an original video encoding and after the user has entered a manual quality priorities. In Figures 19A and 19B, the bit rate (at a scale of 10^6 which illustrate megabits) is plotted against a Group Of Pictures Number. In the illustrated examples, one GOP contains 15 pictures and each GOP is displayed per one half second. Also, the solid lines illustrate the original encoding and the dashed lines illustrate the encoding after the user has manually adjusted the quality of the picture.

30 **[0149]** Figure 19A illustrates an encoding having five different priority regions with priorities of 2, -3, 3, 0, and protected. The time period with the priority of 3 uses more bits after the re-encoding as compared to the original encoding. As the priorities are merely relative to each other, the section marked with a priority of 2 does not have a significantly increased bit rate whereas the sections of -3 and 0 have a reduced bit rate. The protected section should have a bit rate which is similar after the re-encoding as existed before the re-encoding.

35 **[0150]** Figure 19B is a second example having regions with user selected priorities of 0, 2, 0, 4, and 0 respectively. Again, as the priorities are relative to each other a priority of 0 does not mean that the bit rate is unchanged but is simply a priority relative to the other sections having priorities of 2 and 4. In order to give the section with the priority 4 a high bit rate, the three sections with the user selected priority of 0 are reduced and the section with a priority of 2 has approximately the same bit rate before and after the encoding.

40 **[0151]** The processing for the temporal manual editing occurs in workstation 30 of Figure 1A in the preferred embodiment of the invention. However, as the temporal manual editing does not occur during the video encoding process, the workstation 10 may perform the temporal manual editing calculations without degrading the speed of the video encoding process.

45 **[0152]** It is to be noted that the present invention is described in detail as changing the picture quality of various time segments within an edit segment. Of course, it is clearly possible to extend this concept to allow the changing of the quality of sections of pictures which are not in the same time segment. For example, it is possible to apply the teachings of the present invention to increase the quality of the last 5 minutes of a movie at the expense of the first 5 minutes. If it is desired to change the quality of time periods outside of a single edit segment, as described below, it would be necessary for the user to view and check out the segments of interest sequentially, reassign the bits, and ensure that the constraints relating to the total bits of the movie, as well as the buffer occupancy at the trailing edge of each section is within the prescribed limits. However, this may be accomplished by one of ordinary skill in the art without undue experimentation based on the teachings contained herein.

50 **[0153]** The temporal manual editing has been described as keeping the number of bits for the original encoding of the edit segment to be the same as the number of bits of the re-encoding of the edit segment. However, if it is desired, the re-encoded edit segment may consume less bits or if sufficient storage capacity exists, the number of bits for the re-encoded edit segment may exceed the number of bits of the original encoding. Also, it is possible to perform the original

encoding of the video so that it will not consume all available storage space on the final digital storage medium. Therefore, the total number of bits after re-encoding may consume any amount more or fewer than the original, for example between 0% and 20% more or less which includes 5% and 1% more or less bits.

[0154] The explanation of the temporal manual editing has been made with respect to changing the video after it has been encoded. However, the teachings contained herein may be applied to systems in which there was not an initial encoding process. Additionally, the teachings of the temporal manual editing pertaining to the use of a common bit pool may be applied to the concept of spatial manual editing for video frames, described below. Also, the manner in which the spatial manual editing operates on individual frames, as described below, may be applied to a series of frames in order to achieve temporal editing of quality.

B. Changing the Quality Within Regions of Frames

[0155] Figure 20 illustrates the general procedure for altering the quality of regions within one or more frames. After starting, step 450 performs automatic video encoding which encodes the input video into a compressed digital format such as an MPEG format or other format, as described above in the sections pertaining to the system components and general system operation. Step 452 displays the encoded video to a user and step 454 allows the user to enter commands indicating quality changes for regions within one or more frames.

[0156] Figure 21 illustrates a frame of video which is displayed on a monitor such as the video display monitor 61 illustrated in Figure 1. The frame of video in Figure 21 is illustrated as having a tree 465, a person 467, a bird 468 and two clouds 471.

[0157] In Figure 21, regions within the frame have been defined including region 466 surrounding the tree 465, region 469 surrounding the person, and region 470, overlapping region 469 and including the bird 468 and the head of the person 467. These regions were drawn using the graphical pointing device 33 illustrated in Figure 1A. Quality priorities set by the user in Figure 21 may include -5 for region 466 containing the tree, +2 for region 469 containing the person, and +4 for region 470 containing the bird and the person's head. The other regions of the frame of video in Figure 21 do not contain any user defined priorities and are therefore assigned a "Don't Care" priority. As will be explained later, the "Don't Care" regions are the regions first modified after changing the quantizer levels for the user defined regions in order to adjust the total number of bits for the encoding of the frame containing the user defined priorities to equal the number of bits resulting from the original encoding of the frame. If the user did not want the quality of regions of the frame to change, these regions would be marked by the user as having a priority of zero.

[0158] In the preferred embodiment of the invention, when defining priority regions within a frame of video, the user initially creates rectangular regions on a displayed frame of video using a graphical pointing device. The later defined regions are placed on top of and may overlap earlier defined regions. Even though the present invention is discussed with respect to defining rectangular regions, of course the teachings of the invention can be applied to any shaped region including regions having curves such as circles or ellipses, polygonal regions such as an octagon, or hexagon, or any other user defined shape including curves and/or straight lines. After the user defines each region, the user defines the priority for the region. Alternatively, the user can define the shape of all regions and thereafter, assign the priorities for these regions.

[0159] When the user initially defines priorities within a region, the regions correspond to pixel locations. However, the process of assigning priorities described below operates on a macroblock basis, a macroblock being a unit used in digital video encoding. Accordingly, if a macroblock is within or on the outer peripheral of a user defined region, that macroblock is assigned the priority of the region. One of ordinary skill in the art can simply determine the macroblocks corresponding to the user defined region using simple mathematics. For the outer peripheral of a user defined region, as an alternative to simply including a macroblock in a user defined region if the outer periphery of the region intersects the macroblock, a more sophisticated process of determining whether the macroblock should be included within the region may be performed by including the macroblock within the user defined region if more than 50% of the macroblock lies within the user defined area and excluding the macroblock from the user defined region is less than 50% of the macroblock is within the user defined region.

[0160] After the regions and their priorities are defined, the user may change the overlapping of the regions, if desired. For example, the user can "click" on region 469 of Figure 21. while holding a predetermined function key and change region 469 to be on top of the region 470 which would make the entire person have the priority of region 469 and only the bird 468 would have the priority assigned to the region 470. The overlapping regions are given a stacking priority when entered by the user. This stacking priority is adjusted as necessary when user defined regions are added, deleted, or modified.

[0161] After the regions are defined, the user also may reshape the regions as desired or move the regions using the graphical pointing device 33. Also, the regions which have been defined by the user may be deleted using the graphical pointing device 33. It is to be noted that even though the regions are being deleted, only the user defined priorities are being deleted but the original video data within the region being deleted is of course not deleted.

[0162] As explained in the above section on changing the quality over periods of time, during or after the automatic encoding, the user may review the encoded video and whenever a frame or time period of interest occurs, the user can simply press a key on a keyboard or enter input from another device indicating that the particular frame or time period is of interest. Later, the user can go back to that frame or time period and spend more time studying the time period and altering features within that time period, as desired.

[0163] The process of a user defining regions within a frame of video has been explained above for a simple case in which there is only one frame of data being modified by a user. However, it will be quite tedious to work on frames of data in a video sequence one frame at a time and therefore, the invention allows a user to define a region at a first time period and a corresponding region in a later time period. Subsequently, an interpolation process is performed so that all frames of video intermediate to the first frame and last frame automatically have regions created which correspond to the regions defined by the user for the first and last frames, and are assigned the same priorities as the first region and/or last region or a priority based on the priorities of the first and last frames. Additionally, the stacking priorities of overlapping regions are maintained during the interpolation process. After the intermediate frames have their regions automatically created, the user may then alter the automatically created regions by changing the priorities, the shapes of the regions, add new regions, or delete the regions, as desired.

[0164] Below is an example of how the interpolation process operates. Let the user select priority regions for two frames at time instances t_1 and t_n . After the user assigns a region in the frame t_1 and a corresponding region in t_n , the coordinates of these two regions are interpolated over intermediate frames to obtain region specifications for all the frames between t_1 and t_n . Not only can the interpolation be performed for the actual region being defined but the interpolation can be performed for the quality priority as well. For the interpolation process, a simple linear function is used. Suppose a user has defined a region A of frame t_1 having a point a_1 and there is a corresponding point a_n in the corresponding region A in the frame t_n , the corresponding points in the intermediate frames $a_2, a_3, \dots, a_{(n-1)}$ have a horizontal ordinate defined as

$$a_i(h) = a_1(h) + (a_n(h) - a_1(h)) \times (i-1)/(n-1) \quad (23)$$

and a vertical ordinate,

$$a_i(v) = a_1(v) + (a_n(v) - a_1(v)) \times (i-1)/(n-1) \quad (24)$$

where h and v correspond to horizontal and vertical ordinates of the point in question. In this process, only the vertices of a polygon that define the region are used and the vertices are interpolated. The regions are then defined by the vertices. Other interpolation techniques besides linear interpolation may be performed.

[0165] The above interpolation process can be performed as many times as required until all regions for all frames in the segment to be edited are properly defined. Furthermore, this process can be recursive with a fine-tuning process occurring such that a user may move or alter the size of a region determined through interpolation for an intermediate frame and the interpolation process may be executed again for the first frame and intermediate frame which has been altered, and then for the intermediate frame and the last frame. The interpolation process is primarily used for defining regions that move across frames. However, the invention also includes defining regions once which are stationary and used in a series of frames.

[0166] After the regions are defined and the user defined priorities are entered, the new quantizer values for each macroblock are estimated or determined in step 456 of Figure 20 based on the priority input by the user and the old quantizer value of the macroblock. This is determined according to the relationship illustrated in Figure 22 which illustrates the fraction of the original quantizer value used with each of the user selected priorities. For example, if the user inputs a priority of zero, the resulting fraction of the original quantizer value which is used is 1, meaning that there is no change to the original quantizer value. If the user selects a priority of -3, the fraction of the original quantizer value is 1.58, meaning that the original quantizer value for the macroblock having the priority of 3 is multiplied by 1.58 in order to determine the resulting quantizer value. As the priority -3 means reduced quality, the quantizer level must be increased to achieve a reduced quality meaning less bits are to be used for the macroblock. In an opposite manner, if a positive priority such as 3 is input by the user, the quantizer value of a macroblock from the original automatic encoding is multiplied by 0.42 which reduces the quantizer level. A reduced quantizer level means that more bits are required to represent the image within the macroblock and accordingly, the picture quality is increased. The points on the line illustrated in Figure 22 are (-5, 100), (-4, 1.75), (-3, 1.58), (-2, 1.42), (-1, 1.25), (0, 1), (1, 0.72), (2, 0.58), (3, 0.42), (4, 0.25), and (5, 0.01). The mapping illustrated in Figure 22 is based on a piece-wise linear relationship between the q -

levels and the priority. The points determined in Figure 22 have been determined through experimentation and different fractions may be used with the user selected priorities, as desired.

[0167] The priorities of -5 and 5 are intended to obtain the lowest and highest possible qualities, respectively. The lowest quality results from a maximum permissible quantizer level of 112 when the non-linear quantizer scale is used and 62 when the linear quantizer scale is used for MPEG-2 encoding. Also, the lowest quantizer level used to achieve the highest quality is 1. Therefore, the relationship illustrated in Figure 22 may result in values which are off the actual allowed q scale but these calculated q values are simply then placed at their saturation level. As an example, assume that initially, a macroblock has a quantizer scale of 20 and the user selects a priority for the macroblock to be -5. The quantizer value 20 multiplied by 100 yields a new quantizer level of 2,000 which is off of the quantizer scale and therefore, the maximum or saturation quantizer value is used.

[0168] In addition to using saturation levels as described above, the present invention takes a precaution for the I, P, and B frames of MPEG encoding. The present invention limits the highest quantizer level for the I-type frame to be 96 for the non-linear quantizer cases and 58 when the linear quantizer scale is used. Similarly, the highest quantizer level for the B-type and P-type frames is 112 for the non-linear quantizer scale and 62 for the linear quantizer scale cases. The linear and non-linear quantizer scales are defined in the MPEG standard. The lowest quantizer level used for each of the I, P, and B frame types is 1. Of course, the user defined inputs and the fraction of the original quantizer values illustrated in Figure 22 can be modified in order to satisfy the user's requirements.

[0169] After step 456 computes the new quantizer values for each macroblock, step 458 determines the number of bits for each frame resulting from the quantizer values determined in step 456. This step is important because in the preferred embodiment of the invention, the same number of bits for each frame must be obtained after the quality of some regions is increased while the quality of other regions are reduced. In order to determine the number of bits resulting from the new quantizer values, it is necessary to have a function which properly estimates the quantizer level versus the resulting number of bits for each macroblock. To determine the relationship between the resulting number of bits versus the quantizer level, a study of the empirical results of the video encoder, such as an MPEG-2 encoder, was performed and the determined relationship is based on actual empirical results. Simulation results for various quantizer levels and frame type using a number of video sequences were conducted. Based on these results, the following relationship between the quantizer level (q) for a macroblock and the number of bits (b) used in encoding that macroblock, depending on the frame type was determined to be

$$\log(b) = m \log(q) + n \quad (25)$$

where $m = -0.75$ for I-type frames and $m = -1.0$ for B and P-type frames. Additionally, $n = 15, 14.5$ and 13.8 for I, P, and B frames, respectively. These values are, as noted before, averaged over a number of video sequences. The above equation serves as a good approximation measure to predict the behavior of the encoder, and the relationship between the number of bits and the quantizer level is linear in the log domain. Of course, other relationships, if accurate, between the quantizer level and number of resulting bits may be used to determine the number of bits resulting from a given quantization level, including the other rate-quantizer functions described herein.

[0170] The above process describes how to determine the number of bits resulting from a given quantization level. However, if more than one encoding is used, it is possible to further refine the data rate versus quantizer function using the information from the new encoding. During the initial automatic encoding process, there is a first video encoding pass used to gather statistics of the video to be encoded. Then during the second pass which is the actual encoding, another actual point of the data rate versus quantizer function will be obtained such as in a b versus q plot corresponding to the above equation describing the number of bits b generated from a given q-level. After the data is re-encoded according to the user specified selections, as described below, the resulting re-encoded video is either accepted or rejected by the user depending on if it satisfies the desired quality improvements. Each re-encoding generates actual data for the q-level versus bits relationship which was estimated. Given the new q-level assignment, q', a determined data rate will be more accurate for this point based on the additional data obtained by the re-encoding. If the new quantizer level, q', falls between any two points which have been previously obtained, it is possible to do a linear interpolation in the log domain to obtain an estimate for the number of bits to be outputted by the encoder with the new q-level. If the new quantizer level, q', does not fall in between the two determined q-levels, the model can be used to extrapolate from the closest q-level to the new quantizer level, q', to estimate the number of bits to be generated at the encoding time. Note that the more times the data is re-encoded, the higher the probability of obtaining a better estimate will be for the output bits from the encoder.

[0171] After the number of bits for each frame resulting from the estimated quantizer values is determined in step 458 of Figure 20, step 460 determines if the estimated number of bits is close enough to the original number of bits for the frame and corrects the number of bits, if necessary. This is performed according to the process illustrated in Figures

23-25C.

[0172] When step 460 is performed, step 472 of Figure 23 first determines the difference between the estimated number of bits and the number of bits of the original encoding. This is calculated according to the equation

$$D = B' - B \quad (26)$$

where B' is the estimated number of bits of the modified frame, B is the number of bits of the frame resulting from the original encoding, and D is the difference. It is of course possible to use a value for B which is other than the number of bits resulting from the original encoding. For example, if a user wants to increase the total number of bits for a frame, he may set B to a number which is a predetermined amount higher than a number of bits resulting from the original encoding. Similarly, the user may set B to be lower than the original number of bits of the frame. Therefore, the re-encoded frame may consume more or less bits than the original encoding, depending on storage constraints, minimum quality constraints, and the possibility of video buffer underflow. Also, if regions in a frame have not had their quantizer values altered, it is possible to analyze only the numbers of bits resulting from regions having their quantizer values altered and ignoring the regions which have not been changed. Step 474 then analyzes this difference and if there are determined to be too many bits, flow proceeds to B for the process illustrated in Figures 24A-24C in order to increase the quantization level of the macroblocks to reduce the number of bits. If step 474 determines that there are too few bits in the modified frame, the quantization level of the macroblocks of the new frame is to be decreased in order to generate more bits according to process F illustrated in Figures 25A-25C. Otherwise, if the difference is within an acceptable threshold, there is no need to modify the quantization levels of the macroblocks and the process returns to perform step 462 in Figure 20. When keeping the number of bits in the frame the same for the re-encoded video, the bits can be considered to be shifted from one region to another.

[0173] In step 474, the difference D is analyzed by comparing the difference to a threshold, T . The threshold, is, set to $0.001 \times B$ which means that if the number of bits resulting from the change in quality of the region is within 0.001 of the total number of bits, then the difference is acceptable and the process may stop. Of course, the stopping threshold which indicates that the difference is acceptable may be determined in alternative manners, may be increased or reduced, as desired, or may be based on both the original number of bits for the frame and the resulting number of bits resulting from the user selected priorities.

[0174] The processes illustrated in Figures 24A-24C, and Figures 25A-25C are quite similar except that Figures 24A-24B are used to reduce the number of bits resulting from the user selected qualities and Figures 25A-25C are used to increase the number of bits resulting from the user selected qualities. A general overview of the processes performed in Figures 24A-25C is now described before explaining the details of these processes. Initially, the macroblocks in a frame of video are labelled as one of four types; "Don't Care", "Negative Priority", "Positive Priority", and "Don't Change". When a region has not been assigned a priority by a user, or if the system allows the user to assign a "Don't Care" status, that region is considered to be of the "Don't Care" type. If the user assigns a priority between and including -5 and -1 , the macroblocks within these regions are considered to be "Negative Priority". If a region is assigned a priority between and including 1 and 5 , those regions are considered to have a "Positive Priority". Last, all regions assigned with a priority of 0 , or if the system allows the user in some other manner to indicate that a region is not to have its quality changed, the macroblocks in those regions are considered to be of the "Don't Change" type.

[0175] Each of the macroblocks within each of the four macroblock types is labelled with an index, i , such that each of the four types has its own set of indexes. The index i of zero is assigned to the first macroblock and the index of $N-1$ is assigned to the last macroblock where N is the total number of macroblocks within a region type. For example, for all of the "Don't Care" regions, each of the macroblocks within these regions are indexed from 0 to $N-1$. Note that if there are more than two different "Don't Care" regions, there is not two different indices $i=0$ for the "Don't Change" regions. For all "Don't Care" regions, there is only one macroblock with an index 0 , one macroblock with an index 1 , etc.

[0176] If there are too many bits in the frame after the user has determined the priority of the regions, quantizer levels for the frame must be increased in order to reduce the number of bits for the frame. The original quantizer value for the macroblock is taken from the macroblock layer log file having a format as illustrated in either of Figures 10A or 10B. Alternatively, the encoder may be re-run to determine the quantizer values if they were not stored after the original run of the encoder. Yet another alternative is to decode the encoded bitstream to determine the quantizer values of the original encoding. The way the procedure works for decreasing the number of bits is to increase the quantizer level for each macroblock of the "Don't Care" type by one quantizer level, one macroblock at a time. If this does not solve the problem, each macroblock of the "Don't Care" has its q level again increased by one, one macroblock at a time. A predetermined maximum quantizer level is set and the quantizer levels of no macroblocks can be increased beyond this maximum level. If all macroblocks of the "Don't Care" type reach this maximum level, then the "Negative Priority" and the "Positive Priority" regions are increased by one in a similar manner. First, all negative priority macroblocks are

increased by one quantizer level and if this does not solve the bit differential problem, then the macroblocks of the "Positive Priority" macroblocks have their quantizer level increased by one. If this does not solve the problem, then the quantizer levels of the "Negative Priority" macroblocks are again increased by one and if this does not solve the problem, then the macroblocks of the "Positive Priority" type have their quantizer level increased by one. This process continues until all macroblocks in the "Negative Priority" and the "Positive Priority" types have their quantizer levels increased to the maximum level described above. If this does not solve the bit differential problem, then the macroblocks of the "Don't Change" type have their quantizer levels increased by one quantization level at a time until the bit differential problem is solved. In all likelihood, the bit differential problem will have been corrected before the quantization levels of the "Don't Change" type begin to be changed. A similar process occurs when the frame having the user selected priorities has too few bits and the quantization level of the macroblocks needs to be decreased in order to increase the number of bits according to the process of Figures 25A-25C.

[0177] Turning now to the flowchart illustrated in Figure 24A which is executed by Figure 23 when too many bits result, the process begins in Figure 24A by step 480 setting the block type to the "Don't Care" type. Step 482 then sets the macroblocks index to the starting point which means that the index, *i*, is set equal to zero. Step 486 is then performed which determines if the quantizer level of the macroblock in question (the macroblock within index=*i*) having the macroblock type which has been set (the first time through the flowchart, the first macroblock of the "Don't Care" type) has the maximum predefined quantizer level. The maximum quantizer level for this correction process should not be set too high so that there is a great decrease in quality for the macroblocks which are corrected. It is desirable to attempt to decrease the quality in order to decrease the number of bits in a uniform manner so that there is not a great differential in quality of the macroblocks, if possible. Therefore, this maximum is set at a quantization level of 62 when the linear quantization scale is used and 112 when the non-linear quantization scale is used for MPEG-2 encoding, for example. However, other values may be used for the maximum quantizer value. If the quantizer level of the macroblock in question is determined not to be the maximum in step 486, step 488 increments the quantizer level of the macroblock in question by one quantization level. Step 490 then recalculates the bit estimate for the frame using the bit rate-quantizer function described above. Step 490 then increments the index of the macroblocks. If the index of the macroblock indicates that the last macroblock of the macroblock type has not been processed, flow loops back to step 484. Otherwise, flow proceeds from step 492 to step 498 which resets the block index to zero, indicating the first macroblock.

[0178] When flow returns to step 484, the bit estimate recalculated in step 490 is used to recalculate the difference *D*, described above which is compared to a threshold, as described above. If the bit estimate is no longer too high, the number of bits for the frame containing the user selected priorities has been corrected and flow returns to the calling process. Otherwise, flow proceeds to step 486 which again determines if the quantizer value of the macroblock in question (having index *i*) is at a maximum.

[0179] If step 486 determines that the quantizer value of the macroblock in question is at a maximum, the quantizer level of that block is not to be increased and step 494 then increments the block index *i* by one. If step 496 determines that the index of the macroblock type is not past the last block, flow proceeds back to step 486 to determine if the block with the incremented index has a maximum quantizer value. Otherwise, if step 496 determines that the macroblock is the last macroblock of the macroblock type in question, step 498 is executed which resets the block index to the first macroblock (*i*=0). Flow then proceeds to process C illustrated in Figure 24B.

[0180] In Figure 24B, step 500 examines if the block type is of the "Don't Care" type. If it is the "Don't Care" type, step 502 determines if all "Don't Care" macroblocks have the maximum *q* level. If all of the "Don't Care" macroblocks do have the maximum *q* level, no further adjustment can be performed on the "Don't Care" type macroblocks and therefore the macroblock type is changed to "Negative Priority" in step 504 and flow proceeds back to process D illustrated in Figure 24A and step 484 is executed. Otherwise, if step 502 determines that all macroblocks of the "Don't Care" type do not have the maximum *q* level, flow proceeds back to process D without changing the type of macroblock being modified. This procedure continues until the resulting number of bits for the frame is within the threshold or all macroblocks of the "Don't Care" type are set to the maximum quantization level, which results in the block type being changed to "Negative Priority".

[0181] If step 500 determines that the block type is not "Don't Care", step 506 determines if the block type is "Negative Priority". If the block type is "Negative Priority", step 508 determines if all blocks of the "Positive Priority" have a maximum *q* level. If the result of the determination in step 508 is negative, step 510 is performed to set the block type to "Positive Priority" and flow proceeds to process D illustrated in Figure 24A.

[0182] If step 508 determines that all macroblocks of the "Positive Priority" have the maximum *q* level, step 512 examines if all blocks of the "Negative Priority" type have a maximum *q* level. If they do not, the block type is not changed and flow proceeds back to process D in Figure 24A. If all macroblocks of the "Negative Priority" type are determined to have the maximum *q* level in step 512, all "Don't Care", "Negative Priority", and "Positive Priority" macroblocks have the maximum quantization level and the block type is then set to "Don't Change" in step 514. Flow then proceeds to process D in Figure 24A.

[0183] Process E and step 520 of Figure 24C is performed after step 506 of Figure 24B has a negative determination.

Step 520 determines if the block type is "Positive Priority". If it does, step 522 determines if all blocks of the "Negative Priority" type have a maximum Q level. If they do not, the block type is set to "Negative Priority" and flow returns to Process D illustrated in Figure 24A. If step 522 determines that all blocks of the "Negative Priority" type do have the maximum q level, step 526 examines if all blocks of the "Positive Priority" type have the maximum q level. If they do not, flow returns to process D illustrated in Figure 24A. Otherwise, all "Negative Priority" and "Positive Priority" along with all "Don't Care" blocks have the maximum quantization level and step 528 sets the block type to "Don't Change" and flow returns to process D illustrated in Figure 24A.

[0184] If step 520 determines that the block type is not "Positive Priority", the block type must therefore be "Don't Change" and step 530 determines if all blocks of the "Don't Change" type have the maximum q level. If they do, there is an error as all block types have been set to the maximum quantization level without resolving the problem of too many bits being in the frame having the user selected priority. Setting all blocks to the maximum predefined q level should result in a number of bits which does not exceed the original encoding of the frame. If the determination in step 530 is negative, flow proceeds back to process D illustrated in Figure 24A.

[0185] If the process of Figure 23 determines that there are too few bits in the frame having the user selected priority regions, the process in Figures 25A-25C is performed. Figures 25A-25C are based on the process illustrated in Figures 24A-24C except that the quantization levels are decreased instead of increased in order to increase the number of bits for the frame and the quantization levels of the macroblocks cannot exceed a predetermined minimum level, such as a quantizer level of 1, when decreased. As the process of Figures 25A-25C is identical to the process of Figures 24A-24C, except for the above noted differences, a further description of Figures 25A-25C will not be made for the sake of brevity.

[0186] The correction process described in Figures 23-25C is one manner in which the invention may operate. However, alternative procedures of adjusting the quantization levels of the different types of macroblocks is possible. For example, instead of changing the quantization levels of the "Negative Priority" and the "Positive Priority" macroblocks at the same time as described above, the macroblocks of the "Negative Priority", "Positive Priority", and "Don't Change" types can be uniformly increased. Alternatively, if there are too many bits, the quantizer levels of the "Negative Priority" types may be increased until the maximum quantization level is reached and then the quantization levels for the "Positive Priority" type macroblocks are then increased. In an opposite manner, if the number of bits of the frame having the user selected priorities is too low, the "Positive Priority" type macroblocks may have their quantization levels decreased in order to increase the number of bits before the quantization levels of the "Negative Priority" type macroblocks are decreased. In the latter two cases, the altering of the "Don't Care" type macroblocks may occur before, during, or after the "Negative Priority" or "Positive Priority" type macroblocks are altered.

[0187] After the process of Figures 23A-25C is performed, flow returns to step 462 in Figure 20 which re-encodes the video using the new quantizer values. This re-encoding is performed while fine-tuning the quantization values using a rate-control function in order to have the resulting number of bits for the re-encoded frames be as close to the number of bits for the original encoding of the frames as close as possible. The rate-controller process is described in detail in a separate section below. Also, during the re-encoding process, encoding artifacts may result and therefore, special procedures may be performed in order to reduce these encoding artifacts, as described in a separate section below.

[0188] The process for the spatial manual editing occurs in workstation 30 of Figure 1A of the preferred embodiment of the invention. However, as the spatial manual editing does not occur at the same time as the video encoding process, the workstation 10 may perform the temporal manual editing calculations without degrading the speed of the video encoding process. It is to be noted that the present invention is described in detail as changing the picture quality of various regions within a frame. Of course, it is possible to extend the concept in the present application to shifting bits within a region of one frame to a region in a different frame. Additionally, the spatial manual editing may be applied in conjunction with the temporal manual editing described above.

[0189] The spatial manual editing has been described as keeping the number of bits of the frame having a user supplied regional priorities the same as the original encoding of the frame. However, if it is desired, the re-encoded edit segment may consume a predetermined number of bits which is more or less than the original number of bits for the frame, if sufficient storage capacity exists. In these cases, it will be desirable to know the number of bits by which the altered frame can be changed and therefore, the processes illustrated in Figures 24A-25C will operate until the set difference is achieved. Accordingly, the total number of bits for the re-encoding of any frame may consume any amount more or less than the original encoding, for example between 0% and 20% more or less which includes both 5% and 1% more or less bits.

[0190] The explanation of the spatial manual editing has been made with respect to changing the video after it has been encoded. However, the teachings contained herein may be applied to systems in which there was not an initial encoding process.

[0191] The manner in which the spatial manual editing works is to assign a quantizer value based on the user selected priority and then adjust the quantization levels of the macroblocks of a frame, as necessary, in order to achieve the same number of bits for the frame as previously obtained. In contrast, the temporal manual editing described in the section

above removes a common pool of bits from sections of the video and redistributes those bits from the common pool in accordance with the user selected priorities. All teachings relating to the process in which the spatial manual editing is performed with respect to one frame may be applied to the concept of changing the quality of frames of video over time, as described above in the temporal manual editing section, and vice versa. For example, the spatial editing has been described as modifying quantizer levels to achieve the changes in quality whereas bit numbers are adjusted in the temporal editing section. As the number of bits for a frame or macroblock directly depends on quantization levels, the modification of bit numbers and quantization levels achieves the same results and therefore the temporal and spatial editing may modify the number of bits and/or the quantization levels.

VIII. RE-ENCODING THE VIDEO AFTER CHANGING THE QUALITY

A. Bit Allocation Control When the Quality Has Been Changed Within Single Frames

[0192] To control the bit-rate in the video encoding device 50 illustrated in Figure 1A, a controller, such as workstation 10 or any specialized hardware that can perform the same controlling function, loads quantization levels, or q-levels, and an estimated or expected number of bits per slice into a rate-controlled encoder. The encoder then begins encoding, and quantization levels are adjusted after measuring the accuracy of the generated number of bits compared to the previously determined estimated number of bits. The estimated number of bits are determined using a rate-quantizer function, as described above in the sections pertaining to the spatial and temporal manual editing. This process is performed sequentially on a set of n macroblocks at a time, where each set is designated MB(i), and where:

$$1 \leq i \leq (\text{total \# of macroblocks} / n). \quad (27)$$

When re-encoding the last set of macroblocks, the rate-controller assures that the total number of generated bits in the re-encoding process is less than the originally encoded length since any minor overestimation in the number of bits that will be generated can be absorbed using bit-stuffing.

[0193] Figure 26 is a flowchart illustrating a method for re-encoding a set of digital video frames with pre-assigned quantization levels. Adjustments are made to the pre-assigned quantization levels based on how accurately a quantizer model estimated the number of bits that would be generated by re-encoding versus the actual number of generated bits. In Fig. 26, the rate-controlled coding process starts at step 700 and then designates, by setting i=1 in step 702, the first set of n macroblocks as the current set of macroblocks. Step 704 loads the complete set of estimated number of bits, EB(i), that will be used in re-encoding each set of macroblocks, MB(i). MB(i) is then re-encoded in step 706, and the number of actually generated bits is stored as GB(i). The percentage difference between the generated bits, GB(i), and estimated bits, EB(i), is calculated in step 708 as an estimation ratio, PB(i) = GB(i)/EB(i). Additionally, the system can calculate the cumulative estimated number of bits, EB'(i), and the cumulative generated number of bits, GB'(i), for all re-encoded sets of macroblocks, where EB'(1) = EB(1), GB'(1) = GB(1), EB'(i) = EB'(i-1) + EB(i) and GB'(i) = GB'(i-1) + GB(i), for i ≥ 2. Therefore, as an alternative to the illustrated step 708, PB(i) can be calculated using the cumulative estimated number of bits, EB'(i), and the cumulative generated number of bits, GB'(i), by using PB(i) = GB'(i)/EB'(i). This provides a more accurate indication of how the re-encoding process as a whole is meeting the target number of bits, instead of an indication of how accurately a particular set of macroblocks was encoded. In step 710, the estimation ratio is used to determine a correction factor for the q-levels in the next set of macroblocks, MB(i+1).

[0194] This correction factor is determined using one of two methods. The first correction factor determination method steps through a single sorted list of ratios in a table of ratio values, called a ratio lookup table, and determines between which two ratios the estimation ratio falls. The index, j, of the larger of the two numbers in the table is used as a correction factor index. The correction factor index is used to look up a correction factor, Δ(i), in a correction factor table by choosing the jth element in the table as the correction factor, and the next set of macroblocks, MB(i+1), are adjusted by Δ(i). The ratio lookup table, RT, contains, for example, the sorted values:

$$RT = \{ 0.85, 0.90, 0.95, 1.0, 1.05, 1.10, 1.15, 1.2, 1.5 \}$$

and the correction factor table, CT, contains the values:

$$CT = \{-3, -2, -1, 0, 1, 2, 3, 4, 9\}.$$

5 **[0195]** Figure 30 is a flowchart which illustrates a process for determining a correction factor index from a ratio lookup table using an estimation ratio. Assume, for purposes of illustration only, that $PB(i) = 0.87$. The index, j , is initially set to 1 in step 722. The process of comparing $PB(i)$ to $RT(j)$ is begun, and in step 724, $PB(i)=0.87$ is compared to $RT(1) = 0.85$, as defined in the above lookup table. Since 0.87 is greater than 0.85, the process continues to step 726. Since in step 726 it is determined that there are remaining indices, control is passed to step 728 and j is incremented by one. 10 Returning to step 724, $PB(i) = 0.87$ is now less than $RT(2) = 0.90$, so control is passed to step 730 which ends the search for the correction factor index.

[0196] Figure 31 is a flowchart which illustrates how to look up a value in a lookup table given an index into the table. In Figure 31, to determine the correction factor according to the first method, the index j , which is 2 is used to look up the corresponding entry in CT. Reading from the second position in CT, the correction factor is determined to be -2 in 15 step 736. Therefore, the q-levels of $MB(i+1)$ are changed by -2. Likewise, if $PB(i) = 1.12$, then index 7 corresponds to the first entry in RT that is larger than $PB(i)$, so the correction factor in CT at the seventh position is 3. Thus the q-levels of $MB(i+1)$ are changed by 3, e.g., by adding 3 to the q-level of $MB(i+1)$.

[0197] The second correction factor determination method uses a similar process to the first method, except the percentage of sets of macroblocks that have been re-encoded is used as a second parameter in determining the correction factor. Additionally, instead of reading a correction factor from a table, the correction factor is read from a matrix. By using the percentage of sets of macroblocks that have been re-encoded, or the percentage of sets of macroblocks unre-encoded, a more gradual correction can be made at the beginning of a series of frames and more significant changes can be made at the end, if necessary. This allows the system to correct more accurately for variations between the quantizer model and the actual number of generated bits. By using an inverse relation between the number of sets of 20 macroblocks which remain to be re-encoded and the significance of correction, and assuming that an overestimation in one set of macroblocks will be offset by an underestimation in another set of macroblocks, unnecessary corrections are avoided.

[0198] As an example of a correction factor matrix, CTM, of the second correction factor method, assume CTM is defined as below: 30

$$CTM = \begin{bmatrix} -1 & -1 & 0 & 0 & 0 & 1 & 1 & 2 & 4 \\ -2 & -1 & 0 & 0 & 0 & 1 & 1 & 2 & 5 \\ & & & & \cdot & & & & \\ & & & & \cdot & & & & \\ & & & & \cdot & & & & \\ -3 & -2 & -1 & 0 & 1 & 2 & 3 & 4 & 8 \\ -3 & -2 & -1 & 0 & 1 & 2 & 3 & 4 & 9 \end{bmatrix}$$

If at the beginning of a series of frames, the estimation ratio indicates an overestimation of the number of bits necessary, there is no need to immediately reallocate all the unused bits to the next slice. If the unused bits were immediately consumed by $MB(i+1)$, then an $MB(k)$, where $k \geq i+1$, which consumed more bits than estimated, would force $MB(k+1)$ 45 to a higher quantization level unnecessarily. The unused bits could have been carried forward in the re-encoding process from $MB(i)$ until needed and then used by $MB(k)$.

[0199] Figure 32 is a flowchart which illustrates how a correction factor is calculated using both an estimation ratio and the number of remaining unre-encoded sets of macroblocks. For a case of overestimation of the number of bits required, i.e., having previously determined that the correction factor, j , equals 2 for $PB(i)=0.87$, step 744 of Figure 32 determines that the overestimation occurs in the first percent of the re-encoding process. The element at (2,1) of CTM is -1 and is therefore chosen as the correction factor. This dampens the correction effect early in the re-encoding process, which in the first method would have been -2. However, if ninety-nine percent of the re-encoding process has been performed and the estimation ratio is $PB(i) = 1.12$, element (99, 7) which is 3 would be chosen, thus fully correcting at the end of the re-encoding process. 50

[0200] Referring back to Figures 27A - 27C, a frame with pre-assigned q-levels is illustrated as a further example where the number of macroblocks in a set is equal to the number of macroblocks in a single slice. $EB(i)$, the estimated number of bits, is calculated as described above and loaded into a memory used in the rate control process for $MB(i)$, macroblock i , of Figure 27A and then $MB(i)$ is actually re-encoded. The number of generated bits, $GB(i)$, is determined, 55

the estimation ratio is calculated, and the correction factor $\Delta(i)$ is looked up. As is shown in Figure 27B, the macroblocks of MB(i+1) are then corrected by $\Delta(i)$ and the process is repeated for MB(i+1). Figure 27C shows the change in MB(i+2) after the length of MB(i+1) has been estimated and calculated and its estimation ratio determined. This process is continued until all macroblocks and partial macroblocks have been re-encoded and corrected.

[0201] A further example is given in reference to Figures 27A and 27D. In this example, the number of macroblocks per set is the number of macroblocks in two slices. After calculating EB(i), GB(i) and PB(i), the macroblocks of MB(i+1), which comprise two slices, are corrected by $\Delta(i)$, as illustrated in Figure 27D. This re-encoding and correction process is continued as in the previous example until all macroblocks have been re-encoded.

[0202] The changes caused by the re-encoding process are illustrated in Figures 28A and 28B. Two equal size regions, Region A and Region B, of Frame N are initially encoded using a nearly equal number of bits. The user designates that the quality of Region A is to be decreased and the quality of Region B is to be increased. However, it is desired to have the number of bits used for the original encoding be as close to the number of bits of the re-encoding as possible. After re-encoding Frame N, Frame N contains the same number of bytes as were generated from the original encoding, but Region A is coded with fewer bits and Region B uses the additional bits. This decreases the quality of Region A and increases the quality of Region B as requested by the user. The rate control process of the present invention makes the re-encoded number of bits of the frame be as close to the original number of bits as possible by correcting for minor errors in the estimated number of bits determined using a rate-quantizer function.

[0203] Figure 29 illustrates changes to a digital video stream by reallocating bits from a plurality of frames and giving them to a second plurality of frames. Although individual frame sizes change, the total size of the set of frames should remain the same. The rate controller allows the number of bits for the re-encoding of the plurality of frames to be finely tuned to be the same as the number of bits from the original encoding, even if the rate-quantizer function used to determine the estimated number of bits is slightly in error.

B. Re-Encoding While Avoiding Decoding Artifacts At Edit Points

1. Restoring the Encoder to the State Used When Originally Encoding

[0204] The above-described temporal and spatial quality editing operations allow the quality of video to be changed by re-encoding the video with new quantizer values. However, simply substituting the re-encoded video into the original video may result in unacceptable decoding artifacts which appear as a visible glitch at the edit point of the video. These visible glitches will occur even if the new segment being substituted has the proper syntax such as MPEG-2 syntax and the buffer constraints across the edit boundary are satisfied. The problem results from the non-availability of the last reference frame (the last P-frame) preceding the edit segment, and a different reference frame (the last P-frame) at the end of the edit segment. This causes a disparity between the encoding and decoding of the first set of B-frames at the start of the edit segment, and in the first set of B-frames immediately following the edit segment.

[0205] In order to properly perform the edit of the digitally encoded and compressed video, it is first necessary to determine the proper positions in the encoded bitstream which correspond to the desired edit points. This process has been explained above in section VI which describes utilities to access the encoded video. As explained above, this is performed by summing up the number of bits used for each previously encoded frame in order to determine the bit offset for the picture in question. Alternatively, a directory could be used to maintain information indicating the exact location of a specific frame or time period in the encoded bitstream.

[0206] Turning now to Figures 33A-33E, there are illustrated time periods of video which are used as examples for explaining the operation of the invention. In Figure 33A, an original section of encoded video, O, is illustrated. It is desired to replace the encoded time period of 5m 0s to 10m 0s (five minutes, zero seconds to ten minutes, zero seconds) with a different section of video. Figure 33B illustrates a new section of unencoded video, N, which is to be substituted into the encoded video of Figure 33A. The dashed lines in Figures 33B and 33C are used to denote unencoded video and the solid lines in Figures 33A, 33D and 33E are used to denote the encoded video.

[0207] If only the section of new video illustrated in Figure 33B was encoded using a normal MPEG-2 mode in which a GOP referenced a previous GOP, and substituted into the video illustrated in Figure 33A, decoding artifacts would exist at the edit points 5m 0s and 10m 0s due to the lack of an accurate reference frame necessary for the bidirectional decoding used in MPEG-2 video compression. This problem exists at both the five minute and the ten minute points which are handled differently by the invention, due to the fact that the beginning of a GOP refers back to a previous GOP but the end of the GOP does not refer forward to the next GOP.

[0208] The manner in which decoding artifacts are prevented at the beginning of a GOP is to re-create the last P-frame of the preceding GOP which is used as a reference frame for the first two B-frames of the GOP. This is accomplished by placing, for example, one GOP of the original video at the beginning of the video to be inserted and this added video is illustrated from 4m 58.5s to 4m 59s in Figure 1. In the examples, one GOP is equal to one half of a second, although other time periods for the GOPs are possible. The encoder is re-run for this section with the same quantizer values used

for the original encoding to create the last P-frame of the GOP immediately before 4m 59s. All information resulting from the re-encoding of this short section of video, except for the last P-frame may be discarded.

[0209] In order to prevent decoding artifacts at the end of the edit segment, a one second period (e.g. from 10m 0s to 10m 1s in Figure 33C) of the original video is added to the end of the new video to be substituted and this one second segment is encoded using the same quantizer values as were used during the original encoding. If this one second period were not added to the end of the video, the two B-pictures immediately beyond the 10m 0s mark, assuming that 10m 0s is the dividing point between two GOPs, would refer back to a reference picture which is different after the edit than before the edit. While the examples have been described as adding one half second or one second periods of video to new the section, the concept can be generalized to adding GOPs.

[0210] Even though the decoding artifacts will no longer appear at the 10 minute mark, as the edit point at the end is now the 10m 1s point, one must consider if decoding artifacts will result at 10m 1s. Visible decoding artifacts will not result because the two B-frames immediately after 10m 1s mark will refer back to essentially the same reference frame to which they previously referred. Even though the two B-frames immediately after the 10m 0s mark will not refer back to the same reference frame to which they previously referred, the two B-frames were re-encoded and therefore properly refer back to a new reference frame. Accordingly, no visible decoding artifacts will be present at the 10m 0s or the 10m 1s points.

[0211] The two GOPs (one second) added to the end in Figure 33 should be sufficient time for the last P-frame to be nearly equivalent to the original encoding. If it could be guaranteed that the exact same quantizer values are used when re-encoding the original video after the 10 minute point, one GOP (one half second) should be sufficient to appropriately generate a correctly encoded reference picture. However, two GOPs are preferred to assure proper quality. Even though it is desired to use the same quantizer values as originally used, in practice there may be minor deviations from the original quantizer values and therefore, two GOPs are preferred.

[0212] In addition to original periods of the video being used from 4m 58.5s to 4m 59s and 10m 0s to 10m 1s, Figure 33C illustrates additional original video from 4m 59s to 5m 0s. This video is not essential to preventing decoding artifacts and therefore, the one half second to determine the last P-frame could be used immediately before the 5m 0s mark (e.g. from 4m 59.5s to 5m 0s). However, the 4m 59s to 5m 0s period is added to the new video to eliminate any chance of perception by the operator that the manual changing of quality has had an effect beyond the edit boundary. This misperception is due to the fact that the coding order for frames at the beginning of an MPEG GOP is IBB but the frames are displayed as BBI. The one second period is encoded using the same quantizer values as were used during the original encoding of that time period. The manner of obtaining the previously used quantizer values is explained below. If the one second period was not added from 4m 59s to 5m 0s, the substitute encoded video illustrated in Figure 33D would begin at 5m 0s.

[0213] The useful data resulting from the encoding of the video in Figure 33C is illustrated in Figure 33D as the substitute data, S, which runs from 4m 59s through 10m 1s. This substitute data, S, is then substituted into the original encoded data illustrated in Figure 33A which results in the final bitstream as illustrated in Figure 33E.

[0214] The unencoded and encoded information illustrated in Figure 33A is video in the preferred embodiment, but can be audio, or other types of information which use previous and/or subsequent information to decode the encoded information.

[0215] The new section of video in Figure 33B is obtained from any video source and may be video generated by the temporal or spatial editing process described above. The original video in Figure 33C from 4m 59s to 5m 0s and 10m 0s through 10m 1s is the original video having the same quantizer values used for the original encoding. This video at the ends of the new video N in Figure 33C is simply obtained using the temporal editing process and setting the quality of the end sections to protected as described in the temporal manual editing section, meaning that the quantization values are to remain the same. Alternatively, the video in Figure 33B may be completed unrelated to the video in Figure 33A and may have a different length. Additionally, it is possible to use the invention to cut out a section of video and not add new video. In this case, no video would exist in Figure 33B and the 'N' segment and its corresponding encoding would be absent from Figures 33C-33E. As long as the last P-frame before the edit segment is properly constructed and one half to one second (one or two GOPs) of the original video is added to the end of the new video to be encoded and encoded with the same quantizer scales used in the original encoding, no visible decoding artifacts will occur in any situation.

[0216] Figure 34 is a flowchart illustrating the process described with respect to Figures 33A-33E. After starting, step 780 encodes the video into the compressed format to generate the encoded video illustrated in Figure 33A. Step 782 determines the new section of video which is in an unencoded format which is to be substituted into the encoded video. This new section of video is illustrated as N in Figures 33B and 33C and may represent scenes related to, unrelated to, or have a different time period than the original frame. The start and end point of the video to be substituted are 5m 0s and 10m 0s, respectively. However, the actual start and end points of the final encoding to be substituted, may be different, as illustrated in Figure 33D.

[0217] Step 784 adds periods of unencoded video to the ends of the encoded new sections of video for the reasons

described above in order to prevent decoding artifacts at the edit points 5m 0s and 10m 0s. These added periods are from 4m 59s through 5m 0s and 10m 0s through 10m 1s, although the period from 4m 59s to 5m 0s is not essential.

[0218] Step 786 determines the P-frame immediately before the edit point. As the video being substituted in Figure 33E is from 4m 50s to 10m 1s, the last P-frame immediately before 4m 50s needs to be determined. This P-frame is determined by running the encoder for the time period from 4m 58.5s to 4m 50s with the quantizer values used to originally encode the P-frame. The quantizer values are determined by retrieving the original quantizer values from the macroblock log file having the format illustrated in either Figure 10A or 10B. However, the quantizer values for each macroblock of an entire movie consume a large amount of storage space and it may not be desirable to store the quantizer values. As an alternative, the quantizer values can simply be created using the rate control status pertaining to the video buffers for the I, P, and B frames, the target bits for the picture shown in the picture layer log file of Figure 9B as allocated bit, the activity level of the picture, stored as mean_activity in Figure 9C and the activity level of the macroblocks which are computed by the encoder during encoding. The rate control status is stored in the picture layer log file as illustrated at the bottom of Figure 9C as S1_i, S2_i, S1_p, S2_p, S1_b, and S2_b which stores the rate control status information for the I, P, and B video buffers respectively. One of ordinary skill in the video encoding art, based on the teachings contained herein, could determine the quantizer values for the desired P-frame without undue experimentation.

[0219] As an alternative manner of determining the quantizer values of the reference frame, and/or the encoded reference frame itself, the encoded version of the reference frame in the encoded reference stream can be decoded in a known manner. Further, any other method of obtaining the needed reference frame can be performed.

[0220] After the reference P-frame is obtained, the video is encoded in step 788 while maintaining the same quantizer levels as the original encodings of the end sections. The quantizer levels are obtained as described above. The resulting encoded video is illustrated in Figure 33D.

[0221] Step 790 then substitutes the newly encoded video into the originally encoded video and the final product, as illustrated in Figure 33E is obtained. This encoded video containing substituted encoded video has minimal decoding artifacts at the edit points.

2. Re-Encoding in a Closed Group of Pictures Mode

[0222] As an alternative to the procedure described in Figures 33A-34, it is possible to perform a different procedure for the encoding of the beginning of the new video while still reducing decoding artifacts. In this procedure, the manner of treating the trailing end of the video is the same as described above with respect to Figures 33A-34.

[0223] As an example of this alternative embodiment, use Figure 33A as the original encoded video and Figure 33B as the unencoded video to be substituted into Figure 33A. No original unencoded video is added at the beginning edge of Figure 33B, and as described for Figure 33C, two GOPs (one second) of corresponding original unencoded video are added to the end of Figure 33B and the video to be encoded is as illustrated in Figure 35A.

[0224] If the encoder encodes the video of Figure 35A in the closed GOP mode for the first GOP of the video while increasing the bits for the B-frames without a second reference picture, the visible decoding artifacts at the start edit point will be eliminated. The encoding of the end portion using the same quantizer levels as used for the original encoding from 10m 0s to 10m 1s is the same as described above with respect to Figures 33A-34, and the resulting encoded video is illustrated in Figure 35B. The closed GOP mode is described in the MPEG-2 standard at section 6.3.8. The closed Group Of Picture mode flag is set to 1 which indicates that encoded B-frames have been encoded using only backward prediction and the reference frame of the previous GOP is not used. However, as one of the reference frames normally used for the bidirectional frames is not available, the B-frames will have reduced quality, unless a procedure is performed to eliminate the reduced quality.

[0225] In order to solve the problem of reduced quality of B-frames encoded in the closed GOP mode, the number of bits for the B-frames having only one direction of prediction is increased so that the quality of the frame is not substantially reduced. As the B-frame has only one direction of prediction, the number of bits is preferably increased to approximately the number of bits used for ordinary P-frames. However, the number of bits may vary, depending on the desired picture quality.

[0226] The encoded video of Figure 35B can be directly substituted into the previously encoded video, as illustrated in Figure 35C.

[0227] A flowchart describing the process of using the closed GOP mode is illustrated in Figure 36. In Figure 36, the video is encoded into the compressed format in step 800 and the new section of video which is to be substituted into the encoded video is determined in step 802 in a similar manner as described with respect to steps 780 and 782. Next, step 804 adds GOPs as needed (one or two GOPs) of unencoded original video to the trailing end of the new section of video.

[0228] Step 786 then encodes the video. The first GOP of the new video is encoded in the closed Group Of Pictures mode and the number of bits for the bidirectionally predicted frames which are missing the reference frame is increased

to approximately the number of bits used for single direction predicted frames, P-frames. If the total number of bits for the encoded video is to stay the same, the extra bits needed for the bidirectional frame must be removed from other frames. This may be accomplished above as described in the sections on spatial and/or temporal manual editing. Next the remainder of the new section of video is normally encoded. Last the added period at the trailing end of the new video is encoded using the same quantizer values as were used during the original encoding of the added period to prevent decoding artifacts at the end point of the edit segment. Finally, the newly encoded video is substituted into the originally encoded video in step 808.

[0229] By either of the processes described above and illustrated in Figures 34 and 36, the final version of the encoded video containing the substitute encoded sections has a reduced number of decoding artifacts and therefore, the visible video glitches are reduced, increasing the quality of video as compared to if no correction steps were taken to prevent the decoding artifacts. The examples described above relate to edit points at boundaries of GOPs. However, the above teachings can be applied to edits which are not at GOP boundaries.

[0230] The process to reduce decoding artifacts is carried out using workstations 10, 30, and 40 along with the playback VTR 51 and the video encoder 50. However, other hardware alternatives are possible and the above aspect of the invention may be implemented using any general purpose computer or specific hardware to perform the process. The calculations used to determine the edit points and to determine the video to be added at the end of the new section of video to be encoded are performed either in workstation 10 or workstation 30.

IX. RATE-QUANTIZER MODELING USED FOR ENCODING

[0231] To accurately estimate the number of bits that will be needed to represent a series of macroblocks resulting from an encoding process, a frame can be pre-encoded using several quantization levels that are representative of normal quantization levels for a frame. The frame then can be actually encoded using the representative quantization levels as well as other quantization levels. In the preferred embodiment, illustrated in Figure 1A, workstation 10 loads predetermined quantization levels into the video encoding device 50. The video encoding device 50 then performs the pre-encoding. The most accurate method of generating a quantization model would involve encoding each frame at each possible quantization level and recording the statistics on the resulting number of bits used. However, this process would require n passes over the raw video where n is the total number of quantization levels used to encode the raw video. As this method is prohibitively long, this invention provides an improved method, which generates similar results to the optimal method in a fraction of the time needed for the conventional method. The method involves assigning various parts of plural frames with different quantization levels such that a representative bit-rate versus quantization level average can be calculated for each of the plural frames during a pre-encoding phase. This saves $n-1$ pre-encodings for each frame while generating accurate quantization level versus bit-rate statistics. These statistics can then be interpolated to estimate the number of bits to be used by a macroblock whose quantization level versus bit-rate was not determined in the pre-encoding process. In two alternate embodiments, the method pre-encodes a majority of frames from the raw video or all the frames of the raw video.

[0232] Figure 37A shows a frame of raw video that has two different quantization levels assigned on a macroblock by macroblock basis in the pre-encoding process so that a two point quantization level versus bit-rate line is generated for plural frames and the results are stored for the plural frames. In alternate embodiments, the quantization levels could be assigned in groups or strips of macroblocks. Figure 37B illustrates the uniform distribution of four quantization levels across a frame where the quantization levels are assigned in different positions in the frame to avoid statistical anomalies based on position. In the illustrated frame, for each row, j , the first quantization level in the row is $q_{j \bmod 4}$. This results in a four point quantization level versus bit-rate curve. Figure 37C shows the same process as Figure 37B, but the quantization levels are assigned in a block fashion. Although a single pass of this process produces an accurate quantization level versus bit-rate curve, in an alternate embodiment, this process is repeated a number of times across the plural frames to generate more accurate statistics that approach the statistics that would be gathered by analyzing each frame at each quantization level.

[0233] Figure 38 illustrates the general process of determining the quantization level versus bit-rate characteristics of plural frames. The process starts in step 850 and continues in step 852 by obtaining video signals which represent the raw video to be pre-encoded. In step 854, the process signifies that the first frame, $i=1$, where i is the frame index, is to be pre-encoded. Continuing to step 856, the quantization levels for frame _{$i=1$} are assigned uniformly across the frame to prevent statistical anomalies. In step 858, frame _{$i=1$} is pre-encoded using the quantization levels assigned in step 856. The results of step 858 are determined in step 860 such that the bit-rates for each quantization level assigned in step 856 is recorded for frame _{$i=1$} . In step 862, the method determines whether or not all of the plural frames have been pre-encoded. Since only one frame has been pre-encoded, the process continues to step 866 where it designates the second frame as the frame to be encoded by adding one to the frame index, i . Control is then returned to step 856 which assigns quantization levels to frame _{$i=2$} . The process of assigning, pre-encoding and determining the bit-rates in steps 856, 858, 860 and 862 are continued for all remaining frames. When step 862 determines that all of the plural frames have been

pre-encoded, the process ends by moving to step 864. When step 864 has been reached, a quantization level versus bit-rate curve is approximated by using the quantization levels assigned in step 856 and their corresponding bit-rates.

[0234] Having generated quantization level versus bit-rate curves for plural frames of a raw video, the pre-encoding process moves on to an encoding process by which parts of the raw video are encoded to fit on a digital storage medium, i.e., a compact disc. One, plural, a majority or all of the raw video segments may be encoded in this second phase. If the raw video segments are to be encoded automatically, an encoder uses an activity measure of each macroblock combined with a corresponding quantization level versus bit-rate curve to determine quantization levels to be assigned to macroblocks of each frame in the segments. Each frame in the segments are then encoded and their resultant number of bits are compared with an estimated number of bits based on the quantization levels assigned. In cases where the resultant number of bits varies greatly from the expected number of bits, the automatic encoder attempts to compensate for the variation by changing the quantization levels of the next set of macroblocks to be encoded. This sequential process allows an automatic encoder to encode segments of the entire raw video to match a target number of bits.

[0235] In addition, the quantization level versus bit-rate curves of later frames in a video sequence can be used to regulate quality throughout a resultant encoded digital video. As is shown in Figure 39, Frame N is known to generate fewer bits than Frame M for the same quantization levels. Therefore, Frame M requires more bits to generate the same quality image as in Frame N. When Frame N is encoded at a desired quality without using all its allocated bits, the encoder can carry the extra bits forward to be used in encoding Frame M which requires more bits per unit time for the same quality. This capability is not available in systems with only a single frame look ahead buffer when $M > N+1$.

X. COMBINING AUDIO, VIDEO, AND SUB-PICTURE DIGITAL SIGNALS USING DATA STRUCTURES

[0236] The above portion of this patent specification is primarily concerned with the techniques used during encoding of audiovisual information to generate encoded and compressed digital audiovisual signals. However, in order for the audio and video information to be combined, stored, and subsequently decoded in order to recreate or reconstruct the audiovisual information, such as on a television, it is necessary to combine and interleave the audio, video and other information which has been encoded so that it can be stored and subsequently decoded. The combining of the audio, visual, and other information has been briefly described above as a formatting process which occurs in the workstation 20 illustrated in Figure 1A. Further details of the formatting process and resulting data structures and signals will now be given.

[0237] Turning now to Figure 1A, the MPEG video and encoded audio are stored in the disk 21. Additionally, sub-picture information which includes subtitles and/or graphical representations which are displayed and/or overlaid on top of the video are earlier encoded by a separate workstation, for example, and the encoded sub-picture information is stored on a digital storage medium such as a floppy disk, tape, or other type of disk. This stored sub-picture information is read by one of the digital storage devices 22, combined in a formatting process with the encoded audio and video in disk 21, and formatted into a single data stream by workstation 20. The output of the formatting process is stored on the disk 21 and then written to a digital tape of the digital storage devices 22. The tape is then used to create optical discs by known optical disc manufacturing methods. Accordingly, not only is the invention related to the encoding of the audio and visual data but is also related to a process and system for combining encoded, audio, video, and sub-picture data into a data stream, the digital storage medium and data structures thereon which store the formatted audio, video, and sub-picture information, and is also related to a decoding process and a decoder which separates and reconstructs the audio, video, and sub-picture information for display to a user or consumer. It is to be noted that the data structures on the optical disc or within the formatted data pertain to specific electronic structural elements which impart a physical organization on the information stored in memory. These specific electronic structural elements are stored in an optical disc which can be used for a video on demand system or which are decoded for a user or consumer using a digital video disc player.

[0238] Before describing the specific features of the format structure of the disc or other storage medium for the encoded audiovisual information, known standards on which the invention improves will be described. ISO/IEC 13818-1 describes system aspects of MPEG video and audio. ISO/IEC 13818-1, referred hereinafter as the MPEG system description, describes the basic multiplexing approach for video and audio streams. These streams are broken up into packets called Packetized Elementary Stream (PES) packets. These packets from different streams including audio, and video streams have a common time base and are combined into a single stream. Also described in the MPEG system description is the use of a transport stream which combines one or more programs with one or more independent time bases into a single stream. The MPEG system description describes the use of Presentation Time-Stamps (PTS) which are used to synchronize multiple elementary streams. The time-stamps are generally in units of 90kHz and are used in conjunction with a System Clock Reference (SCR), the Program Clock Reference (PCR) and an optional Elementary Stream Clock Reference (ESCR). A complete description of the details of how the data is formatted according to the MPEG system description are found in ISO/IEC 13818-1.

[0239] In addition to using a formatting process based on ISO/IEC 13818-1, the data is also formatted and stored

according to ISO 9660, 1988, Information Processing - Volume and file structure of CD-ROM for information interchange. This standard describes how the disc volume (not loudness but disc content) and the file structure is organized.

[0240] The preferred embodiment, at the current time, of the optical disc which is used to store the audio, video, and sub-picture data is a single layered double-sided disc having a capacity of 5 gigabytes per side with a total storage capacity of 10 gigabytes. Future formats will allow for a multi-layer disc in order to increase storage capacity, and in addition to using a read-only disc, write-once and write-over technology can also be used. Other aspects of the disc which may be used with the present invention are describe in U.S. patent application 08/304,849 filed September 13, 1994, and entitled "Optical Disk and Optical Disk Apparatus" (US 5 459 712 A). The outside diameter of the disc is preferably 120 mm.

[0241] Figure 40 illustrates the structure of the formatted output of workstation 20 which is ultimately included on an optical disc. This structure 900 includes a lead-in area 902. The lead-in area contains individual fields which controls subsequent processing of the data read from the disc. Exemplary lead-in information included in lead-in area 900 is illustrated in Figure 3 of U.S. Patent 5,400,077 to Cookson et al.

[0242] The system area 904 and volume management information 906 comply with ISO 9660. The volume management information 906 includes a primary volume descriptor, a volume descriptor set terminator, directory records, and path table records. The primary volume descriptor contains basic information of the disc volume such as a descriptor type indicating whether the disc is a video disc of audio disc, a standard identifier, indicating to what standard the disc conforms with, a system identifier, a volume identifier, the volume size, logical block size, path table size, path table size, etc. These and other fields of the system area and volume management information are described in detail in ISO 9660 and a description thereof is omitted for sake of brevity. Additionally, there is a supplementary volume descriptor, if desired, which conforms to ISO 9660.

[0243] The volume descriptor set terminator contains the volume descriptor type, the standard identifier, and the volume descriptor version. The directory records contain various directory information of the disc in accordance with ISO 9660. The path table records contain an L-type path table and an M-type path table, as described in ISO 9660.

[0244] The disc information file 908 includes further information about the content of the disc and menu information which a user may choose when playing the optical disc. Details of the disc information file 908 are described with respect to Figure 41.

[0245] Each disc contains a minimum of one data file 910 up to a maximum of 99 data files. For example, if there are two different audiovisual programs on the disc, the disc information file 908 would contain the information necessary to generate a menu for the user to choose which of the data files the user wish to use. The data files are described in detail with respect to Figures 42-56. The lead-out area 916 but may contain other information describing the disc, how processing is performed, or information concerning related discs.

[0246] Figure 41 illustrates the disc information file 908 of Figure 40. The disc information file 908 includes file management information 920 and menu data 922 which is information making up a menu with which a user interacts. The menu data includes video data, audio data, and sub-picture data. The file management information 920 includes a file management table 924, disc structure information 926, menu structure information 928, and a menu cell information table 930 which includes a plurality of menu cell information fields 932 and 934.

[0247] The file management table 924 includes information related to every aspect of the disc information file. Included within the file management table 924 are a file name which describes the same contents as the corresponding file identifier in the corresponding directory record of the volume management information 906 for identifying a file name. The file identifier identifies the type of file which in this case is a disc structure information file. There is a field which describes the size of the file using the number of logical blocks. Throughout this writing, the term "logical block" is used as defined in ISO 9660. There is an entry describing the size of the file management table using a number of logical blocks, a start address of the disc structure information 926 using the relative logical block number from the head of the file, a start address of the menu structure information 928 described using the relative logical block number from the head of the file, a start address of the menu cell information table 930 using the relative logical block number from the head of the file. If there is no menu data, the value for this field will be zero. There is a field which describes the start address of the menu data 922 using the relative logical block number from the head of the file. When no menu data exists, this entry is zero.

[0248] The file management table 924 also contains a field which describes video attributes of video for the menu data. This information includes whether the video compression mode was MPEG-1 or MPEG-2, the frame rate of the video (either 29.97 frames per second or 25 frames per second), a field indicating whether the display aspect ratio is 3/4 or 9/16, and a display mode field indicating both pan-scan and letter box modes are permitted, indicating that the pan-scan mode is permitted but the letterbox mode is prohibited, or indicating that the pan-scan mode is prohibited and the letter box mode is permitted.

[0249] Similar to the field describing the video attributes, there is also a field describing the audio stream in the file management table 924. Included within this field is an indication of the audio coding mode including whether the audio was encoded according to Dolby AC-3, whether the audio is MPEG audio, or whether the audio is linear PCM audio (16

bits at 48 kHz). There is also an indicator of whether the audio is monaural, stereo, or digital surround.

[0250] The file management table 924 further contains a field which describes the sub-picture stream attributes for the menu data 922. This field indicates a run-length sub-picture coding mode will be used for the sub-pictures, as described below. The file management table 924 also contains a field describing 16 sets of color pallets used in all sub-pictures for the menu. The pallet numbers from 1 through 16 contain a luminance signal Y, a color difference signal Cr=R-Y, and a color difference signal Cb=B-Y.

[0251] The disc structure information 926 includes configuration information of the video and audio files stored on the disc. The disc structure information 926 includes a file name which describes the same contents as the corresponding file identifier in the directory record which is used for identifying a file name. There is a file identifier field which identifies the file as a disc structure information file. There is a field for indicating the number of data files on the disc such as data files 910 and 914 illustrated in Figure 40. There is also a field indicating the type of each of the data files including whether the file contains both video and audio information or only audio information, and whether or not the file is a karaoke file.

[0252] The disc structure information 926 also contains a field describing the sub-picture and audio information in the data file. Indicated within this field are the number of sub-picture channels. Each sub-picture channel may be used to display different sub-picture information such as different sub-picture languages. There is also an indication of the number of audio streams, an indication of the language code of each sub-picture channel in the order of channel number, a sequential description of the language code of the audio stream in the order of the audio stream number, and a description of parental management and angle information of the files. The parental management information is used to describe the parental level of a particular sequence of information to be displayed. It is the maximum parental level of specific cells contained within a sequence. The levels vary from 1 through 5 and the higher the level, the more restriction over what is heard and seen. Details of the sequence and cells and parental management scheme are explained below. There is also a field which describes the number of angles of a video sequence. For example, if a music video had three different camera views throughout the entire music video, the user may select which particular camera angle is desired to be viewed. There is also a field indicating the number of programs in a sequence.

[0253] The menu structure information 928 contains the start cell number of each menu type. A display sequence includes a plurality of cells. One sequence can contain up to 256 cells, for example. A cell is an integral number of GOPs (group of pictures) between a playback information (PBI) pack, to be described later, and a pack immediately before the next PBI pack. Data for video is separated into cells as a record unit in which the data should be reproduced successively. A cell contains the information for display and is divided depending on its purpose. It is desirable that the first audio pack and the first sub-picture pack in a cell include a presentation time-stamp (PTS) close to a PTS in the video head pack recorded behind a PBI pack at the head of a cell. Different cells are displayed depending on the desired angle and parental restriction of the program. As an example, suppose there was a section of a movie which was inappropriate for younger audiences. There can be two sequences sharing cells, depending on the parental management level which may be viewed. For example, for unrestricted viewing, cells 1, 2, 3 and 5 may be viewed for a particular sequence. For a restricted level of viewing, the cells viewed for the sequence may be 1, 2, 4 and 5.

[0254] Regarding the different angles, it is possible to have each video recorded at plural camera angles which proceed simultaneously and whose time required for reproduction is substantially equal. Each angle is constituted by one cell called an angle cell. A set of plural angle cells which precede simultaneously is called an angle block. If an angle block is defined in plural locations in the sequence, the number of angle cells and the camera angle of each angle number must be the same through the angle blocks.

[0255] A menu cell is the cell constituting the menu screen which informs the user of the contents of each video or audio file in a volume. A set of menu cells is recorded together in the disc information file as data for the menu. The menu cell types, also referred to as menu types, are classified into the title menu, a program menu, an audio menu, a sub-picture menu, and an angle menu. A menu screen includes video for the menu. Items for selection are displayed on the screen of reproduction equipment are recorded as sub-picture data of the menu cell.

[0256] The menu cell information table 930 is a table on which plural menu cell information such as 932 and 934 in Figure 41 are stored. One menu screen includes one menu cell. The menu cell information includes a plurality of fields. A first field indicates whether or not copying of information of the disc is permitted, the level of parental management, the menu cell type such as whether the menu is a title menu, program menu, audio menu, sub-picture menu, or angle menu, and the language code of the menu. Each language is assigned a code and the specific code appearing in the language code sub-field indicates the language used with the menu.

[0257] There is also a field which describes the start number of selection items displayed by the sub-picture on the menu screen and the number of the selection items. The selection item start number describes the smallest number of selection items on the menu screen, there is a field which indicates whether or not a next page of the menu exists, and a number which describes the number of items between 1 and 16 which may be selected. The menu cell information also includes the lower 32 bits of the system clock reference (SCR) in the start pack of the menu cell, a field which describes the start address of the menu cell with a relative logical block number from the head of the file, and a field

which describes the number of logical blocks constituting the menu cell.

[0258] The menu data 922 contains the actual information which is displayed for the menu. This information is constructed in the same manner as the information in the data files 910 and 914 in Figure 40 is constructed and accordingly, an in depth description of each aspect the menu data 922 is omitted to prevent redundant descriptions. The items which are selected on the menu screen are formed by sub-picture information. The processing of menu selections by a disc player is performed using a microprocessor executing computer code. When a menu selection for a title is made, the result of the selection is reproduced from the beginning of the file or sequence number corresponding to the selected title number. For program menu selections, the selected program is reproduced from the beginning of the selected program. For audio selections from the menu, the audio stream is set to correspond to the selected audio stream. Similarly, for sub-picture menu selections the sub-picture channel number is set equal to the selected sub-picture channel. Last, an angle selection from the menu changes the angle cell number to equal the selected number. Other information, as desired, may be included in and selected through menus.

[0259] The data files such as the data file 910 of Figure 40 contain two sections, as illustrated in Figure 42; data file management information 940 and the actual data 942 used to display audio, video, and sub-picture information. The data file management information 940 includes four major components; the data file management table 944, the sequence information table 946, the cell information table 948, and a data search map 950.

[0260] The data file management table 944 includes information related to every aspect of the data file 910. This information includes a field for the file name which is the same contents as the corresponding file identifier in the directory record for identifying the file name, the file identifier which indicates that the file contains both audio and video information, the size of the file as defined by the number of logical blocks, the size of the file management table, the number of sequences for the file, the number of cells for the file, the number of disc structure information (DSI) packs, a start address of the sequence information table 946, a start address of the cell information table 948, a start address of the data search map, a start address of the data 942, a field describing the video attributes of the data 942 including the video compression mode such as MPEG-1 or MPEG-2, the frame rate including whether the frame rate is 29.97 frames per second or 25 frames per second, a display aspect ratio indicating whether the display aspect ratio is 3/4 or 9/16, or whether the display mode permits one or both of pan-scan and letterbox format.

[0261] The data file management information further includes a field describing the number of audio streams in the file, the audio stream attributes such as the audio coding mode, the audio mode such as monaural, stereo, or digital surround, the audio type which indicates whether or not there is a specific language included, and a field for a specific code which indicates the code number of the language.

[0262] The data file management table 944 further includes a field describing the number of sub-picture channels, a field describing the sub-picture channel attributes such as a field which indicates run-length coding of the sub-pictures is used and whether or not there is a specified language and the language which is specified. Additionally, there is a field indicating the Y, Cr, and CB colors for 16 color pallets used in all sub-picture channels of the file. There is also a field describing the size of the sequence information table, a field describing the smallest cell number and number of consecutive cells which follows, and the range of cell numbers used within one sequence of the file. The cell numbers used within one sequence should be included in a set of numbers figured out by letting 32 consecutive numbers make a set, and placing the smallest cell number to be the first number of the set. This field includes a sub-field including the number of cells used in the sequence, and the smallest cell number used in the sequence.

[0263] The sequence information table 946 includes a plurality of sequence information entries 952 and 954. A sequence is the order in which cells within the range specified by this sequence are reproduced selectively. There are two types of sequences; completion and connection. A completion-type sequence is terminated after reproducing itself. Connection-type sequences are sequentially reproduced. The sequence information number such as sequence information 1, indicated by reference number 952, and sequence information i, indicated by 954, is equal to the sequence number and is numbered in the described order in the sequence information table beginning with 1. The sequence information entries of the sequence information table are described in the order of completion-type sequences or connection-type sequences. Each sequence information contains field indicating the sequence type, the number of included programs, the number of included cells, the sequence reproduction time, the number of connection-type sequences, the connectable sequence number, and sequence control information.

[0264] The sequence type field of the sequence information describes copy and parental management of the sequence. There is a sub-field indicating whether copying is permitted or prohibited, a sub-field specifying the level of parental management from 1-5, a sub-field indicating the sequence type which may be one of a completion-type sequence, a connection-type head sequence, a connection-type midsequence or a connection-type end sequence. There is also an application type sub-field which indicates whether or not the sequence is for use as a karaoke application.

[0265] The number of included programs field describes the number of programs, up to 100 in a sequence. The number of included cells field describes the number of total cells in a particular sequence, up to 256 cells in total. The sequence or reproduction time describes the total reproduction time of this sequence in units by hour, minute, second, and video frame.

[0266] The number of connection-type sequences field includes the number of sequences which can be connected immediately after them, the maximum number being 8. The connectable sequence number field describes sequence numbers of connectable sequences and the level for parental management of the sequence for the numbers specified. The selection number of connectable sequences are assigned from one in the described order. This field contains sub-fields for the parental management number, and a sub-field for the sequence number. The last field of the sequence information is the sequence control information which describes the cells included in the sequence in the order of reproduction. This field is also used to describe angle blocks which is a set of plural cells. The reproduction time of those cells is substantially equal. Each reproduction section is constituted by one cell. An angle block includes nine cells at most with the first cell numbered 1. The sub-fields of the sequence control information include a program number which defines a program as a reproduction unit combining plural cells which are sequentially reproduced. There is a sub-field for cell reproduction control which indicates whether to proceed to the following cell sequentially, to pause, or whether the end of a sequence has been reached, there is a block mode sub-field which indicates if a block is not a configuration cell, whether there is a first cell of block configuration, whether it is a cell in a block configuration, or a final cell in a block configuration. There is a block type sub-field which indicates if there is not a block or whether there is an angle block. Last, there is a sub-field which indicates the cell numbers of the cells to be reproduced.

[0267] The data file management information 940 further includes a cell information table 948 containing a plurality of cell information entries 956 and 958. Each cell information entry includes cell type information which indicates whether copying is permitted or prohibited, and the level of parental management. There is also a field indicating cell reproduction time which describes the total cell reproduction time of the cell by hour, minute, second and video frame. There is a field which describes the lower 32 bits of the system clock reference described in the first pack of the cell, a field which describes the address at the cell's starting point with the relative logical block number from the beginning of the file, and a field which describes the number of logical blocks included in the cell.

[0268] The last section of the data file management information 940 is the data search map 950. The data search map includes a plurality of pointers of data search information packs 960 and 962. These pointers, referred to as playback information (PBI) pack pointers, give the address of the playback information pack which exists in the data 942. In order to perform a fast forward or fast reverse mode and allow viewing of the information, it is best to rely on intra pictures in the MPEG-2 video sequence. These intra pictures are located using playback information packs within the data 942 whose addresses are contained in the data search map. A further description of the PBI pack is given below.

[0269] The data 942 of the data file 910, illustrated in Figure 43, includes interleaved packs of playback information (PBI), video, sub-picture, and audio information. For example, in Figure 43, the data includes playback information 940, video information 942 and 944, sub-picture information 946, audio information 948 etc. The structure of the information within the data 942, and presentation timing complies with the program stream defined in the MPEG system description (ISO/IEC 13818-1). However, the playback information and sub-picture information and the manner in which this information is stored, encoded, and decoded constitute part of the present invention.

[0270] The various information making up the data 942 in Figure 43 includes information packs. Exemplary packs 970A and 970B are illustrated in Figures 44A and 44B. Each pack comprises a pack header 972A or 972B, and a packet 982A or 982B which includes video, audio, sub-picture, or playback information data.

[0271] The structure of pack 970A is used when the packet 982A occupies between 2034 and 2027 bytes. The stuffing field 980A allows 1 to 8 bytes of stuffing to bring the total size of the pack to 2048 bytes. When the packet for video, audio, sub-picture, or playback information is less than 2027 bytes, a pack structure 970B as illustrated in Figure 44B is used which has a stuffing 980B of one byte and a packet for padding 984 which makes the total number of bytes for the packets 982B and 984B to be 2034 bytes. The packet headers 972A and 972B each contain a pack start field 974, a system clock reference 976, a multiplex (MUX) rate field 978 and a stuffing field 980. The packets in Figure 44A and 44B are constructed in accordance with ISO/IEC 13818, and described in §2.5.3.4 thereof pertaining to the semantic definition of fields in a program stream pack.

[0272] Figure 45 illustrates a playback information (PBI) pack. The pack header 972 is constructed as described with respect to Figures 44A and 44B. The system header 986 of Figure 45 is constructed in conformance with the requirements of the program stream system header described in ISO/IEC 13818-1. The semantic definition of fields within the system header is described in §2.5.3.6 of ISO/IEC 13818-1.

[0273] The packet header 988 is constructed in accordance with the fields set forth in the semantic definition of fields in PES packet described at §2.4.3 of ISO/IEC 13818-1. However, only fields up to the presentation time-stamp are needed in the packet header 988.

[0274] The playback information pack further includes a sub-stream ID field 990. This is an 8 bit field which indicates the identification of the sub-stream. The different identifications which may be used are a sub-picture stream, a video blanking information (VBI) stream, an AC-3 stream, or a linear PCM stream. It is also possible to include other stream types such as an MPEG audio sub-stream. As the sub-stream ID 990 precedes the data of the video blanking information 992, the sub-stream ID is set to indicate that the information which follows is a video blanking information stream. The data of the video blanking information 992 may be constructed in accordance with ISO/IEC 13818-1 and conform to the

semantics required for ISO/IEC 13818-1.

[0275] Also included within the playback information pack illustrated in Figure 45 is a packet header 994 for a data search information packet which includes the data of the data search information 996. The packet header 994 includes a packet start code prefix having a 24 bit value of 000001h, a stream identification indicating that the data search information 996 does not conform to previously defined standards, and a field indicating a packet length. As the data search information 996 is not disclosed in the MPEG standard, the stream ID indicates that the information which follows is a private type of data stream.

[0276] The specific content of the data of the data search information 996 illustrating Figure 45 is shown in Figure 46. The data search information 996 includes general information 1000, angle address information 1020, highlight information 1022, synchronous reproduction information 1024, and PBI pack address information 1036.

[0277] The general information 1000 is illustrated in further detail in Figure 47. As previously stated, a purpose for the data search information is to allow a quick search of the digitally encoded video and specifically allows a rapid location of an intra picture for trick-modes executed during video decoding. Therefore, the playback information pack illustrated in Figure 46 is the head pack of each group of pictures (GOP) and allocated immediately before the video head pack starting with an I-picture. The general information 1000 includes a time-stamp of the GOP 1002 which is the reproduction start time of an I-picture. Next, there is a system clock reference (SCR) of the PBI. This describes the lower 32 bits of the SCR described in the PBI pack header. 1006 indicates the end address of the I-picture and describes by the relative logical block number from the beginning of the file, the address of the pack on which the final data of the I-picture is recorded. The cell number is stored in field 1008 and describes the cell number to which the GOP belongs. Field 1010 describes the parental management information including the level of restriction and also whether or not copying is permitted. Last, field 1012 describes the PBI pack address which is the address of this PBI pack using the relative logical block number from the head of the file.

[0278] Field 1020 of the data search information 996 illustrated in Figure 46 is the angle address information. The angle address information includes address information of other angles. The angle address information includes nine fields, each pertaining to a different angle cell number. For each field in the angle address information 1020, if a cell to which a specific PBI pack belongs constitutes an angle block, the field in the angle address information describes the address of the PBI pack for the angle cell number which has the closest video start presentation time-stamp not exceeding the video start presentation time-stamp of the corresponding data search information. If no angle cell exists or if an angle block is not constituted, the angle address information for a specific cell is set to zero.

[0279] The third section of the data search information 996 is the highlight information 1022. The highlight information describes the position of selection items on the menu screen and their color and contrast to be changed when selected. This information is used only when the cell is part of a menu. The highlight information includes a field describing the selection item start number and the number of items displayed by the sub-picture on the menu screen. The specific sub-fields of the selection item start number and number of items include a sub-field which describes the smallest number of selection items on the menu screen, a sub-field which describes whether or not a next page of the menu exists, and a sub-field which describes the number of selection items on the menu screen. The second field of the highlight information includes position, color, and contrast of the selection item. This field describes a rectangular display area for each selection item on the menu screen and the corresponding color and contrast to be changed, when selected. The rectangular display area is defined in an X-Y coordinate plane. Information in this field describing the position, color, and contrast of the selection item include defining the rectangular area of the menu, and information describing how the various types of pixels are changed when selected. As will be described in more detail later, a sub-picture allows for four different types of pixels. There are first and second emphasis pixels which may be used to display text or other information, pattern pixels which are used to display patterns such as line drawings or other graphical patterns, and background pixels. When these four different types of information are highlighted or selected, it is necessary to know the color which will be chosen when the item is highlighted and also the contrast of the pixel, when highlighted. Therefore, there are eight sub-fields within the highlight information describing the color and contrast of the four different types of pixels when they are selected.

[0280] The fifth item within the data search information 996 is the synchronous reproduction information 1024. The synchronous reproduction information 1024 is illustrated in detail in Figure 48. The purpose of the synchronous reproduction information is to be able to find the start time and address information of the audio and sub-picture which is synchronized with the video data. The synchronous reproduction information 1024 illustrated in Figure 48 contains the audio pack address 1026 which is the target of the presentation time-stamp of the audio. The most significant bit of this field indicates whether the audio pack is located before or after this PBI. The presentation time-stamp (PTS) of the audio 1028 describes the PTS of the audio packet having the reproduction start time just behind the reproduction start time of the I-picture with the relative presentation time-stamp from the video start PTS. The presentation time-stamp of the audio allows for up to eight audio streams to have the presentation time-stamp stored therefore. Additionally, there may be a sub-field indicating whether or not the audio packet comprising audio frames which is reproduced during the GOP. If there is no audio for the picture, it will not be necessary to store or decode the presentation time-stamp of the audio.

As there are up to eight audio channels, each audio channel has corresponding address and time stamp fields.

[0281] Data structure 1024 also includes a sub-picture pack address 1030 which describes the address, relative to the PBI pack, of the sub-picture pack at issue. The presentation time-stamp of the sub-picture describes synchronization of corresponding sub-picture information. There are sub-fields which indicates whether or not a sub-picture unit which is to be reproduced during the GOP reproduction exists, whether or not the reproduction start time of the sub-picture is before the video presentation time-stamp, the production start time of a sub-picture unit reproduced during the GOP reproduction, and the reproduction termination time of a sub-picture unit reproduced during the GOP reproduction. As up to 32 sub-pictures which may be included into a single data stream, each sub-picture which exists has a corresponding address and time-stamp field.

[0282] The last section illustrated in Figure 46 of the data search information 996 is the PBI pack address information. The PBI pack address information comprises addresses of other PBI packs within the file and allows other PBI packs to be quickly located. The PBI pack address information describes the address of the surrounding PBI packs using relative logical block numbers. The PBI pack address information includes two fields which describe the next and previous PBI pack addresses, respectively. There are fifteen fields which define, using the relative logical block number from the current PBI pack address, the address of the PBI pack which belongs to the GOP immediately ahead of the GOP with its accumulated reproduction time being between $n \times 0.5$ seconds and $(n+1) \times 0.5$ seconds, and when more than one PBI pack falls in this range, selecting the PBI pack closest to $n \times 0.5$ seconds, $1 \leq n \leq 15$.

[0283] There are fifteen fields which define, using the relative logical block number from the current PBI pack address, the address of the PBI pack which belongs to the GOP with its accumulated reproduction time being between $n \times 0.5$ seconds and $(n+1) \times 0.5$ seconds, and when more than one PBI pack falls in this range, selecting the PBI pack closest to $n \times 0.5$ seconds, $-15 \leq n \leq -1$.

[0284] There are two fields which define, using the relative logical block number from the current PBI pack address, the address of the PBI pack which belongs to the GOP immediately ahead of the GOP with its accumulated reproduction time being between $n \times 0.5$ seconds and $[n \times 0.5 + 1]$ seconds, and when more than one PBI pack falls in this range, selecting the PBI pack closest to $n \times 0.5$ seconds, $n=20$ or $n=60$.

[0285] There are two fields which define, using the relative logical block number from the current PBI pack address, the address of the PBI pack which belongs to the GOP with its accumulated reproduction time being between $n \times 0.5$ seconds and $[n \times 0.5 + 1]$ seconds, and when more than one PBI pack falls in this range, selecting the PBI pack closest to $n \times 0.5$ seconds, $n=-20$ or $n=-60$.

[0286] The structure of a video pack 1040 is illustrated in Figure 49. The video pack contains a pack header 972 constructed in accordance with the headers 972A or 972B illustrated in Figures 44A and 44B. The packet header 1046 for the video is constructed in accordance with ISO/IEC 13818-1 as is the video data 1048. However the video data 1048, represented as MPEG video signals is preferably encoding in accordance with the techniques described above.

[0287] An MPEG encoded group of pictures is illustrates as 1050 in Figure 50. The manner of breaking up this GOP 1050 into a plurality of video packs 1040 to generate the video stream 1060 is illustrates at the bottom portion of Figure 50. The group of pictures 1050 is broken up into appropriate units to fit into the video packs. The last video pack 1040G, video pack j, does not take up the full 2048 bytes used for each of the packs and therefore, stuffing is used to make the last video pack j equal to 2048 bytes.

[0288] Figure 51A illustrates a video pack for MPEG audio. This video pack 1080 includes a pack header 972, a packet header 1084 constructed in conformance with the MPEG requirements, and the encoded MPEG audio data 1086.

[0289] Figure 51B illustrates an audio pack 1090 constructed in conformance with either linear PCM audio or Dolby AC-3 audio. For the packet illustrated in Figure 51B, the pack header 972 is arranged as described above, the packet header 1094 is constructed in conformance with the MPEG requirements with the stream_id field set to indicate a private audio data stream corresponding either to linear PCM or AC-3. The sub-stream id 1096 indicates whether the audio data is for liner PCM or AC-3 (or MPEG). The audio frame pointer 1098 includes a number of frames header which indicates the number of audio frame headers within the audio packet and also includes a first access unit pointer which indicates the head of the first access unit of the audio frame using a relative byte number from the last byte of the pointer. It is standard for linear PCM audio to be broken into audio frames, each frame including a frame header and frame data. The audio frame pointer 1098 may similarly be used with dolby AC-3. The audio data 1100 is constructed in accordance with the corresponding audio encoding method.

[0290] Figure 52 illustrates an encoded audio stream 1100. This audio stream is converted into a stream of audio packs 1120 comprising individual packs 1122. Each pack is 2048 bytes in length and the last audio pack 1122G, audio pack j, is adjusted by adding stuffing bits in order to bring its length to 2048 bytes.

[0291] A novel feature of the present invention is the use of sub-pictures and the interleaving of sub-picture packs with audio and video information. Sub-picture information allows the display of any graphical information and allows for overlapping the sub-picture information over the MPEG video. A major difference between the sub-picture information used by the present invention and conventional closed caption subtitles is that the sub-picture information is bit mapped graphics whereas closed captioning transmits character codes to display characters within character sets which are

stored in the decoder. Accordingly, conventional closed captioning is limited by the character set within the decoder. However, as the sub-picture information is bit mapped, any type of characters including foreign language characters may be displayed without the need to store character sets within the decoder for each language.

[0292] Each display screen of sub-picture information is called a sub-picture unit. There is a sub-picture unit 1140 illustrated in Figure 53. A sub-picture unit contains one screen of bit mapped pixel data and this screen of pixel data is displayed across a plurality of video frames. The sub-picture unit 1140 includes a sub-picture unit header 1142. The sub-picture unit header 1142 includes a field describing the size of the sub-picture unit and the start address of the display control sequence table 1146, relative to the number of bytes from the head of the sub-picture unit.

[0293] The pixel data 1144 is run-length compressed bit mapped information. The pixels may be background pixels which form the background of the displayed information, pattern pixels which allow the display patterns within the sub-picture including graphical line drawings, and two types of emphasis pixels which allow graphics or characters to be displayed which have two different attributes. The bit mapped information may be created using conventional graphics generating techniques. The run length compression is performed in accordance with the following rule. If one to three pixels of the same kind follow, enter the number of the pixels in the first two bits and the pixel data in the following two bits. The four bits are considered a unit. If four to fifteen pixels of the same kind follow, specify zero in the first two bits and enter the number of pixels in the following four bits and the pixel data in the next two bits. The eight bits are considered to be one unit. If sixteen to sixty-three pixels of the same kind follow, specify zero in the first four bits and enter the number of the pixels in the following six bits and the pixel data in the next two bits. The twelve bits are considered to be one unit. If sixty-four to 255 pixels of the same kind follow, specify zero in the first six bits and enter the number of the pixels in the following eight bits and the pixel data in the next two bits. The sixteen bits are considered to be one unit. If the same pixels follow to the end of a line, specify zero in the first fourteen bits and describe the pixel data in the following two bits. The sixteen bits are considered to be one unit. If the byte alignment does not result when the description for pixels on one line is completed, insert dummy data of four bits for adjustment. It is also possible to use other compression mechanisms to represent the image data, as desired. For example, JPEG or GIFF formats may be used to represent the sub-picture images.

[0294] The display control sequence table 1146 contains a plurality of display control sequences 1146A, 1146B, 1146C, etc., which describe the chronological order in which control sequences occur. The display control sequence table contains information regarding how the information in the sub-picture unit is displayed. For example, words forming a single sub-picture unit may appear one-by-one or in groups over time, or their colors may change, which is useful when using sub-pictures to display karaoke information.

[0295] Each display control sequence (DCSQ) allows for commands changing the pixel data to be executed over a time period during which the sub-picture unit is displayed. The first field of a display control sequence is a display start time. This field describes the execution start time of the display control command included in the DCSQ having a relative PTS from the PTS described in the sub-picture packet with a sub-picture unit header. From the first video frame after the described execution start time, display control is started according to the DCSQ display control start time. The second field of each DCSQ is the address of the following display control sequence. This field describes the start address of the following DCSQ with the relative byte number from the first sub-picture unit. If no following DCSQ exists, this field describes the start address of this DCSQ with a relative byte number from the first sub-picture unit. Thereafter, the DCSQ contains one or more display control commands. These commands allow the attributes and display of the pixel data to be controlled and changed. The commands include a command to forcedly start the display of a sub-picture unit, irrespective of the on/off state of the sub-picture. For example, if a user uses a menu to turn off sub-pictures, it is possible to override the user's setting by setting this command. Another command starts the display of an updated sub-picture unit. This command must appear once in each DCSQ. There are commands to set the colors and contrast of the above-described four types of pixels including the two types of emphasis pixels, the pattern pixel, and the background pixel. A command exists which sets the rectangular area display and position of the pixel data making up the sub-picture. This command allows the setting of the upper and lower X and Y coordinates. There is also a command used to set the head address of the pixel data used for displaying. This command allows the head address of both the top and bottom field to be entered. In the extended fields, each head address of the pixel data for the top field and bottom field with the relative byte number from the head of the unit is used. The first pixel data represents the first pixel on the line. This command must be used at least in the first DCSQ, DCSQO.

[0296] There is a command which may appear in a DCSQ to change the color and contrast of the pixel data. This command is not to be used when the highlight information of the data search information is used. The command to change the color and contrast of pixel data includes pixel control data to control the color and contrast of the pixel data during display. The content of the control described in the pixel control data is executed for every video frame after the specified start time and continued until new pixel control data is encountered or a new sub-picture unit is encountered. The pixel control data includes line control information used to specify lines on which the same change is made. Also, pixel control information can be used to specify a position on a line at which the change is made. The line control command allows the changing of the start line number, the number of change points, and the change termination line

number. The pixel control information includes the change start pixel number, and the new color and contrast for the emphasis pixels 1 and 2, the pattern pixels, and the background pixels. There is also a command which indicates the end of a display control sequence. Each DCSQ must end with this command.

[0297] Over a time period of video frames such as for a movie, there are many different sub-picture units which may be used. The sub-picture units are broken into sub-picture packs 1150, as illustrated in Figure 54. The sub-picture pack contains a pack header 972, as described above, a packet header which conforms to the MPEG system requirements, a sub-stream ID which indicates that the following data is sub-picture data, and the sub-picture data 1158 itself.

[0298] The sub-picture units such as sub-picture unit 1140 is divided into sub-picture packs 150, as illustrated in Figure 55. The last sub-picture pack 1050G contains padding to make it 2048 bytes in length.

[0299] Figure 56 illustrates the manner in which successive sub-picture units are displayed. When the time equals to the time of a PTS described in a new sub-picture packet, the sub-picture unit currently being displayed (1160) is cleared (1162) and when the time specified by the display control sequence of the next sub-picture unit occurs, that sub-picture is displayed (1164). Between 1160 and 1162, there is a dividing line between the sub-picture unit n and the sub-picture unit n+1. This dividing line occurs at the PTS described in the packet header of the sub-picture unit n+1. The actual display time for sub-unit n+1 is set forth in the DCSQ of sub-picture n+1.

[0300] The creation of the pixel data used for the sub-picture information may be performed using a computer to create bit maps of the information which is desired to be displayed. The process of creating bit-mapped information is well-known in the art.

[0301] Figures 40-56 emphasize the data structures used to encode the audio, video, and sub-picture information. However, by the disclosure of the data structures in Figures 40-56, one of ordinary skill in the art of MPEG encoding and in particular with the assistance of the MPEG system description ISO/IEC 13818-1, could format encoded audio, video, and sub-pictures into the data structures. Similarly, with the knowledge of the structures in which the data is stored, the decoding of the data structures to generate the video, audio, and sub-picture information can be performed by one of ordinary skill in the art.

[0302] An exemplary video decoder which constitutes part of the present invention includes an optical disc reader which reads data stored on the optical disc. The information which is read is parsed using conventional parsing techniques in order to decode the information. The video, audio, sub-picture, and PBI packs must all be decoded. The video may be decoded using commercially available MPEG decoders as are the audio packs decoded using conventional decoders. The sub-picture information is decoded by constructing sub-picture units from the sub-picture packs in an inverse manner as they were encoded. Specifically constructed hardware or a general purpose microprocessor programmed using software coding may be used to decode the sub-picture information. The playback information packs, as illustrated in Figure 45 contain data search information 996. The data search information is processed using a programmed microprocessor and is different from the audio, video, and sub-picture decoders. As the menu functions are also performed by a programmed microprocessor, the highlight information 1022 included within the data search information and all other menu functions are performed by the programmed microprocessor and not by the sub-picture decoder. Data structures, encoding and decoding process are described in Japanese Patent Applications 7-81298 (EP 737 009 A), and 7-85693 (EP 737 975 A).

[0303] The encoding system of the present invention allows the control of quality within pictures. This control is important as the people creating the final video product will be able to closely control the final product of the encoding system. Therefore, even though digital video encoding results in decoding artifacts which are not present in analog video, the final digital product will be quite acceptable. The teachings in any of the above sections are applicable to the other sections.

[0304] The determinations, calculations, and steps of the present invention may be conveniently implemented using a conventional general purpose digital computer programmed according to the teachings of the present invention, as will be apparent to those skilled in the computer art. Appropriate software coding can readily be prepared by skilled programmers based on the teachings of the present disclosure, as will be apparent to those skilled in the software art.

[0305] The invention may also be implemented by the preparation of application specific integrated circuits or by interconnecting an appropriate network of conventional component circuits, as will be readily apparent to those skilled in the art.

[0306] The present invention includes a computer program product which is a storage medium including instructions which can be used to program a computer to perform a process of the invention. The storage medium can include, but is not limited to, any type of disk including floppy disks, optical disks, CD-ROMs, and magneto-optical disks, ROMs, RAMs, EPROMs, EEPROMs, magnetic or optical cards, or any type of media which is suitable for storing electronic instructions. The present invention further includes a computer program product which is a storage medium including encoded data output by the present invention stored on any of the above described media suitable for storing electronic instructions or data.

Claims

1. A method for processing video signals included in video frames to determine characteristics of the video frames, comprising the steps of :
 - a) obtaining video signals included in plural video frames ;
 - b) assigning plural quantization levels to one of said plural video frames such that different parts of said plural frames are assigned different quantization levels;
 - c) pre-encoding said one of said plural video frames using said plural quantization levels to obtain plural bit-rates corresponding to each of said plural quantization levels ;
 - d) determining bit-rates for each of said plural quantization levels resulting from said pre-encoding step ;
 - e) repeating steps b-d sequentially for plural succeeding video frames ; and
 - f) encoding a plurality of said plural video frames after determining the bit-rates for each of said plural video frames;
 - g) determining a target number of bits for said plurality of said plural video frames ;
 - h) assigning a first set of quantization levels for each frame in said plurality of said plural video frames using said target number of bits and the plural determined bit-rates;
 - i) encoding one video frame of said plurality of said plural video frames into an encoded frame with a resultant number of bits ;
 - j) estimating an encoded number of bits for a succeeding video frame ;
 - k) adjusting the first set of quantization levels for said succeeding video frame based on said target number of bits, said bit-rates for succeeding video frames, the estimated encoded number of bits and the resultant number of bits ; and
 - l) repeating steps i-k sequentially for plural succeeding video frames.
2. The method of Claim 1, wherein the step of obtaining video signals comprises obtaining said video signals from a video camera.
3. The method of Claim 1, further comprising the step of storing the plural determined bit-rates.
4. The method of Claim 1, wherein the step of assigning comprises assigning said plural quantization levels uniformly across each of said plural video frames.
5. The method of Claim 1, wherein the step of assigning comprises assigning two quantization levels uniformly across each of said plural video frames in a checkerboard pattern.
6. The method of Claim 1, wherein the step of assigning comprises assigning four quantization levels, q_1 , q_2 , q_3 , q_4 , uniformly across each of said plural video frames such that a first quantization level of each row, j , is $q_{j \bmod 4}$.
7. The method of Claim 1, wherein the step of determining a target number of bits comprises determining the target number of bits automatically.
8. The method of Claim 1, wherein the step of determining a target number of bits comprises determining the target number of bits through user-intervention.
9. The method of Claim 1, wherein the step of assigning further comprises assigning said first set of quantization levels based on an activity measure of plural macroblocks of corresponding video frames.
10. The method of Claim 1, wherein the step e) repeats for a majority of video frames in said plural video frames.
11. The method of Claim 1, wherein the step e) repeats for all video frames in said plural video frames.
12. A system for processing video signals included in video frames to determine characteristics of video frames, comprising :
 - means for obtaining video signals included in plural video frames ;
 - means for assigning plural quantization levels to a first plurality of said plural frames of the video signals such that different parts of said plural frames are assigned different quantization levels;

means for pre-encoding each video frame in said first plurality of said plural video frames using said quantization levels to obtain plural bit-rates corresponding to each of said plural quantization levels ;
 means for determining said plural bit-rates for each of said plural quantization levels for each video frame in said first plurality of said plural video frames generated by said means for pre-encoding ; and
 5 means for encoding a second plurality of said plural video frames after determining said plural bit-rates for each video frame in said first plurality of said plural frames;
 means for determining a target number of bits for said second plurality of said plural video frames ;
 and
 means for assigning a first set of quantization levels for each frame in said second plurality of said plural video frames using said target number of bits and the plural determined bit-rates;
 10 means for encoding one video frame of said second plurality of said plural video frames into an encoded video frame with a resultant number of bits ;
 means for estimating an encoded number of bits for a succeeding video frame ; and
 means for adjusting the first set of quantization levels for said succeeding video frame bases on said target number of bits, said bit-rates and the estimated encoded number of bits for succeeding video frame and the resultant number of bits of said one of said second plurality of frames.

13. The system of Claim 12, wherein said means for obtaining comprises means for obtaining said video signals from a video camera.
14. The system of Claim 12, further comprising means for storing the determined bit-rates.
15. The system of Claim 12, wherein said means for assigning comprises means for assigning said plural quantization levels uniformly across each of said plural video frames.
16. The system of Claim 12, wherein said means for assigning comprises means for assigning two quantization levels across each of said plural frames in a checkerboard pattern.
17. The system of Claim 12, wherein said means for assigning comprises means for assigning four quantization levels, q_1, q_2, q_3, q_4 , uniformly across each of said plural video frames such that a first quantization level of each row, j , is $q_{j \bmod 4}$.
18. The system of Claim 14, wherein the means for determining a target number of bits comprises means for determining the target number of bits automatically.
19. The system of Claim 14, wherein the means for determining a target number of bits comprises means for determining the target number of bits through user-intervention.
20. The system of Claim 14, wherein the means for assigning further comprises means for assigning said first set of quantization levels based on an activity measure of plural macroblocks of corresponding video frames.
21. The system of Claim 12, wherein the means for assigning comprises means for assigning plural quantization levels to a majority of said plural video frames.
22. The system of Claim 12, wherein the means for assigning comprises means for assigning plural quantization levels to all of said plural video frames.

Patentansprüche

1. Verfahren zum Verarbeiten von Videosignalen, die in Videobildern beinhaltet sind, um Eigenschaften der Videobilder zu bestimmen, die folgenden Schritte aufweisend:
 - a) Erlangen von Videosignalen, die in mehreren Videobildern beinhaltet sind;
 - b) Zuordnen von mehreren Quantisierungsebenen einem von mehreren Videobildern derart, dass unterschiedliche Teile der mehreren Bilder unterschiedlichen Quantisierungsebenen zugeordnet sind;
 - c) Vorcodieren des einen der mehreren Videobilder unter Verwendung der mehreren Quantisierungsebenen zum Erlangen mehrerer Bitraten entsprechend jedem der mehreren Quantisierungsebenen;

- d) Bestimmen von Bitraten für jede der mehreren Quantisierungsebenen, die aus dem Vorcodierungsschritt resultieren;
e) Wiederholen der Schritte b-d sequenziell für mehrere aufeinanderfolgende Videobilder; und
f) Codieren einer Mehrzahl der mehreren Videobilder nach Bestimmen der Bitraten für jedes der mehreren Videobilder;
5 g) Bestimmen einer Zielanzahl von Bits für die Mehrzahl der mehreren Videobilder;
h) Zuordnen eines ersten Satzes aus Quantisierungsebenen für jedes Bild in der Mehrzahl der mehreren Videobilder unter Verwendung der Zielanzahl von Bits und der mehreren bestimmten Bitraten;
i) Codieren eines Videobilds der Mehrzahl der mehreren Videobilder in ein codiertes Bild mit einer resultierenden Anzahl von Bits;
10 j) Schätzen einer codierten Anzahl von Bits für ein nachfolgendes Videobild;
k) Einstellen des ersten Satzes Quantisierungsebenen für das nachfolgende Videobild basierend auf der Zielanzahl von Bits, den Bitraten für nachfolgende Videobilder, der geschätzten codierten Anzahl von Bits und der resultierenden Anzahl von Bits; und
15 l) Wiederholen der Schritte i-k sequenziell für mehrere aufeinanderfolgende Videobilder.
2. Verfahren gemäß Anspruch 1, wobei der Schritt zum Erlangen von Videosignalen Erlangen der Videosignale von einer Videokamera umfasst.
- 20 3. Verfahren gemäß Anspruch 1, ferner umfassend den Schritt zum Speichern der mehreren bestimmten Bitraten.
4. Verfahren gemäß Anspruch 1, wobei der Schritt zum Zuordnen Zuordnen der mehreren Quantisierungsebenen einheitlich über jedes der mehreren Videobilder umfasst.
- 25 5. Verfahren gemäß Anspruch 1, wobei der Schritt zum Zuordnen Zuordnen von zwei Quantisierungsebenen einheitlich über jedes der mehreren Videobilder in einem Schachbrettmuster umfasst.
6. Verfahren gemäß Anspruch 1, wobei der Schritt zum Zuordnen Zuordnen von vier Quantisierungsebenen q_1, q_2, q_3, q_4 einheitlich über jedes der mehreren Videobilder derart umfasst, dass eine erste Quantisierungsebene von jeder Reihe $j \bmod 4$ ist.
- 30 7. Verfahren gemäß Anspruch 1, wobei der Schritt zum Bestimmen einer Zielanzahl von Bits automatisches Bestimmen der Zielanzahl von Bits umfasst.
- 35 8. Verfahren gemäß Anspruch 1, wobei der Schritt zum Bestimmen einer Zielanzahl von Bits Bestimmen der Zielanzahl von Bits durch Benutzerintervention umfasst.
9. Verfahren gemäß Anspruch 1, wobei der Schritt zum Zuordnen ferner Zuordnen des ersten Satzes Quantisierungsebenen basierend auf einer Aktivitätsmessung mehrerer Makroblöcke entsprechender Videobilder umfasst.
- 40 10. Verfahren gemäß Anspruch 1, wobei der Schritt e) sich für eine Mehrzahl von Videobildern in den mehreren Videobildern wiederholt.
11. Verfahren gemäß Anspruch 1, wobei der Schritt e) sich für alle Videobilder in den mehreren Videobildern wiederholt.
- 45 12. System zum Verarbeiten von Videosignalen, die in Videobildern beinhaltet sind, zum Bestimmen von Eigenschaften von Videobildern, aufweisend:
- Mittel zum Erlangen von Videosignalen, die in mehreren Videobildern beinhaltet sind;
50 Mittel zum Zuordnen mehrerer Quantisierungsebenen einer ersten Mehrzahl der mehreren Bilder der Videosignale derart, dass unterschiedliche Teile der mehreren Bilder unterschiedlichen Quantisierungsebenen zugeordnet sind;
Mittel zum Vorcodieren jedes Videobilds in der ersten Mehrzahl der mehreren Videobilder unter Verwendung der Quantisierungsebenen zum Erlangen mehrerer Bitraten entsprechend jeder der mehreren Quantisierungsebenen;
55 Mittel zum Bestimmen der mehreren Bitraten für jede der mehreren Quantisierungsebenen für jedes Videobild in der ersten Mehrzahl der mehreren Videobilder, die durch die Mittel zum Vorcodieren erzeugt werden; und
Mittel zum Codieren einer zweiten Mehrzahl der mehreren Videobilder nach Bestimmen der mehreren Bitraten

- für jedes Videobild in der ersten Mehrzahl der mehreren Bilder;
Mittel zum Bestimmen einer Zielanzahl aus Bits für die zweite Mehrzahl der mehreren Videobilder; und
Mittel zum Zuordnen eines ersten Satzes Quantisierungsebenen für jedes Bild in der zweiten Mehrzahl der
mehreren Videobilder unter Verwendung der Zielanzahl von Bits und der mehreren bestimmten Bitraten;
5 Mittel zum Codieren eines Videobilds der zweiten Mehrzahl der mehreren Videobilder in ein codiertes Videobild
mit einer resultierenden Anzahl von Bits;
Mittel zum Schätzen einer codierten Anzahl von Bits für ein nachfolgendes Videobild; und
Mittel zum Einstellen des ersten Satzes Quantisierungsebenen für das nachfolgende Videobild basierend auf
der Zielanzahl von Bits, den Bitraten und der geschätzten codierten Anzahl von Bits für das nachfolgende
10 Videobild und der resultierenden Anzahl von Bits des einen der zweiten Mehrzahl von Bildern.
13. System gemäß Anspruch 12, wobei das Mittel zum Erlangen Mittel zum Erlangen der Videosignale von einer Vide-
okamera umfasst.
- 15 14. System gemäß Anspruch 12, wobei ferner umfassend Mittel zum Speichern der bestimmten Bitraten.
15. System gemäß Anspruch 12 wobei das Mittel zum Zuordnen Mittel zum Zuordnen der mehreren Quantisierungse-
benen einheitlich über jedes der mehreren Videobilder umfasst.
- 20 16. System gemäß Anspruch 12, wobei das Mittel zum Zuordnen Mittel zum Zuordnen von zwei Quantisierungsebenen
einheitlich über jedes der mehreren Videobilder in einem Schachbrettmuster umfasst.
17. System gemäß Anspruch 12, wobei das Mittel zum Zuordnen Mittel zum Zuordnen von vier Quantisierungsebenen
 q_1, q_2, q_3, q_4 einheitlich über jedes der mehreren Videobilder derart umfasst, dass eine erste Quantisierungsebene
25 von jeder Reihe $j \bmod 4$ ist.
18. System gemäß Anspruch 14, wobei das Mittel zum Bestimmen einer Zielanzahl von Bits Mittel zum automatischen
Bestimmen der Zielanzahl von Bits umfasst.
- 30 19. System gemäß Anspruch 14, wobei das Mittel zum Bestimmen einer Zielanzahl von Bits Mittel zum Bestimmen der
Zielanzahl von Bits durch Benutzerintervention umfasst.
20. System gemäß Anspruch 14, wobei das Mittel zum Zuordnen ferner Mittel zum Zuordnen des ersten Satzes Quan-
tisierungsebenen basierend auf einer Aktivitätsmessung mehrerer Makroblöcke entsprechender Videobilder um-
35 fasst.
21. System gemäß Anspruch 12, wobei das Mittel zum Zuordnen Mittel zum Zuordnen mehrerer Quantisierungsebenen
einer Mehrheit der mehreren Videobilder umfasst.
- 40 22. System gemäß Anspruch 12, wobei das Mittel zum Zuordnen Mittel zum Zuordnen mehrerer Quantisierungsebenen
allen der mehreren Videorahmen umfasst.

Revendications

- 45 1. Procédé pour traiter des signaux vidéo inclus dans des trames vidéo afin de déterminer des caractéristiques des
trames vidéo, comprenant les étapes consistant à :
- a) obtenir des signaux vidéo inclus dans plusieurs trames vidéo ;
50 b) affecter plusieurs niveaux de quantification à une desdites plusieurs trames vidéo de sorte que des parties
différentes desdites plusieurs trames se voient affecter des niveaux de quantification différents ;
c) pré-encoder ladite une desdites plusieurs trames vidéo en utilisant lesdits plusieurs niveaux de quantification
afin d'obtenir plusieurs débits binaires correspondant à chacun desdits plusieurs niveaux de quantification ;
d) déterminer des débits binaires pour chacun desdits plusieurs niveaux de quantification résultant de ladite
55 étape de pré-encodage,
e) répéter les étapes b à d séquentiellement pour plusieurs trames vidéo suivantes ; et
f) encoder une pluralité desdites plusieurs trames vidéo après avoir déterminé les débits binaires pour chacune
desdites plusieurs trames vidéo ;

- g) déterminer un nombre de bits cible pour ladite pluralité desdites plusieurs trames vidéo ;
 h) affecter un premier ensemble de niveaux de quantification pour chaque trame de ladite pluralité desdites plusieurs trames vidéo en utilisant ledit nombre de bits cible et les plusieurs débits binaires déterminés ;
 i) encoder une trame vidéo de ladite pluralité desdites plusieurs trames vidéo en une trame encodée avec un nombre de bits résultant ;
 j) estimer un nombre de bits encodé pour une trame vidéo suivante ;
 k) ajuster le premier ensemble de niveaux de quantification pour ladite trame vidéo suivante sur la base dudit nombre de bits cible, desdits débits binaires pour des trames vidéo suivantes, du nombre de bits encodé estimé et du nombre de bits résultant ; et
 l) répéter les étapes i à k séquentiellement pour plusieurs trames vidéo suivantes.
2. Procédé selon la revendication 1, dans lequel l'étape consistant à obtenir des signaux vidéo comprend l'obtention desdits signaux vidéo à partir d'une caméra vidéo.
3. Procédé selon la revendication 1, comprenant en outre l'étape consistant à stocker les plusieurs débits binaires déterminés.
4. Procédé selon la revendication 1, dans lequel l'étape consistant à affecter comprend l'affectation desdits plusieurs niveaux de quantification uniformément à travers chacune desdites plusieurs trames vidéo.
5. Procédé selon la revendication 1, dans lequel l'étape consistant à affecter comprend l'affectation de deux niveaux de quantification uniformément à travers chacune desdites plusieurs trames vidéo selon un motif en damier.
6. Procédé selon la revendication 1, dans lequel l'étape consistant à affecter comprend l'affectation de quatre niveaux de quantification, q_1 , q_2 , q_3 , q_4 , uniformément à travers chacune desdites plusieurs trames vidéo de sorte qu'un premier niveau de quantification de chaque rangée, j , soit $q_{j \bmod 4}$.
7. Procédé selon la revendication 1, dans lequel l'étape consistant à déterminer un nombre de bits cible comprend la détermination du nombre de bits cible automatiquement.
8. Procédé selon la revendication 1, dans lequel l'étape consistant à déterminer un nombre de bits cible comprend la détermination du nombre de bits cible par l'intermédiaire d'une intervention de l'utilisateur.
9. Procédé selon la revendication 1, dans lequel l'étape consistant à affecter comprend en outre l'affectation dudit premier ensemble de niveaux de quantification sur la base d'une mesure d'activité de plusieurs macroblocs de trames vidéo correspondantes.
10. Procédé selon la revendication 1, dans lequel l'étape e) se répète pour une majorité de trames vidéo desdites plusieurs trames vidéo.
11. Procédé selon la revendication 1, dans lequel l'étape e) se répète pour toutes les trames vidéo desdites plusieurs trames vidéo.
12. Système pour traiter des signaux vidéo inclus dans des trames vidéo afin de déterminer des caractéristiques des trames vidéo, comprenant :
- des moyens pour obtenir des signaux vidéo inclus dans plusieurs trames vidéo ;
 des moyens pour affecter plusieurs niveaux de quantification à une première pluralité desdites plusieurs trames des signaux vidéo de sorte que des parties différentes desdites plusieurs trames se voient affecter des niveaux de quantification différents ;
 des moyens pour pré-encoder chaque trame vidéo de ladite première pluralité desdites plusieurs trames vidéo en utilisant lesdits niveaux de quantification afin d'obtenir plusieurs débits binaires correspondant à chacun desdits plusieurs niveaux de quantification ;
 des moyens pour déterminer lesdits plusieurs débits binaires pour chacun desdits plusieurs niveaux de quantification pour chaque trame vidéo de ladite première pluralité desdites plusieurs trames vidéo générée par lesdits moyens pour pré-encoder ; et
 des moyens pour encoder une seconde pluralité desdites plusieurs trames vidéo après avoir déterminé lesdits plusieurs débits binaires pour chaque trame vidéo de ladite première pluralité desdites plusieurs trames ;

des moyens pour déterminer un nombre de bits cible pour ladite seconde pluralité desdites plusieurs trames vidéo ; et

des moyens pour affecter un premier ensemble de niveaux de quantification pour chaque trame de ladite seconde pluralité desdites plusieurs trames vidéo en utilisant ledit nombre de bits cible et les plusieurs débits binaires déterminés ;

des moyens pour encoder une trame vidéo de ladite seconde pluralité desdites plusieurs trames vidéo en une trame vidéo encodée avec un nombre de bits résultant ;

des moyens pour estimer un nombre de bits encodé pour une trame vidéo suivante ; et

des moyens pour ajuster le premier ensemble de niveaux de quantification pour ladite trame vidéo suivante sur la base dudit nombre de bits cible, desdits débits binaires et du nombre de bits encodé estimé pour la trame vidéo suivante et du nombre de bits résultant de ladite une de ladite seconde pluralité de trames.

13. Système selon la revendication 12, dans lequel lesdits moyens pour obtenir comprennent des moyens pour obtenir lesdits signaux vidéo à partir d'une caméra vidéo.

14. Système selon la revendication 12, comprenant en outre des moyens pour stocker les débits binaires déterminés.

15. Système selon la revendication 12, dans lequel lesdits moyens pour affecter comprennent des moyens pour affecter lesdits plusieurs niveaux de quantification uniformément à travers chacune desdites plusieurs trames vidéo.

16. Système selon la revendication 12, dans lequel lesdits moyens pour affecter comprennent des moyens pour affecter deux niveaux de quantification à travers chacune desdites plusieurs trames selon un motif en damier.

17. Système selon la revendication 12, dans lequel lesdits moyens pour affecter comprennent des moyens pour affecter quatre niveaux de quantification, q_1 , q_2 , q_3 , q_4 , uniformément à travers chacune desdites plusieurs trames vidéo de sorte qu'un premier niveau de quantification de chaque rangée, j , soit $q_{j \bmod 4}$.

18. Système selon la revendication 14, dans lequel les moyens pour déterminer un nombre de bits cible comprennent des moyens pour déterminer le nombre de bits cible automatiquement.

19. Système selon la revendication 14, dans lequel les moyens pour déterminer un nombre de bits cible comprennent des moyens pour déterminer le nombre de bits cible par l'intermédiaire d'une intervention d'un utilisateur.

20. Système selon la revendication 14, dans lequel les moyens pour affecter comprennent en outre des moyens pour affecter ledit premier ensemble de niveaux de quantification sur la base d'une mesure d'activité de plusieurs macro-blocs de trames vidéo correspondantes.

21. Système selon la revendication 12, dans lequel les moyens pour affecter comprennent des moyens pour affecter plusieurs niveaux de quantification à une majorité desdites plusieurs trames vidéo.

22. Système selon la revendication 12, dans lequel les moyens pour affecter comprennent des moyens pour affecter plusieurs niveaux de quantification à la totalité desdites plusieurs trames vidéo.

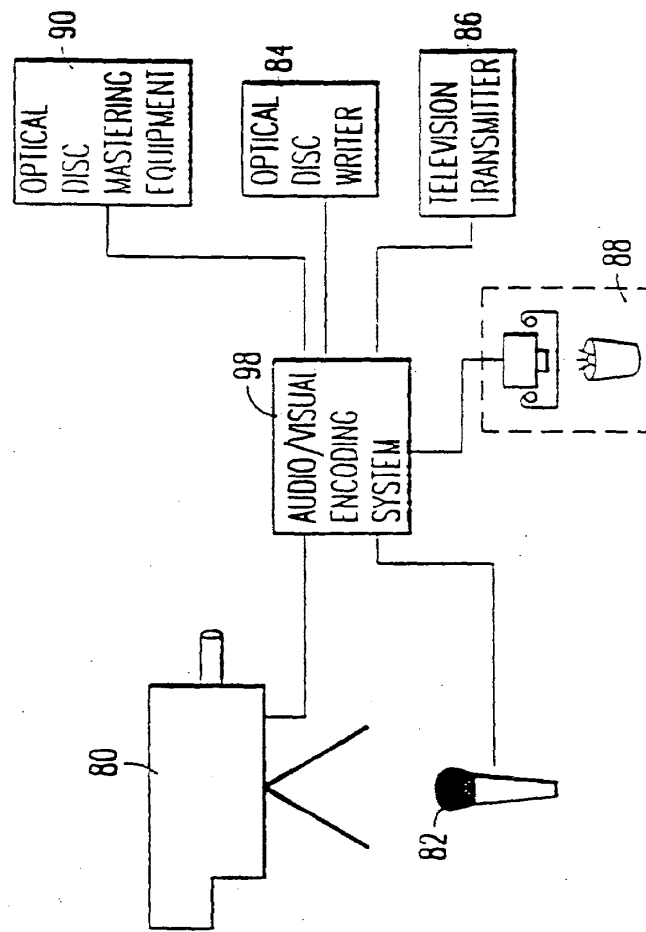


FIG. 1B

FIG. 2

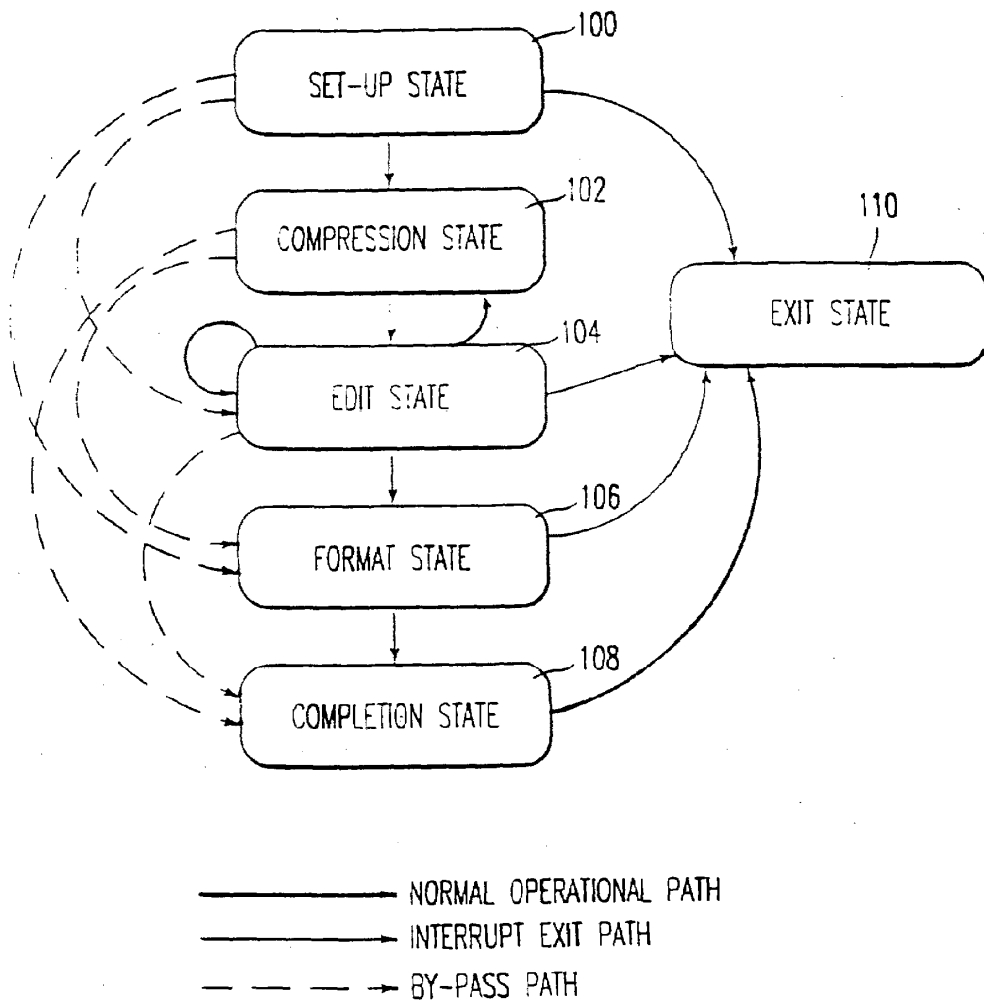


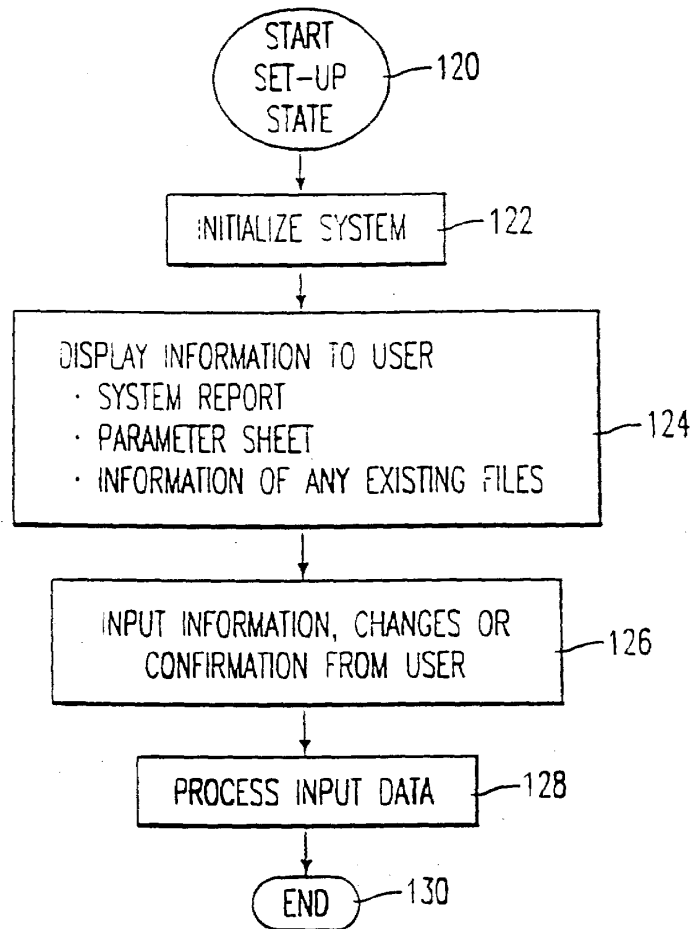
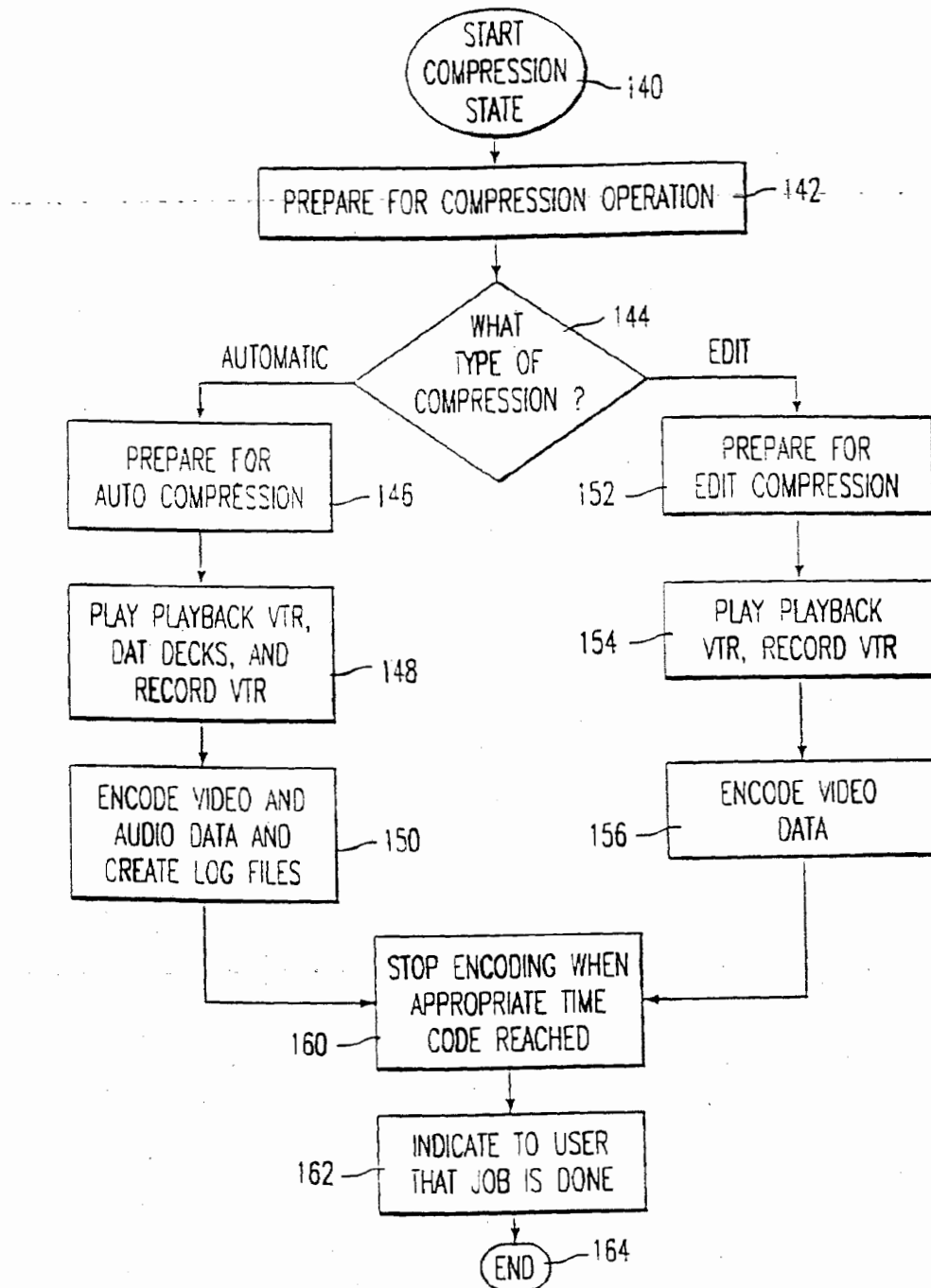
FIG. 3

FIG. 4



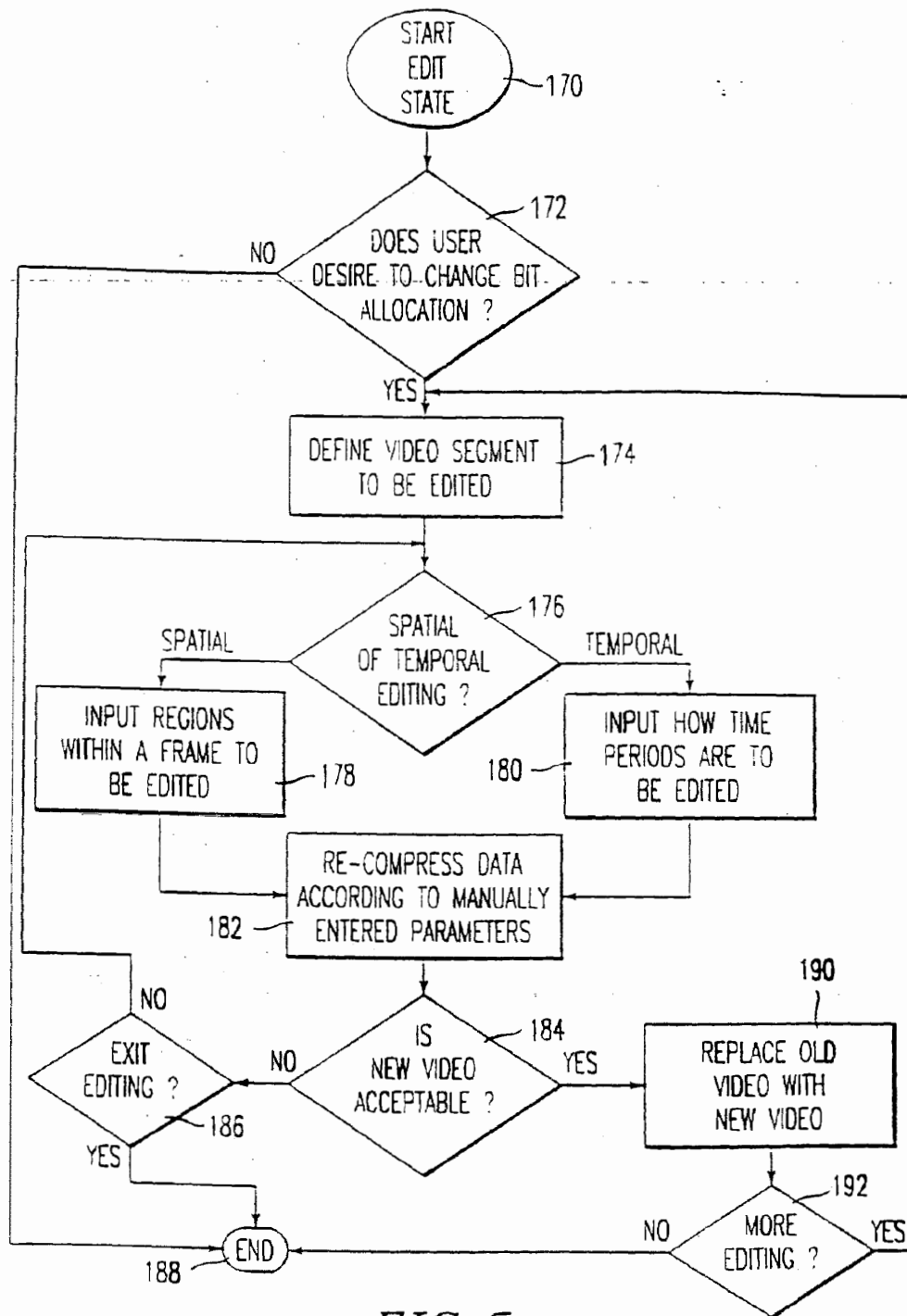


FIG. 5

FIG. 6

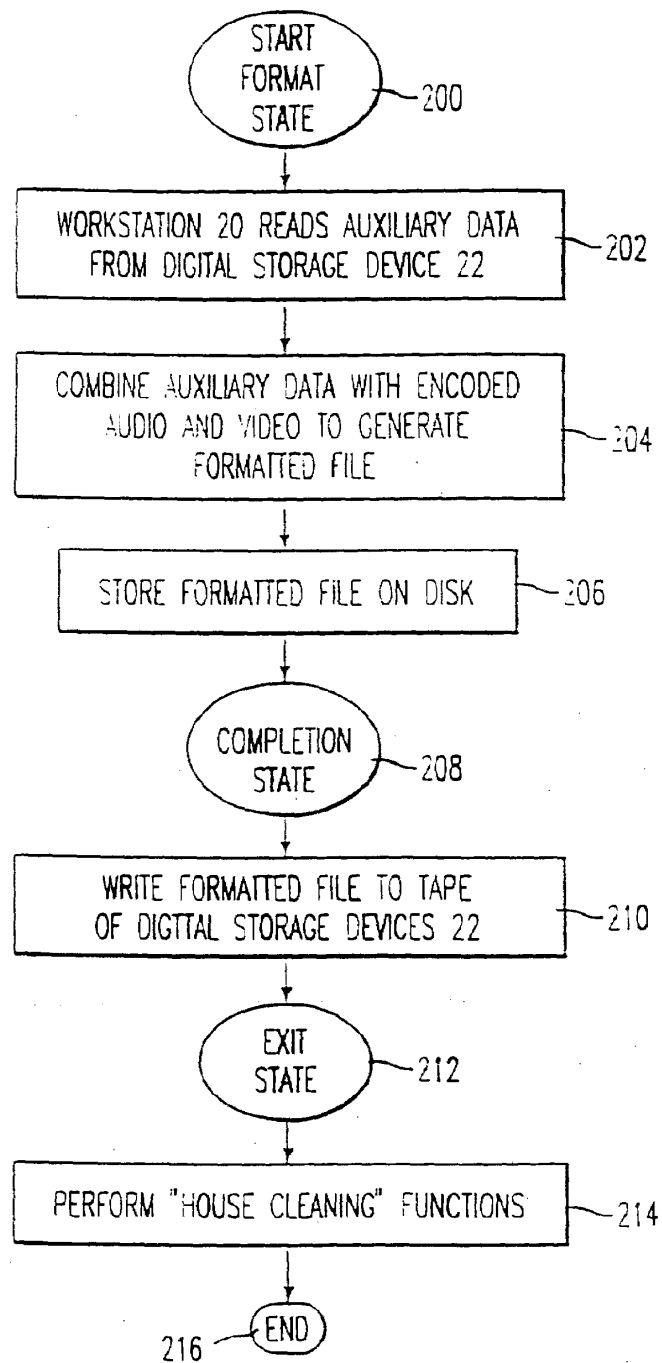
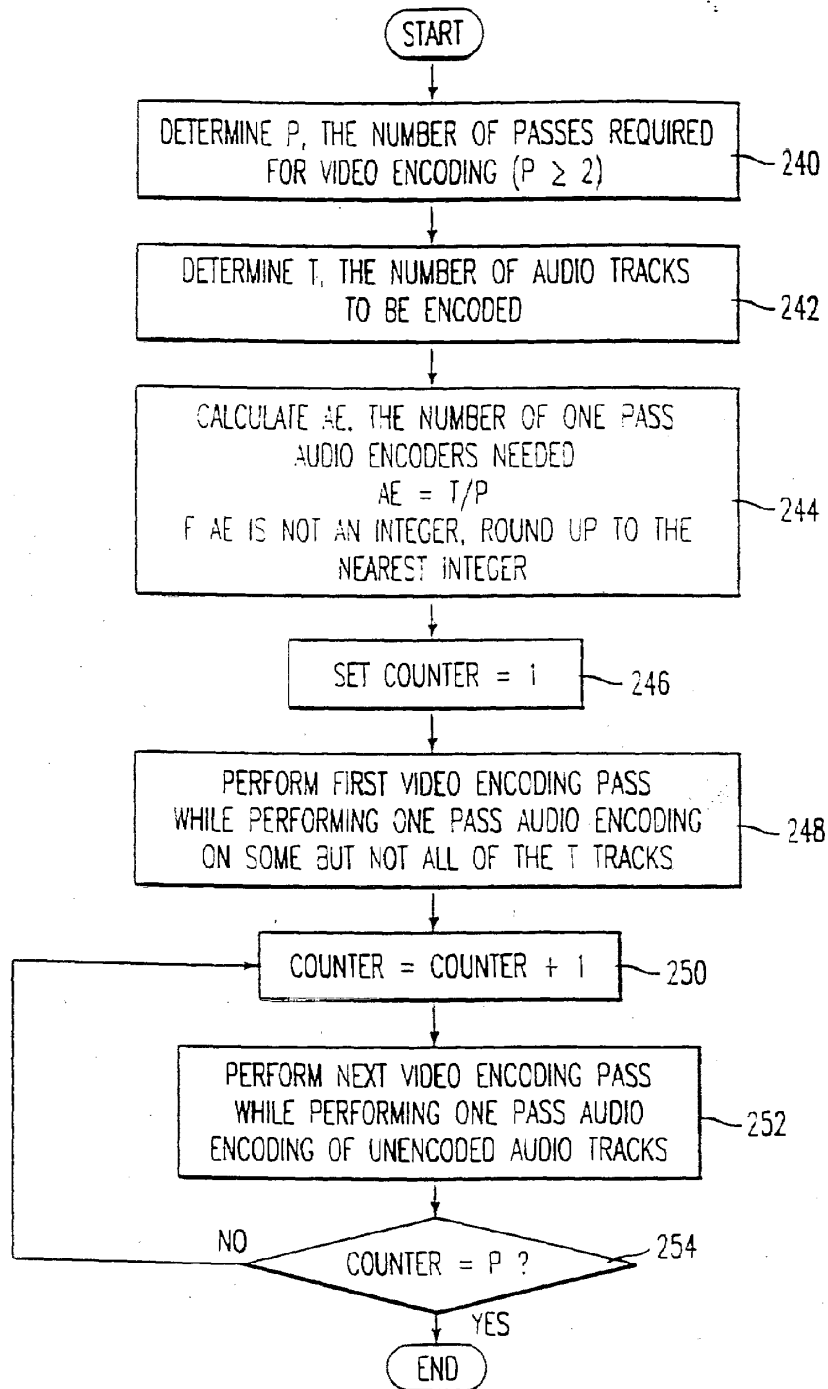


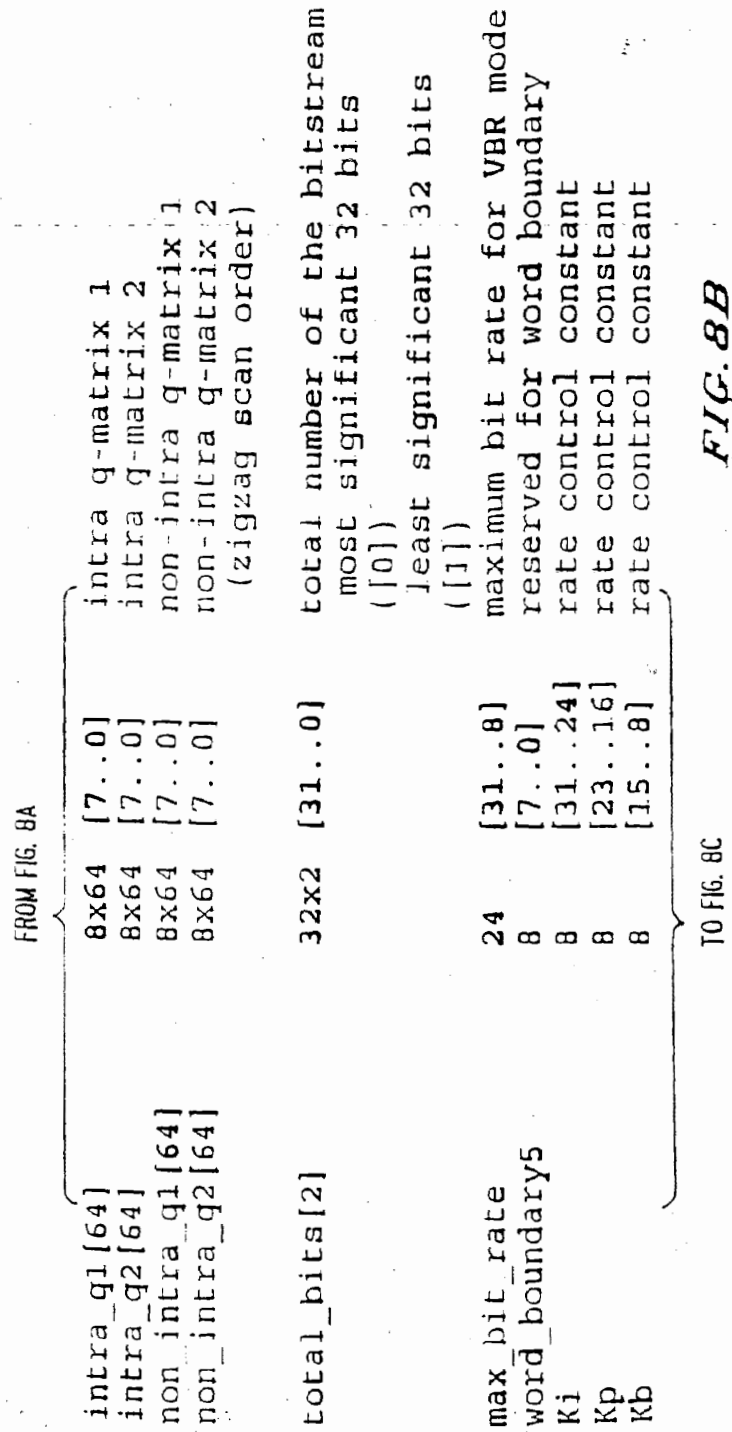
FIG. 7



	bits	bit field	contents
version	8	[31..28]	parameter format version
begin_tc	24	[23..0]	begin timecode
end_tc	24	[31..8]	end timecode
ntsc_pal	1	[7..7]	ntsc:0,pal:1
low_delay	1	[6..6]	lowdelay:1,else:0
enable_3_2	1	[5..5]	use 3 2 pull down; 1,else:0
MPEG1_2	1	[4..4]	MPEG1:0,MPEG2:1
EMode	1	[3..3]	fixed:0,adaptive:1
Mmax	2	[2..1]	maximum size of M
cbr_vbr	1	[0..0]	CBR:0,VBR:1
MB_hsize	6	[31..26]	MB width of the picture
MB_vsize	6	[25..20]	MB height of the picture
initial_delay	15	[19..6]	initial buffer fullness
word_boundary2	5	[5..0]	reversed for word boundary
bit_rate	24	[31..8]	average bit rate
m	2	[7..6]	If M is variable, M is 0.
n	6	[5..0]	If N is variable, N is 0.
VBV_size	24	[31..8]	VBV buffer size in bits
word_boundary4	8	[7..0]	reserved for word boundary
number_of_coded_picture	32	[31..0]	number of total coded pictures

TO FIG. 8B

FIG. 8A



FROM FIG. 8B

	bits	bit field	contents
rc_type	8	[7..0]	rate control type
Ki2	8	[31..24]	rate control constant
Kp2	8	[23..16]	rate control constant
Kb2	8	[15..8]	rate control constant
rc_type2	8	[7..0]	rate control type2
display_hsize	14	[31..18]	display horizontal size
display_vsize	14	[17..4]	display vertical size
word_boundary	4	[3..0]	reserved for word boundary
run_mode	2	[31..30]	running mode
			0: loop control
			1: timecode control
			2: timecode start and loop ctrl
loops	18	[29..12]	specified loops in loop control
pulldown_load_mode	2	[11..10]	the way to get pulldown phase
			0: auto, 1: from log,
			2: from data file
scene_change_adp_enable	1	[9..9]	1: scene change adaptation on
sequence_header_mode	1	[8..8]	1: before every GOP header
pan_scan_enable	1	[7..7]	1: enable
MB_parameter_mode	1	[6..6]	1: full, 0: short
buffer_check_enable	2	[5..4]	0: disable 1-3: enable

TO FIG. 8D

FIG. 8C

FROM FIG. 8C

frame_memory_channel	2	[3..2]	0 : CH1, 1 : CH2, 2 : through
timecode_check_enable	1	[1..1]	1 : enable
word_boundary76	1	[0..0]	reserved for word boundary
frame_memory_file_num	5	[31..27]	frame memory file number
initial_pull_down_th	18	[26..9]	initial pull-down threshold
word_boundary77	9	[8..0]	reserved for word boundary
scene_change_threshold	32	[31..0]	for scene change adaptation
pull_down_phase_threshold_file	8x256		file name for pull-down phase
MB_ctrl_file	8x256		file name for MB control
pan_scan_file	8x256		file name for pan-scan data
VBR_parameter_file	8x256		file name for VBR parameter
M_sequence_file	8x256		file name for M-sequence

FIG. 8D

FIG. 9A

time_code	bits	bit field	contents
picture_coding_type	24	[31..8]	timecode
top_field_first	2	[7..6]	picture coding type
repeat_first_field	1	[5..5]	top field first
display_repeat_field	1	[4..4]	repeat first field
	1	[3..3]	repeat number of displaying picture
load_intra_qmat	1	[2..2]	load intra quantiser matrix
load_non_intra_qmat	1	[1..1]	load non-intra quantiser matrix
pan_scan	1	[0..0]	use
			picture_display_extension()
input_order	32	[31..0]	input order of the coded picture
prefil_coeff	4	[31..28]	prefilter coefficients index
nr_coeff	2	[27..26]	NR coefficients index
mc_rom_type	2	[25..24]	mc_rom_type index
rlc_zero_type	2	[23..22]	rlc_zero_type index
vbv_occupancy	21	[21..1]	VBV occupancy Bn* in MPEG2 DIS
MEmode_type	1	[0..0]	interlace:0, non-interlace:1
forward_ae	23	[31..9]	ME error for forward reference
mean_qscale	8	[8..1]	mean qscale x 4
processing_type	1	[0..0]	4:2:2 → 4:2:0 filter type
			0:field / 1:frame

TO FIG. 9B

FROM FIG. 9A

backward_ae	23	[31..9]	NE error for backward reference
intra_q_matrix_type	1	[8..8]	intra quantizer matrix type
non_intra_q_matrix_type	1	[7..7]	non-intra quantizer matrix type
intra_dc_precision	2	[6..5]	intra DC precision
q_scale_type	1	[4..4]	qscale type
concealment_MVs	1	[3..3]	concealment MV
intra_vlc_format	1	[2..2]	intra VLC format
alternate_scan	1	[1..1]	alternate scan
progressive_frame	1	[0..0]	progressive frame
allocated_bit	21	[31..11]	allocated bit for the picture
temporal_reference	10	[10..1]	temporal reference
scene_change	1	[0..0]	1 : scene change
generated_bit_picture	21	[31..11]	generated bit for the picture
word_boundary6	11	[10..0]	reserved for word boundary
bits_sequence_header	12	[31..20]	If exists for the picture, sequence header bit
GOP_number	20	[19..0]	GOP number
bits_GOP_header	12	[31..20]	If exists for the picture, GOP header bit

TO FIG. 9C

FIG. 9B

FROM FIG. 9B		zero stuffing size of the picture	
stuffing_size	20	[19..0]	
bits_picture_header	12	[31..20]	picture header bit
bits_MB_header	20	[19.0]	slice header, MB header, DC, and EOB bit
word_boundary10	2	[31..30]	reserved for word boundary
mean_activity	30	[29..0]	mean activity
mc_error	32	[31..0]	MC error
coding_error	32	[31..0]	coding error
S1_I	32	[31..0]	rate control status
S2_I	32	[31..0]	rate control status
S1_P	32	[31..0]	rate control status
S2_P	32	[31..0]	rate control status
S1_B	32	[31..0]	rate control status
S2_B	32	[31..0]	rate control status
Qsum	32	[31..0]	sum of Q
reserved(4)	32	[31..0]	

FIG. 9C

MB parameters	bits	bit field	contents
word_boundary0	2	[31..30]	reserved for word boundary
MB_type	5	[29..25]	macroblock type
MC_type	2	[24..23]	MC type
DCT_type	1	[22..22]	DCT type
header_length	8	[21..14]	MB header, DC bits
code_length	14	[13..0]	RLC bits
word_boundary2	1	[31..31]	reserved for word boundary
MB_act	15	[30..16]	MB activity
mc_error_MB	16	[15..0]	MC error
q_scale	5	[31..27]	quantizer scale index
reserved	27	[26..0]	reserved for word boundary

FIG. 10A

MB parameters	bits	bit field	contents
MB_type	5	[31..27]	macroblock type
header_length	8	[26..19]	MB header, DC bits
code_length	14	[18..5]	RLC bits
q_scale	5	[4..0]	quantizer scale index

FIG. 10B

FIG. 11

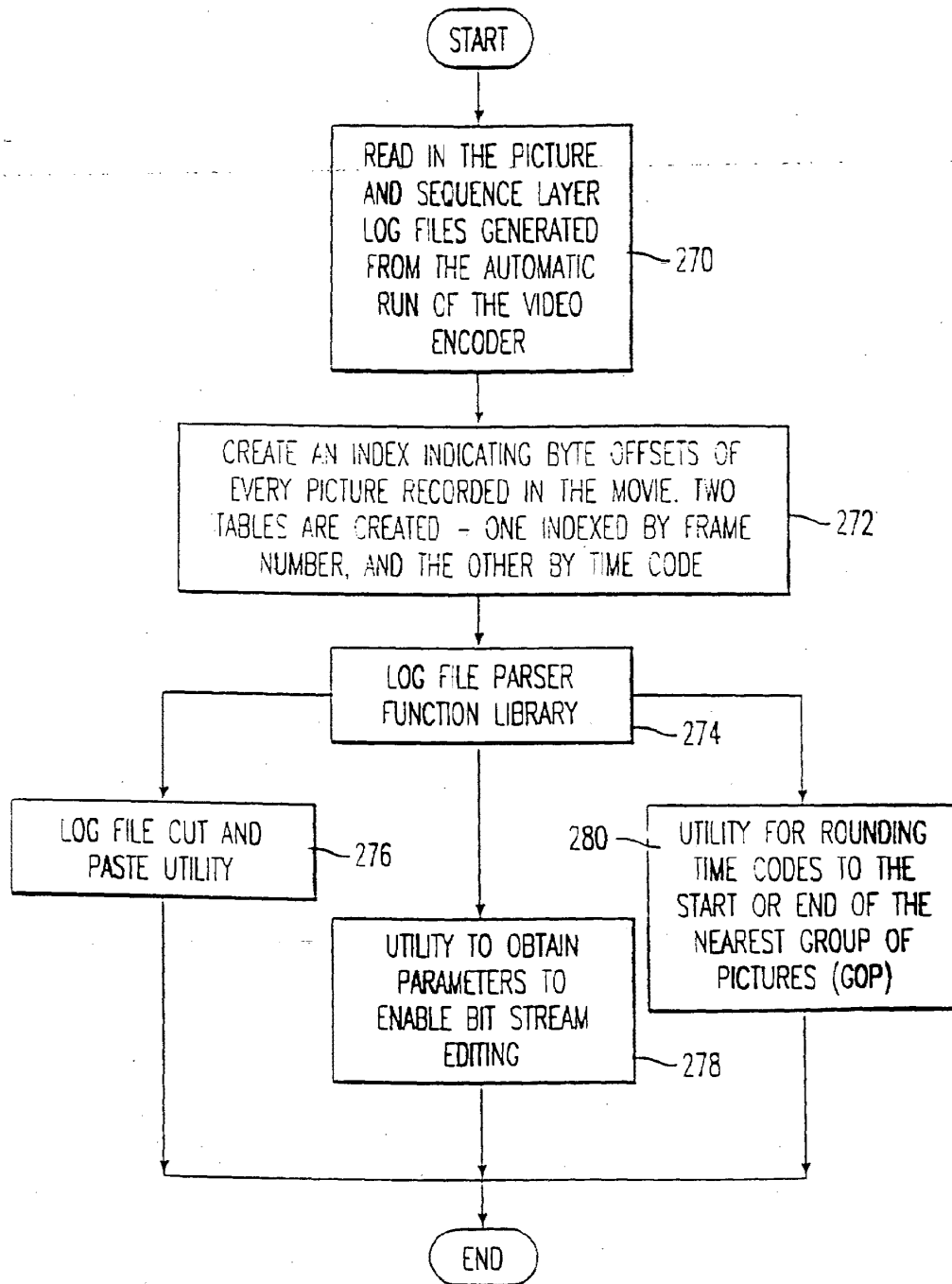


FIG. 12

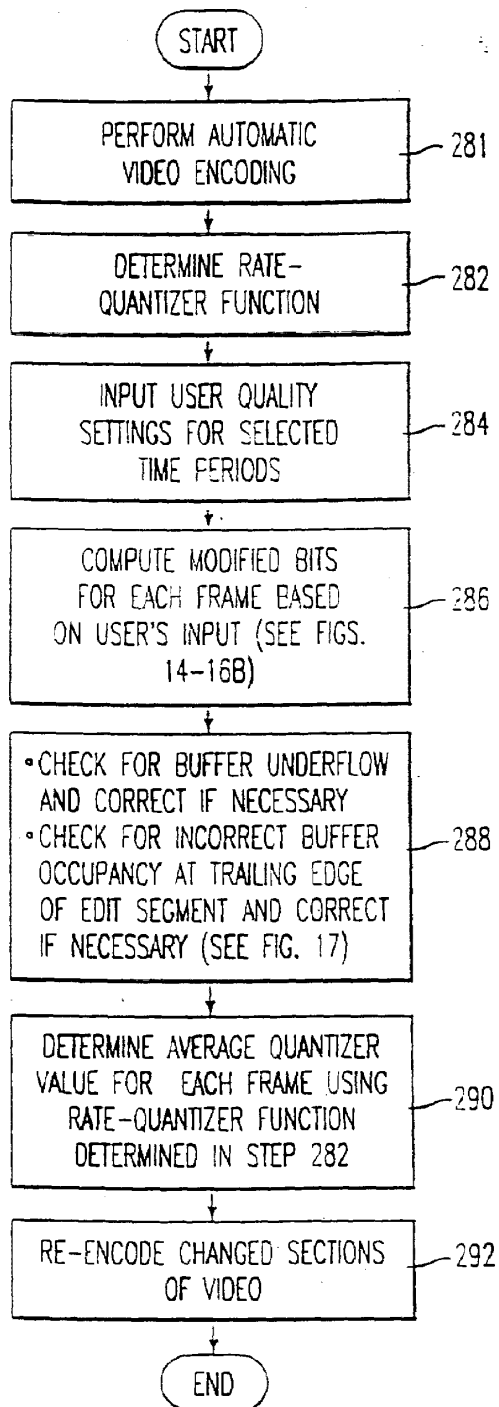


FIG. 13

IN	OUT	PRIORITY
00:00:50:00	00:01:00:00	P
00:01:00:01	00:02:00:00	-1
00:02:07:00	00:03:20:00	-1
00:04:00:00	00:04:50:00	3

IN	00:00:00:00	OUT	00:00:00:00	PRIORITY	P
MINIMUM BIT RATE		5.678	BACKGROUND PRIORITY		P

INSERT	RECALL	UPDATE	DELETE
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FIG. 14

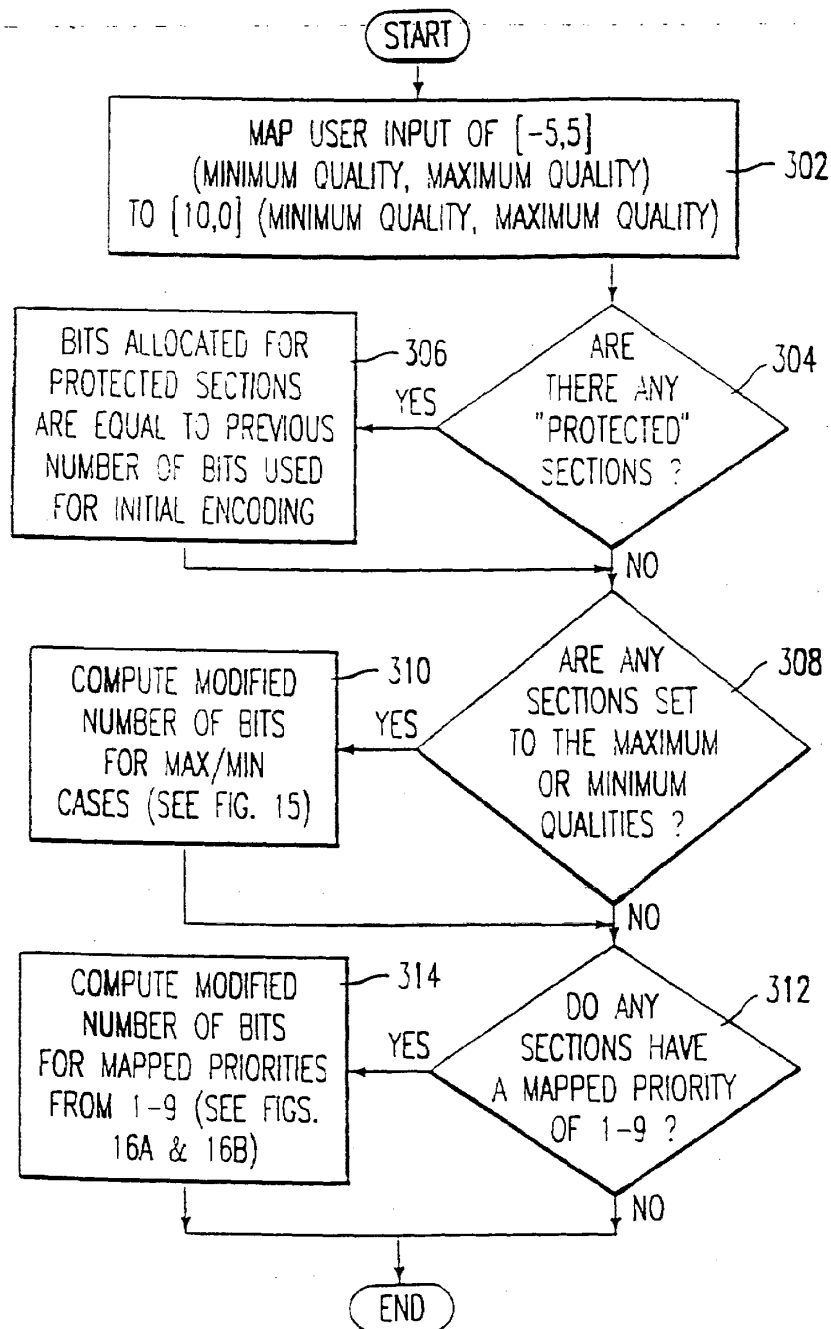


FIG. 15

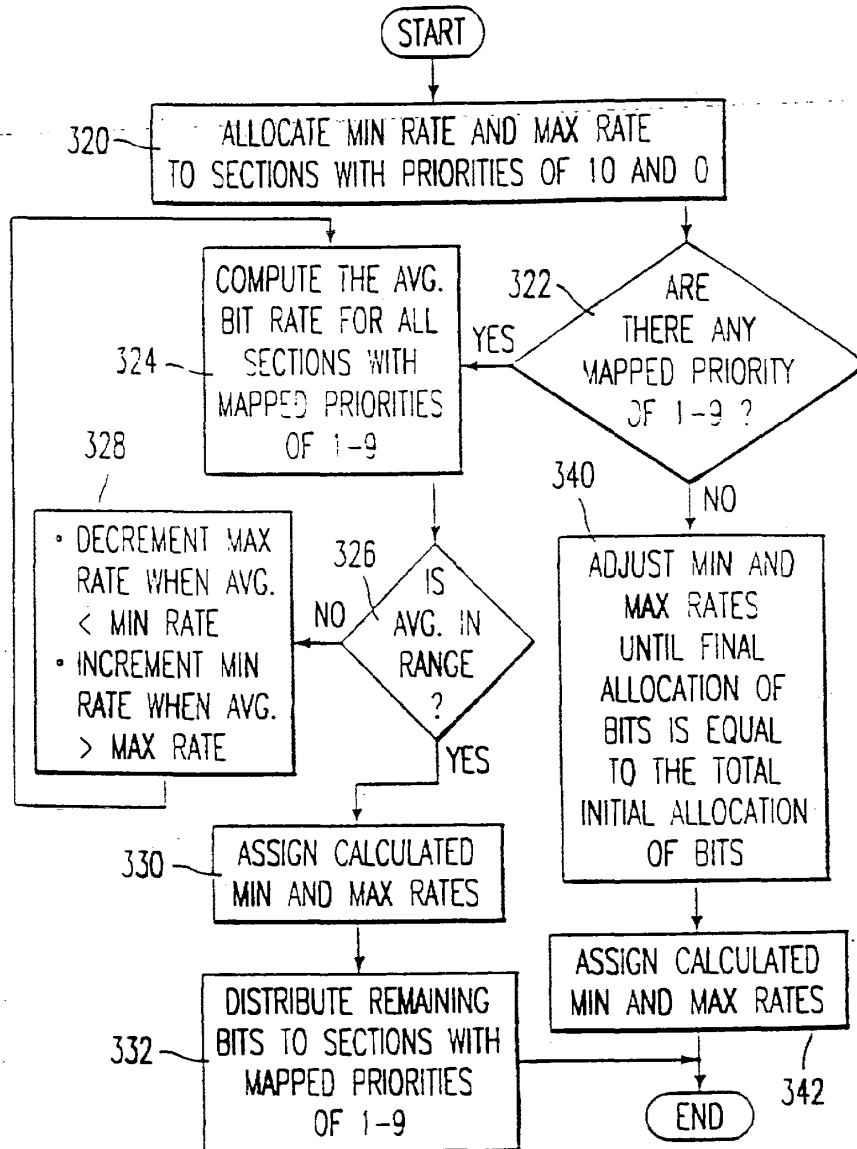


FIG. 16A

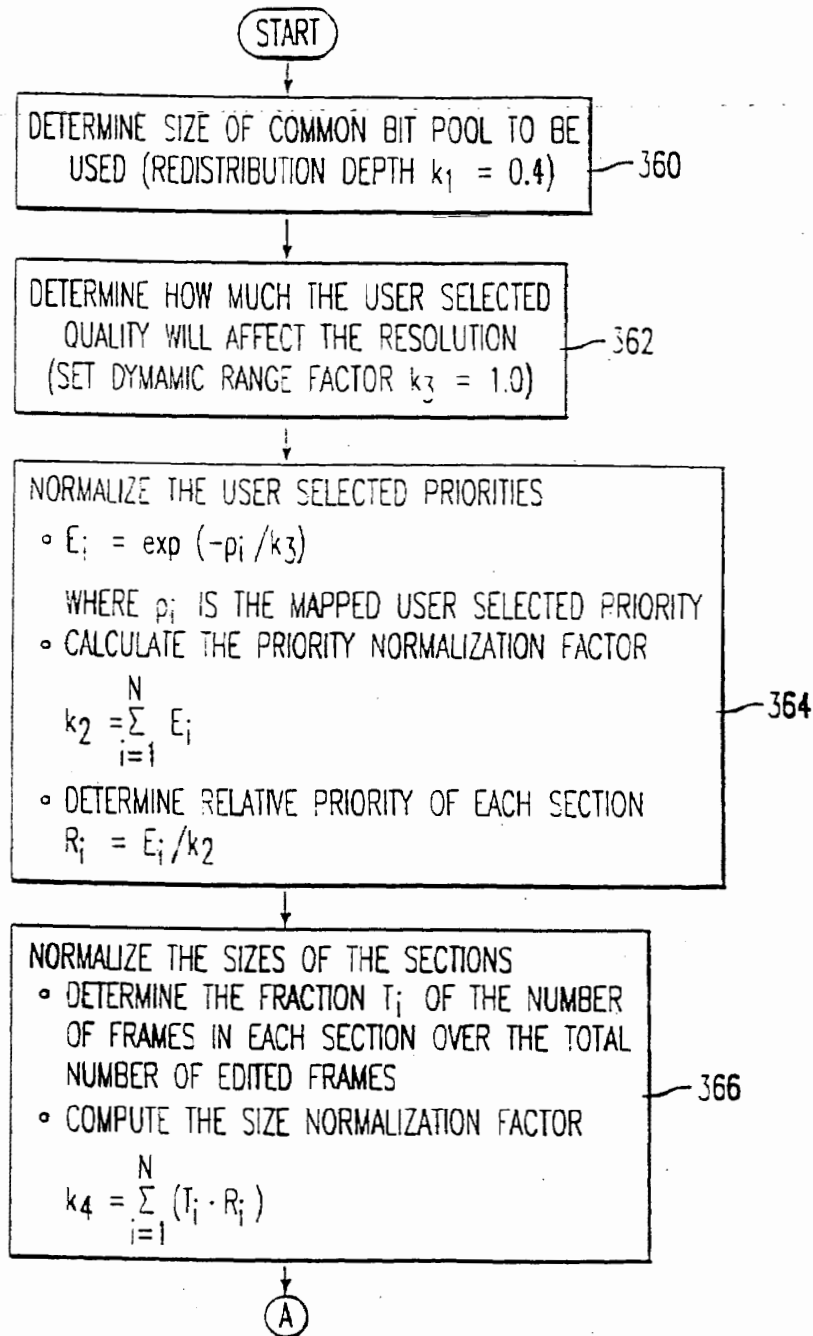


FIG. 16B

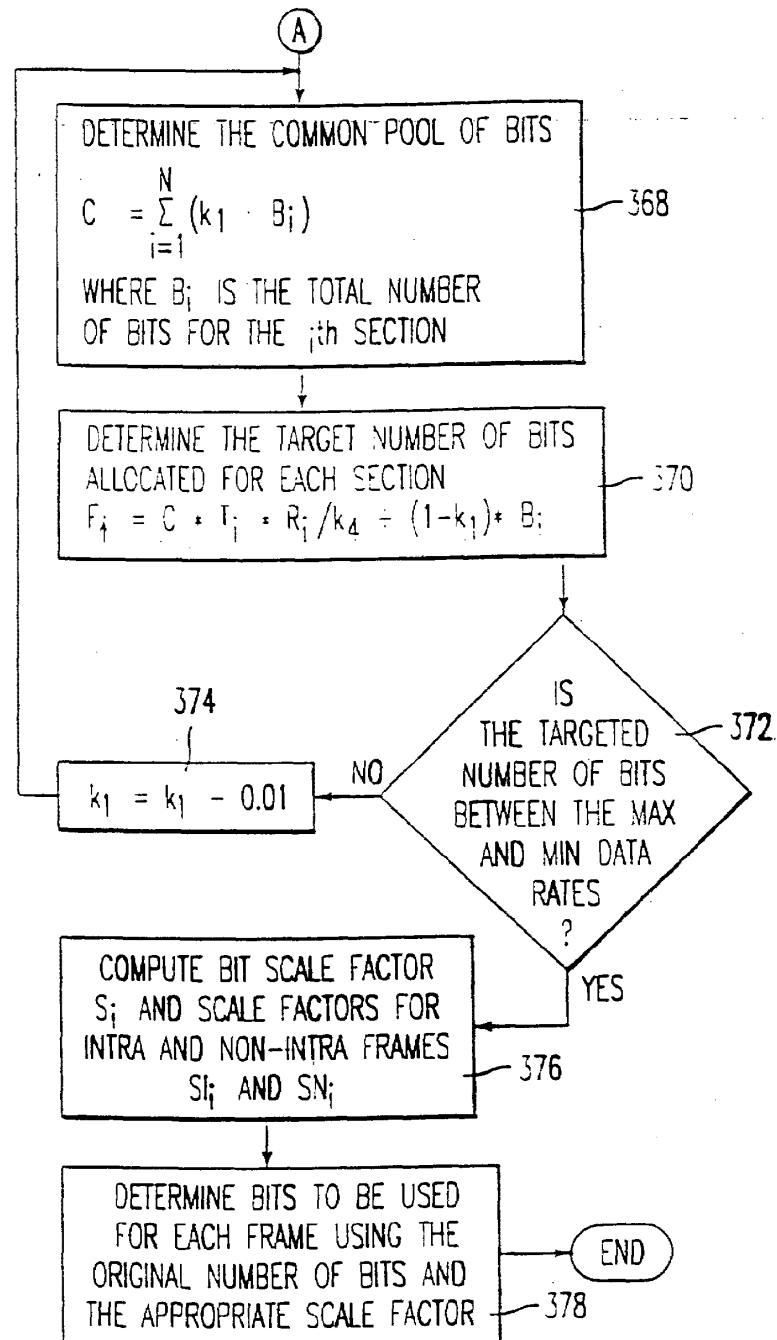
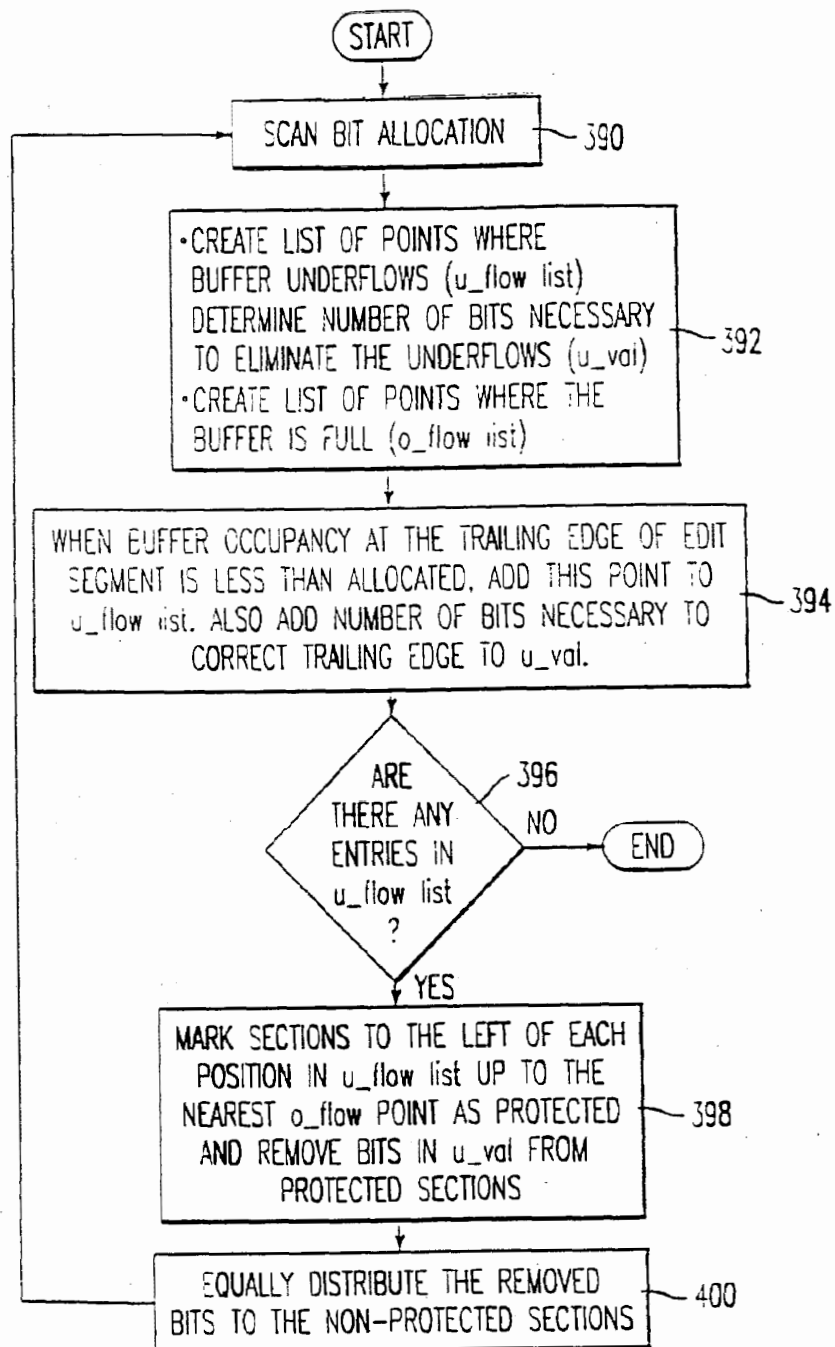
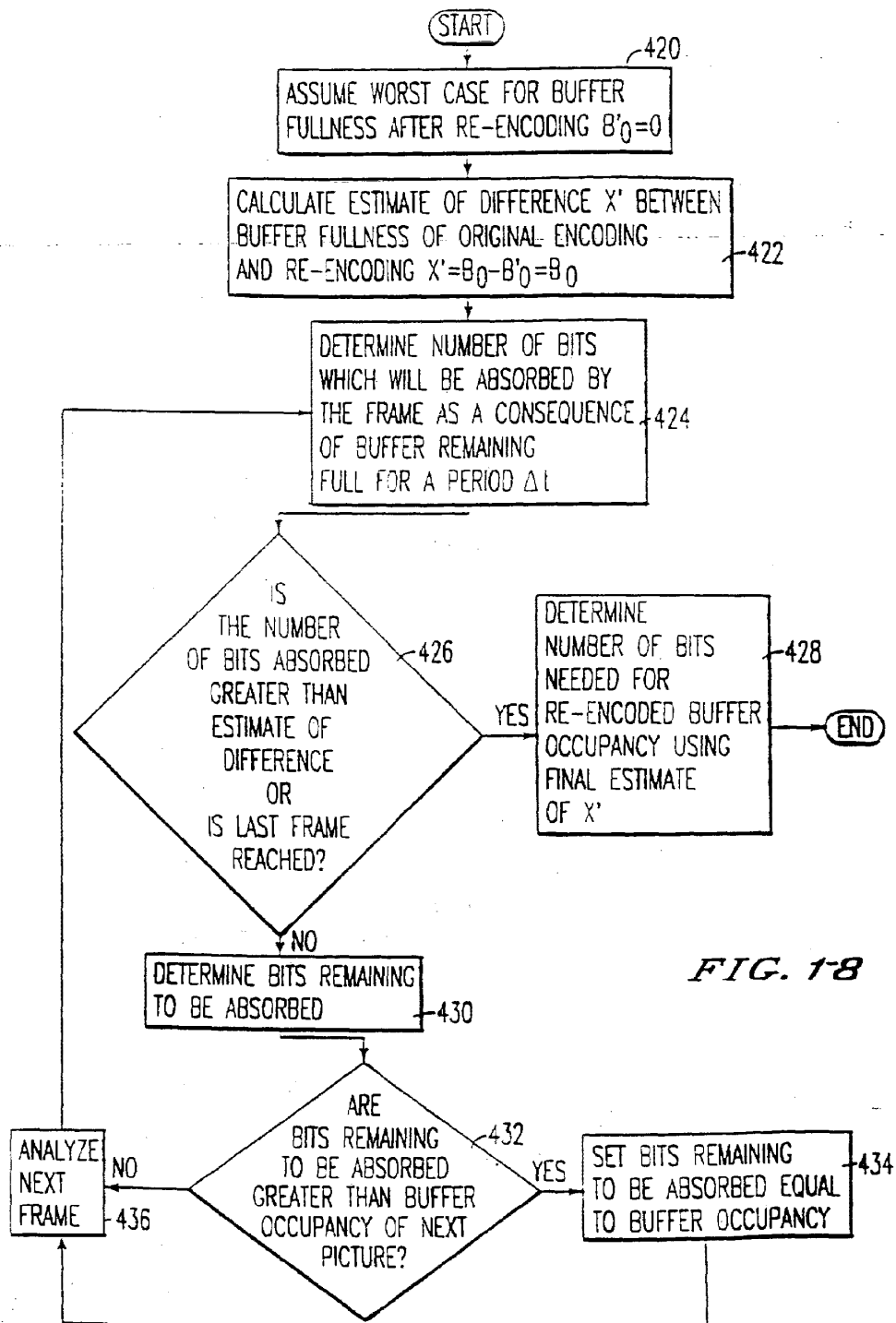


FIG. 17





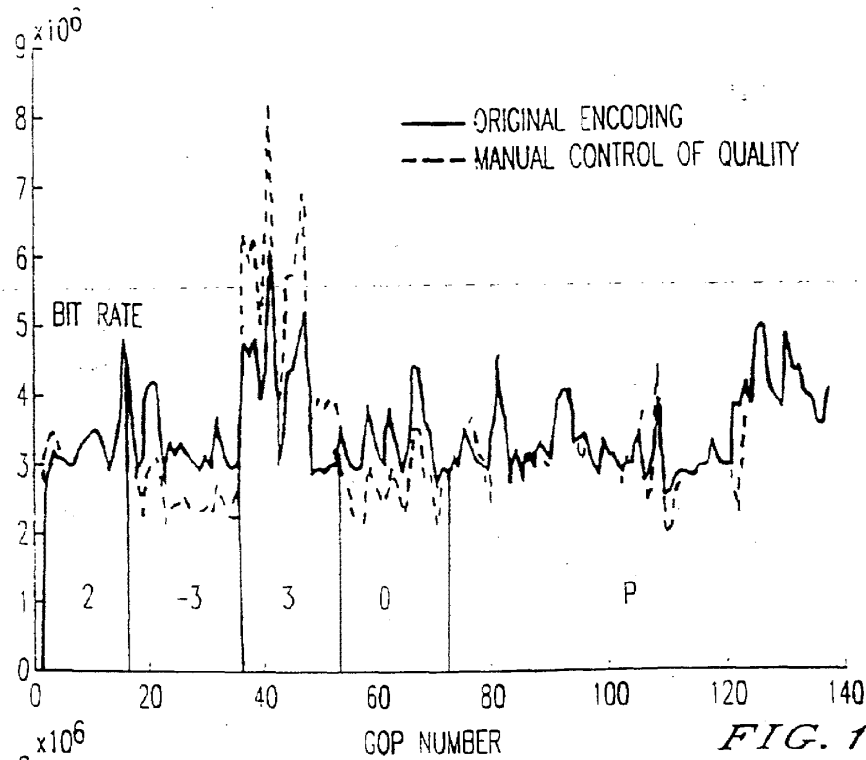


FIG. 19A

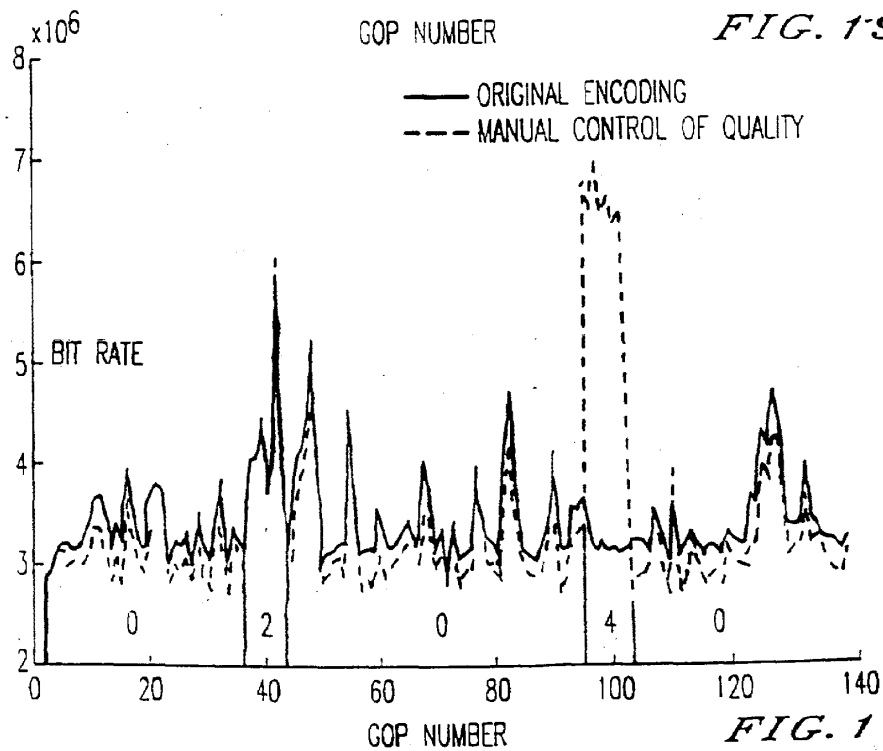
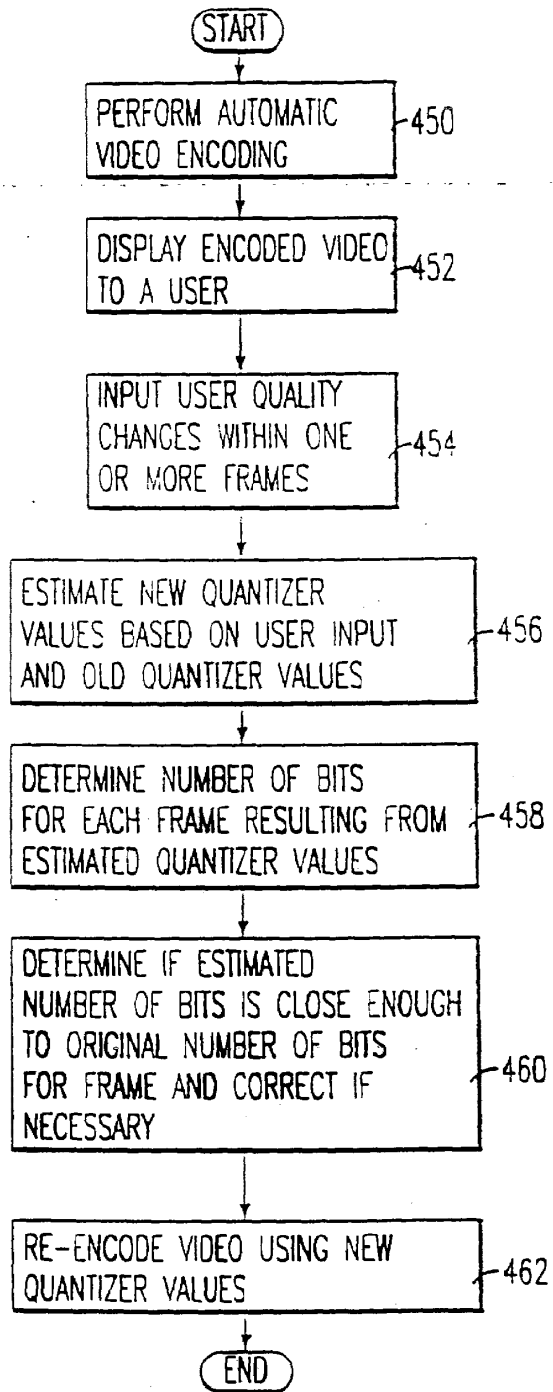


FIG. 19B



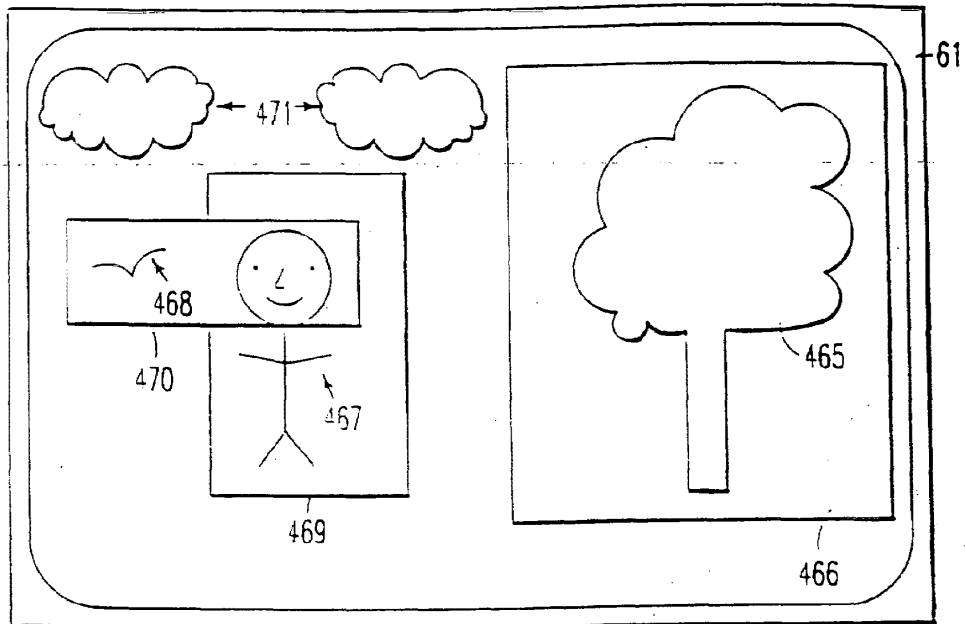


FIG. 21

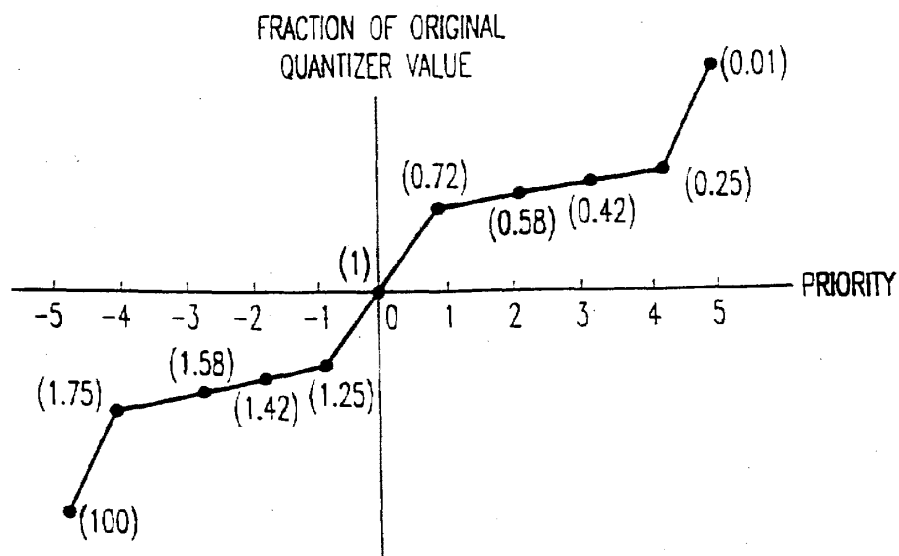


FIG. 22

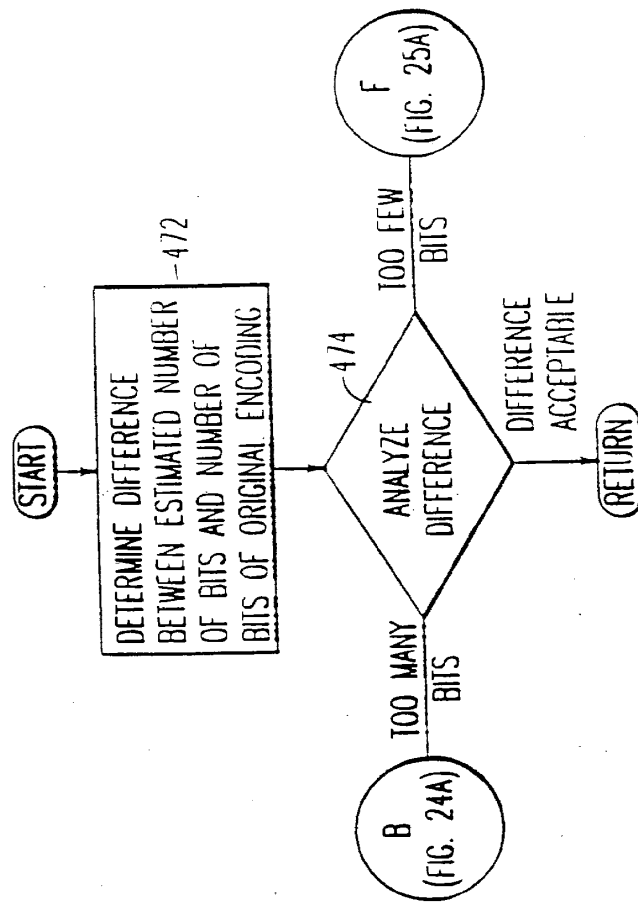
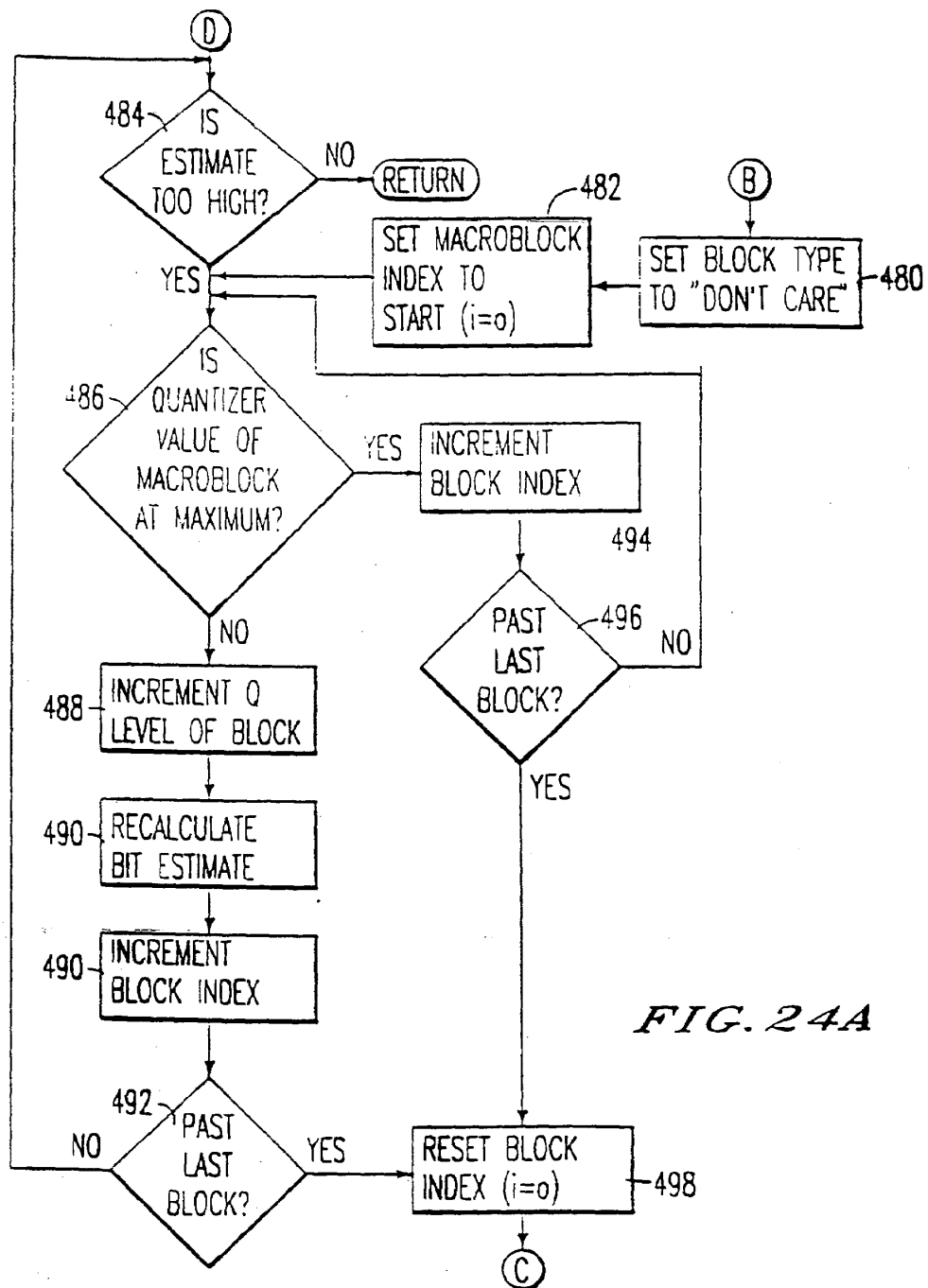


FIG. 23



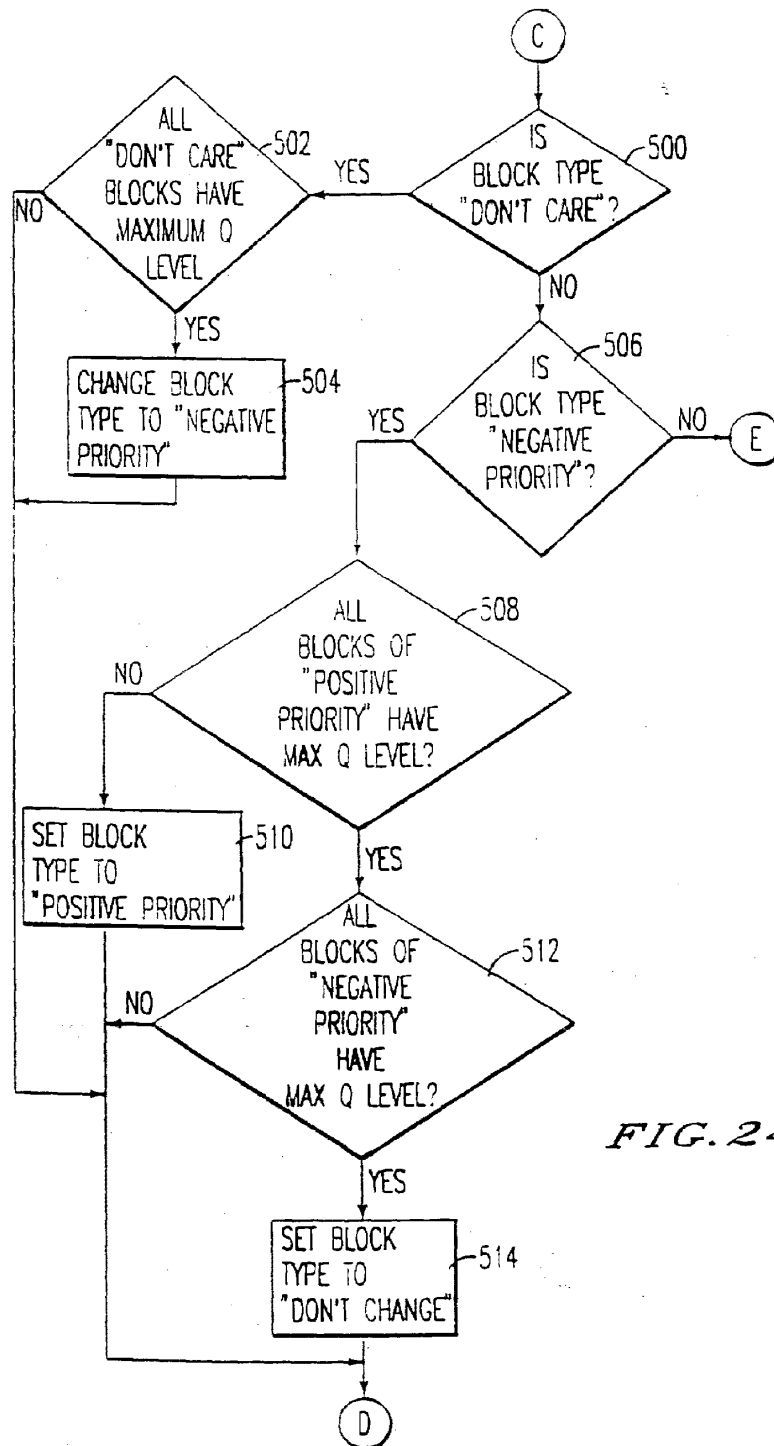
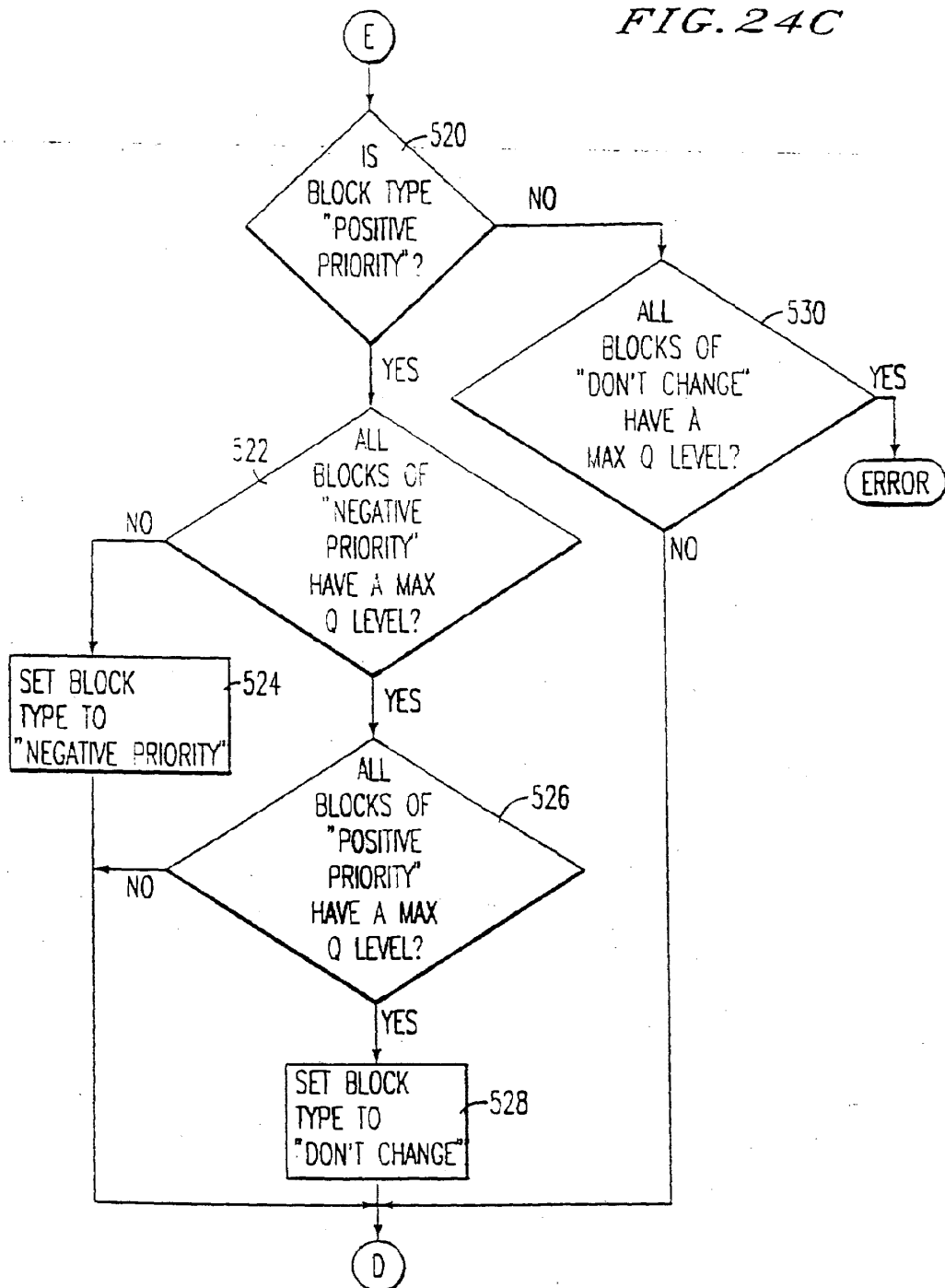


FIG. 24B

FIG. 24C



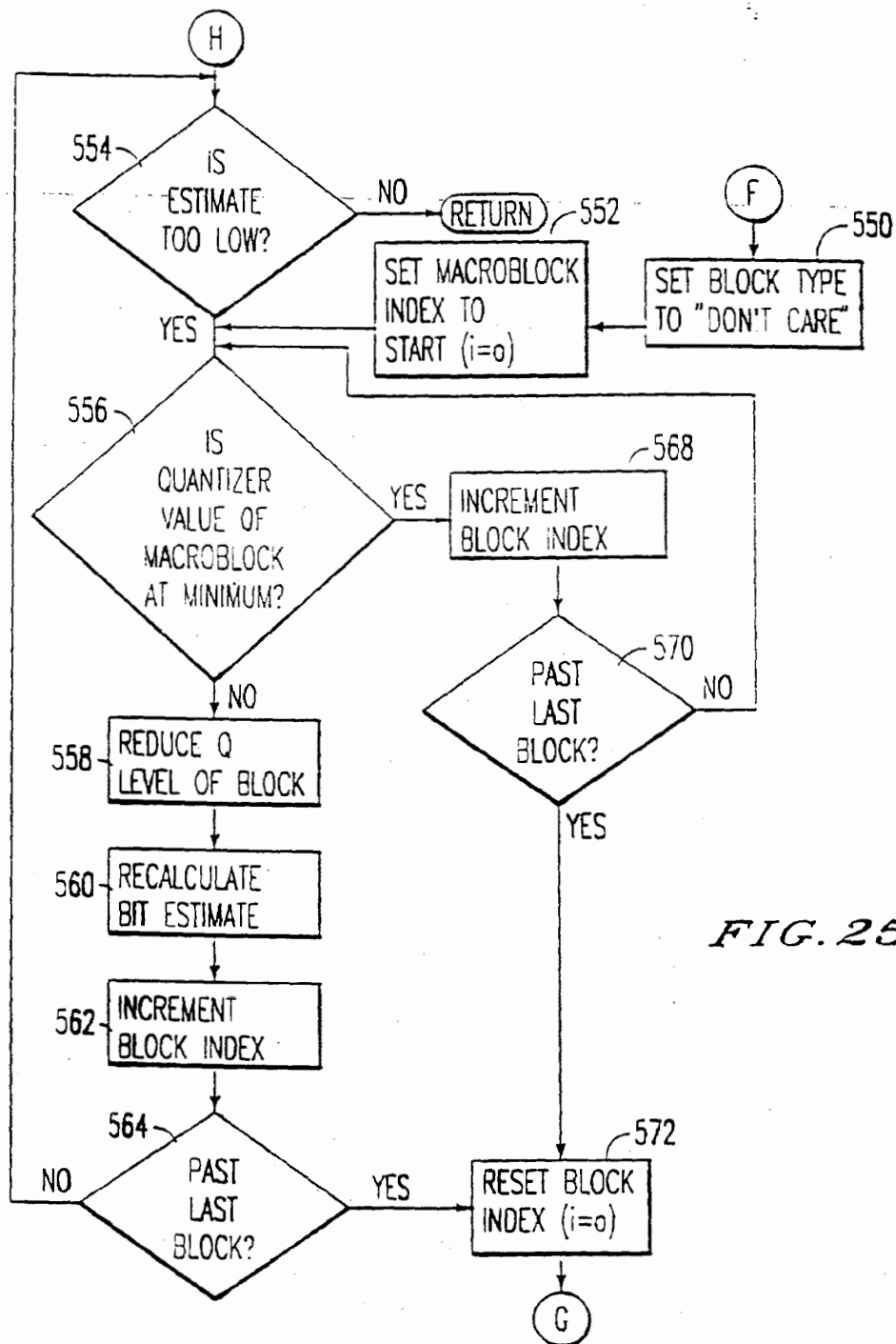


FIG. 25A

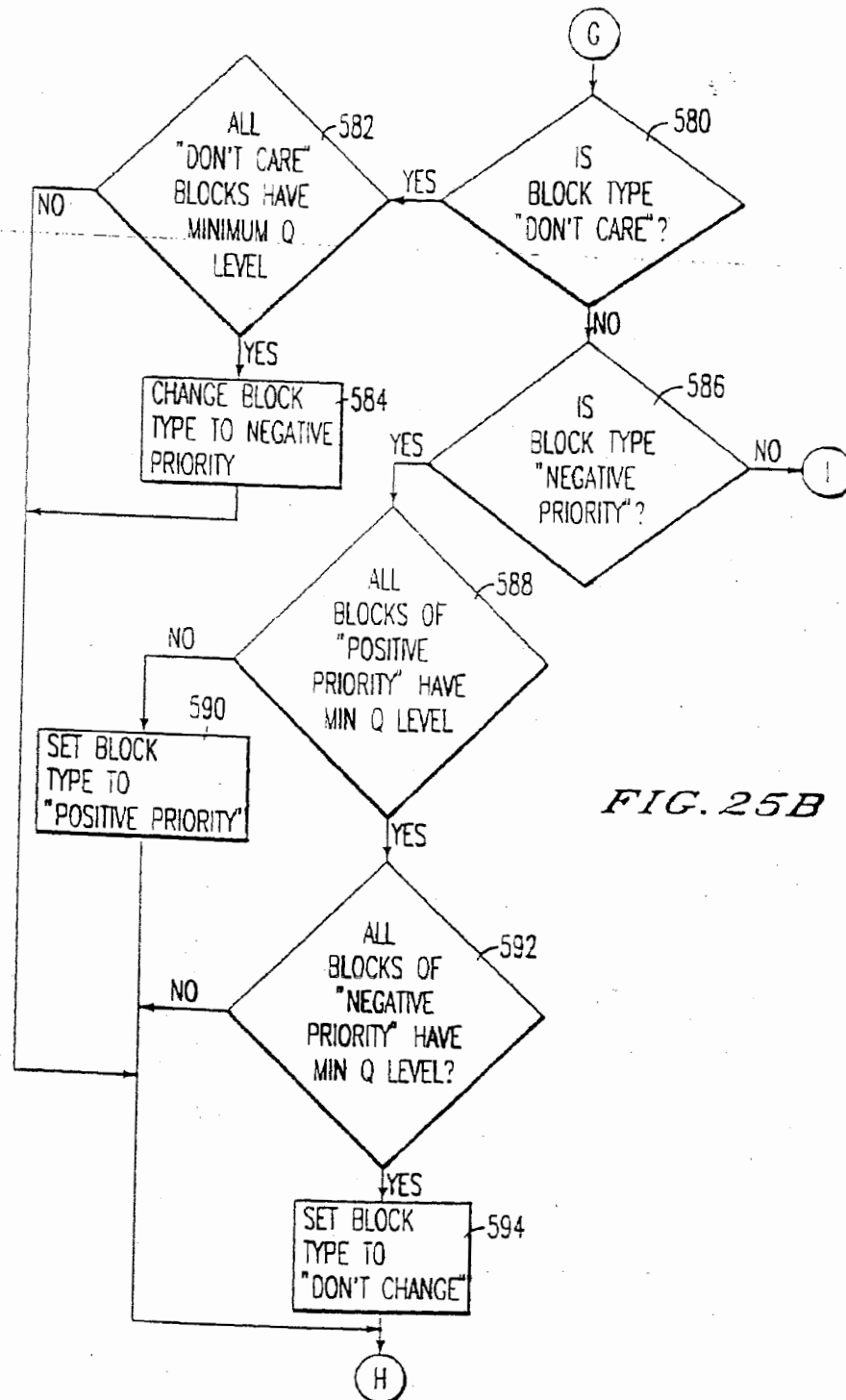
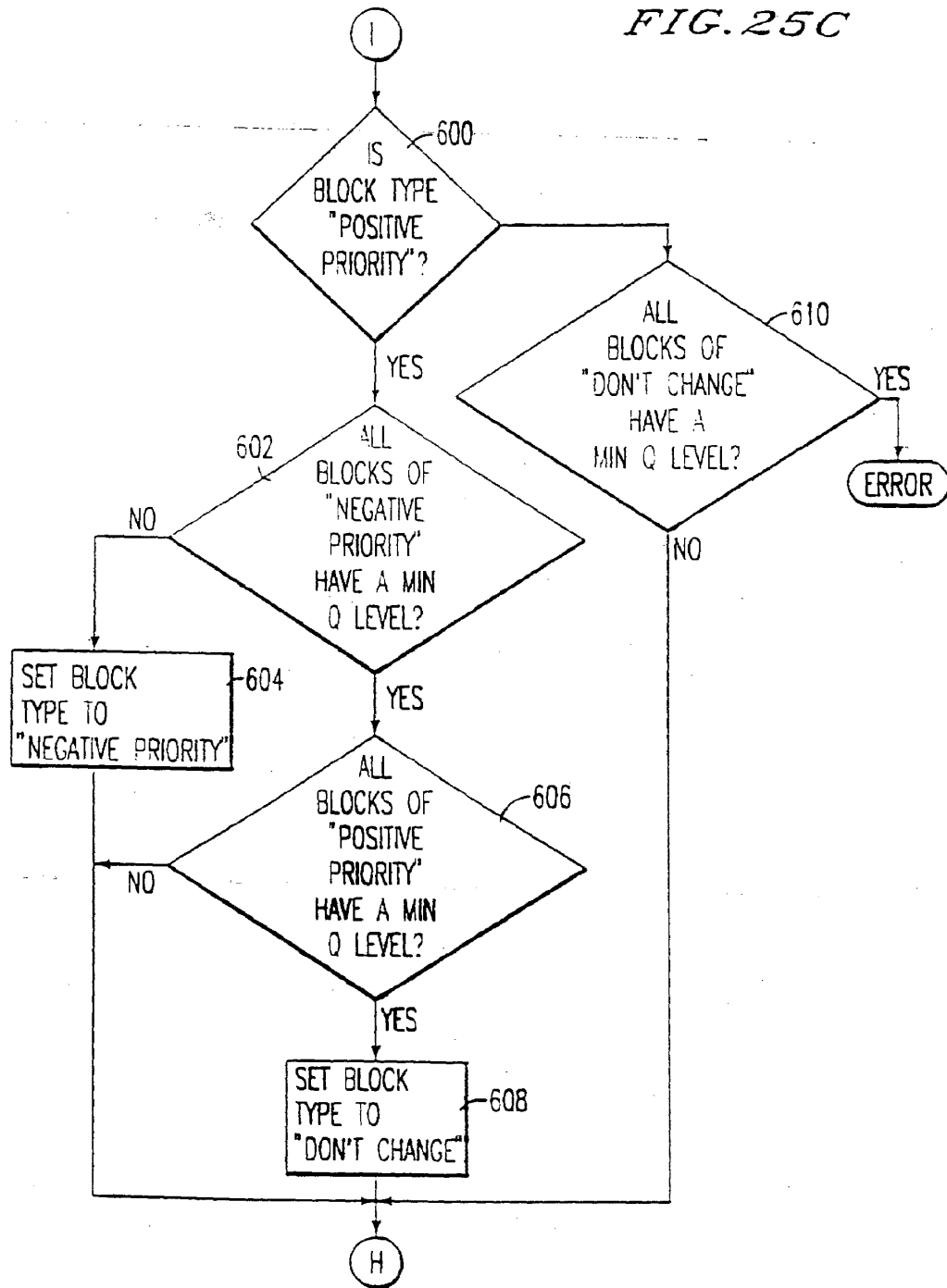


FIG. 25C



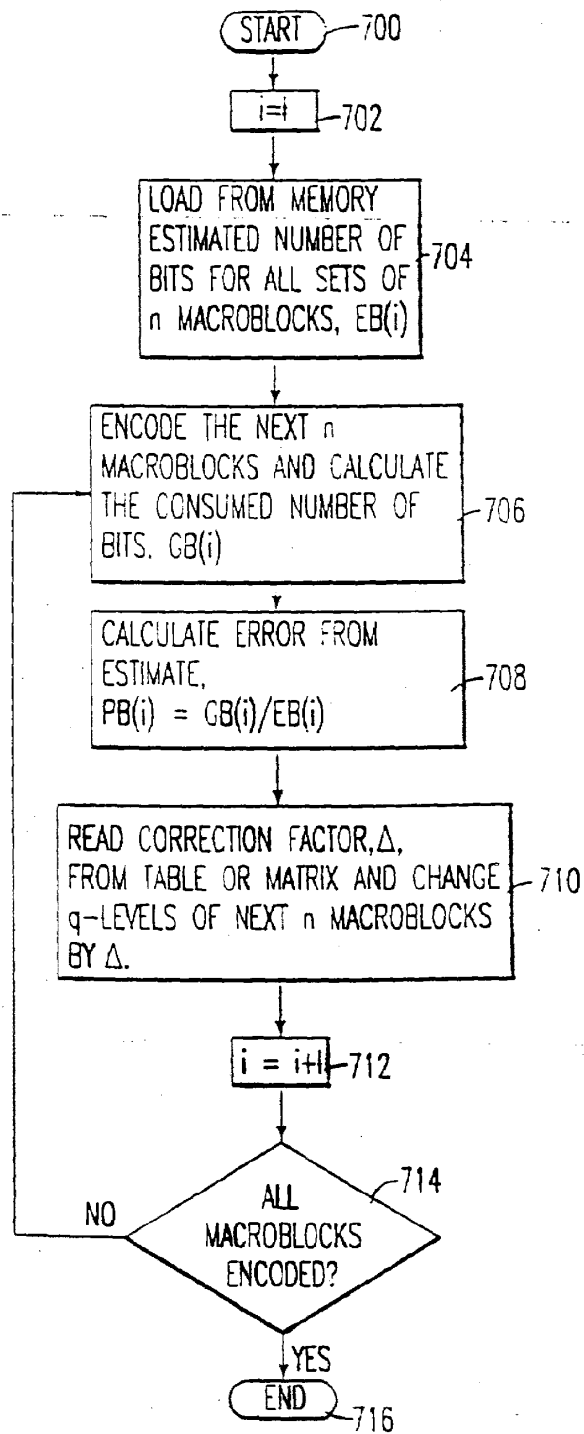


FIG. 26

$q_{i,j}$	$q_{i+1,j}$	$q_{i+n-2,j}$	$q_{i+n-1,j}$	$q_{i+n,j}$
$q_{i,j+1}$	$q_{i+1,j+1}$			$q_{i+n,j+1}$

FIG. 27A

$q_{i,j}$	$q_{i+1,j}$		$q_{i+n,j}$
$q_{i,j+1+\Delta 1}$	$q_{i+1,j+1+\Delta 1}$		$q_{i+n,j+1+\Delta 1}$

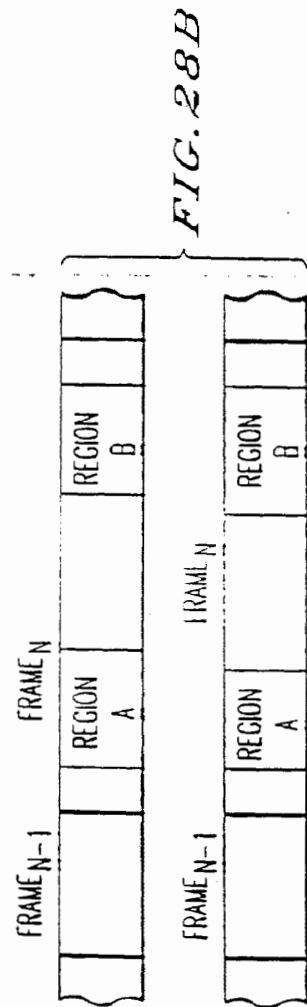
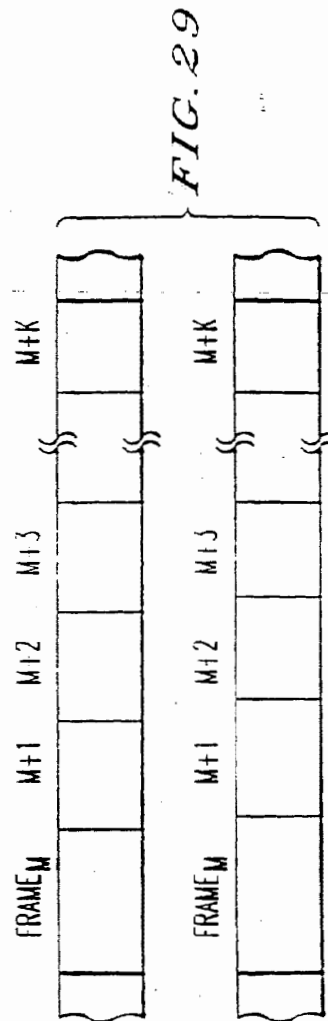
FIG. 27B

$q_{i,j}$	$q_{i+1,j}$		
$q_{i,j+1+\Delta 1}$	$q_{i+1,j+1+\Delta 1}$		$q_{i+n,j+1+\Delta 1}$
$q_{i,j+2+\Delta 2}$	$q_{i+1,j+2+\Delta 2}$		$q_{i+n,j+2+\Delta 2}$

FIG. 27C

$q_{i,j}$	$q_{i+1,j}$		$q_{i+n,j}$
$q_{i,j+1}$	$q_{i+1,j+1}$		$q_{i+n,j+1}$
$q_{i,j+2+\Delta 1}$	$q_{i+1,j+2+\Delta 1}$		$q_{i+n,j+2+\Delta 1}$
$q_{i,j+3+\Delta 1}$	$q_{i+1,j+3+\Delta 1}$		$q_{i+n,j+3+\Delta 1}$

FIG. 27D



DISPLAY OF FRAME N

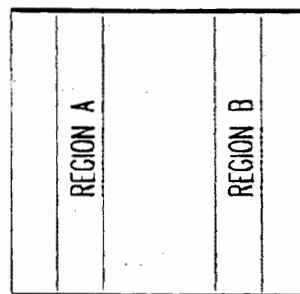


FIG. 28A

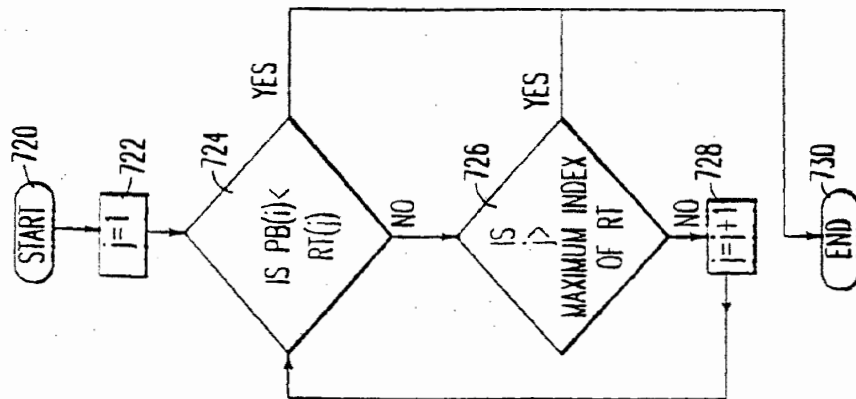


FIG. 30

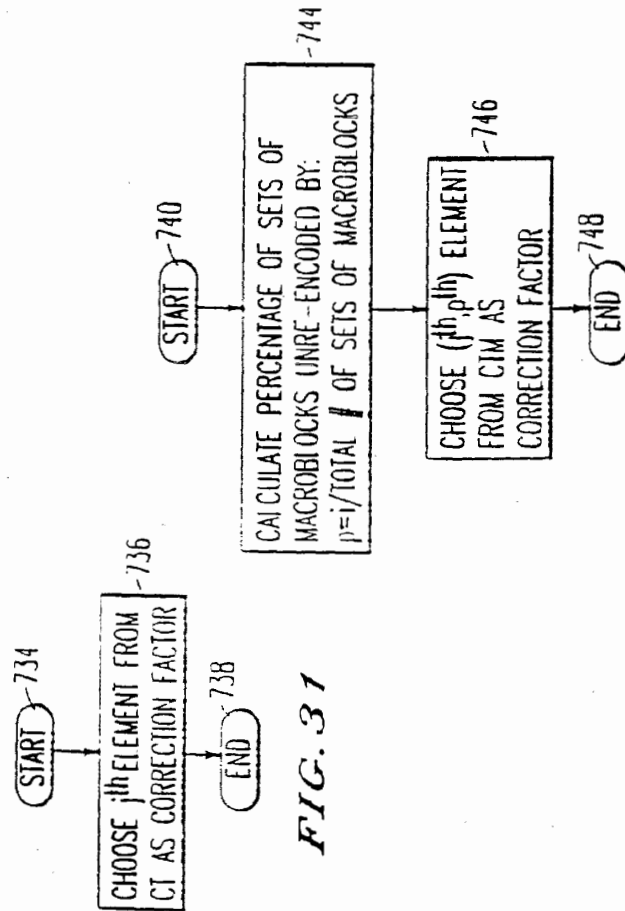


FIG. 31

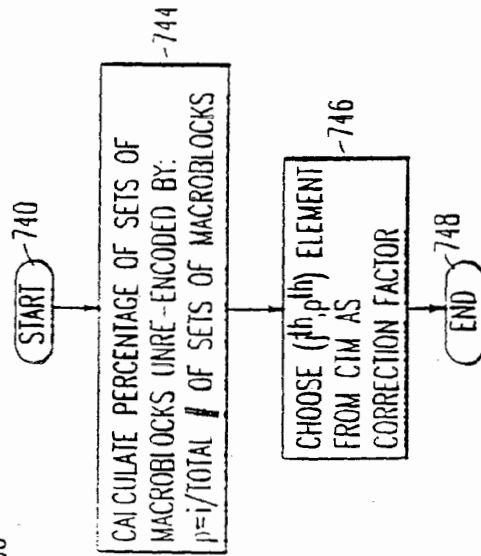
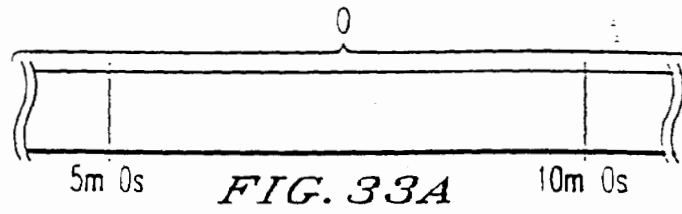
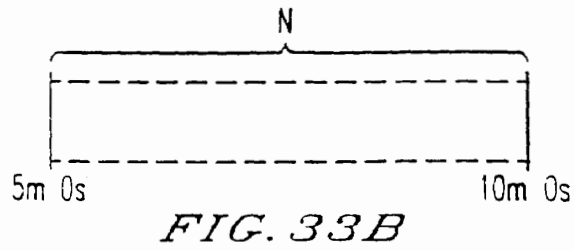
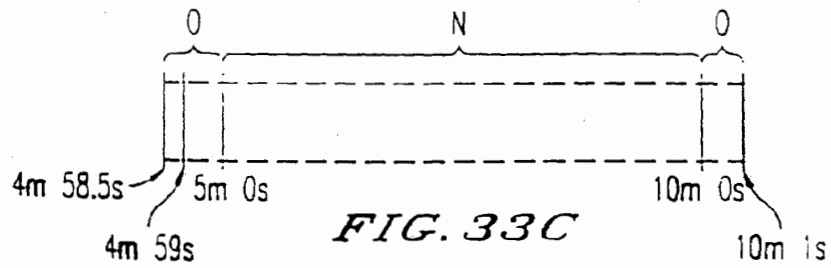
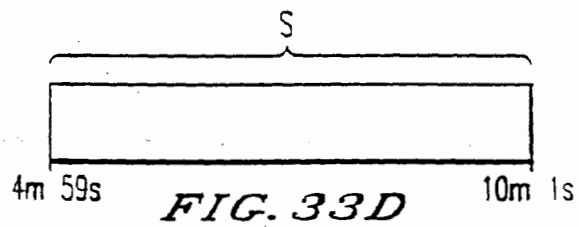
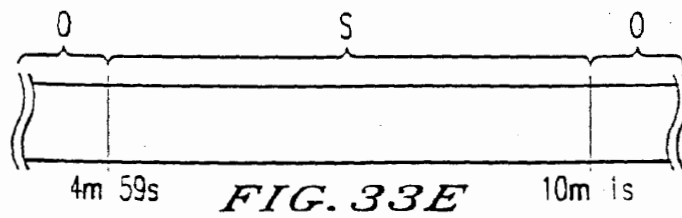


FIG. 32



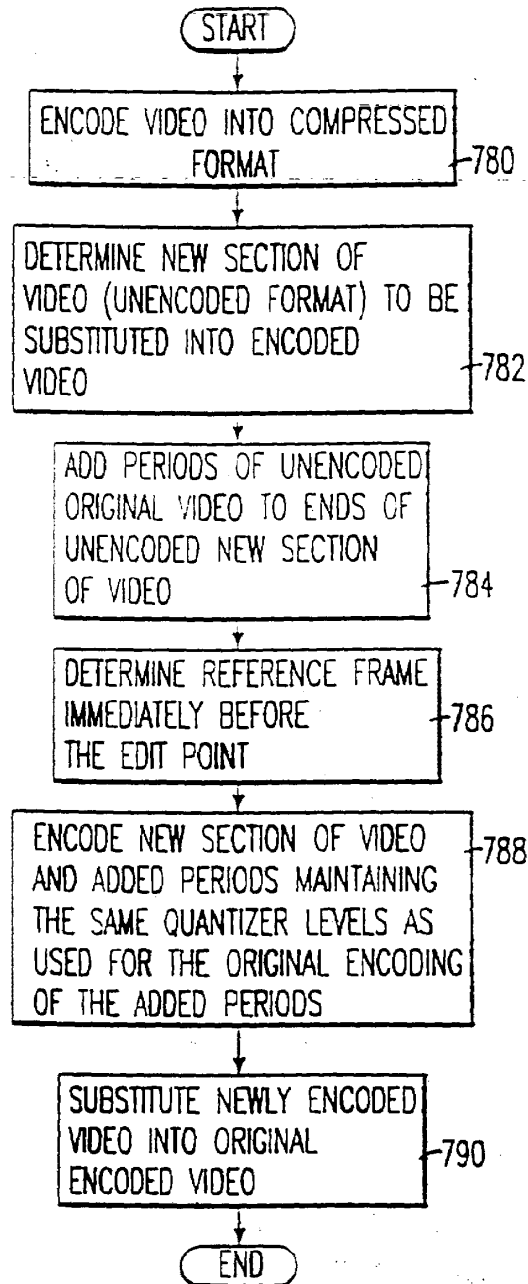
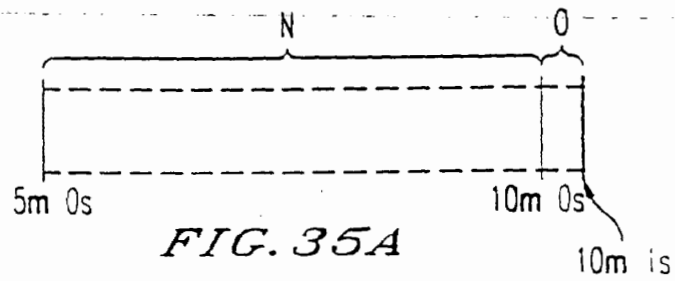
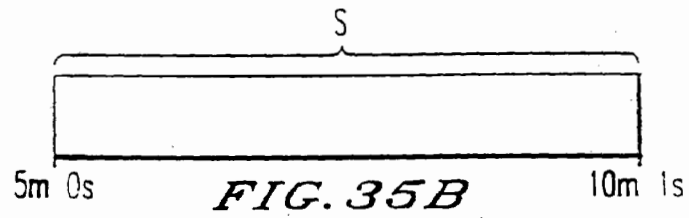
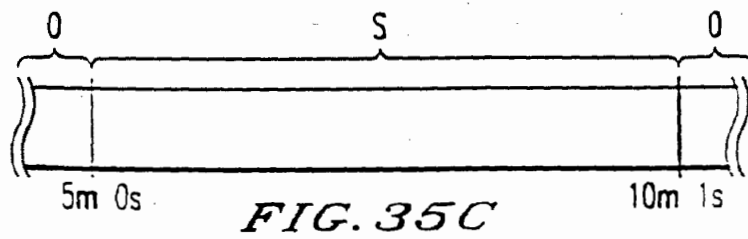


FIG. 34



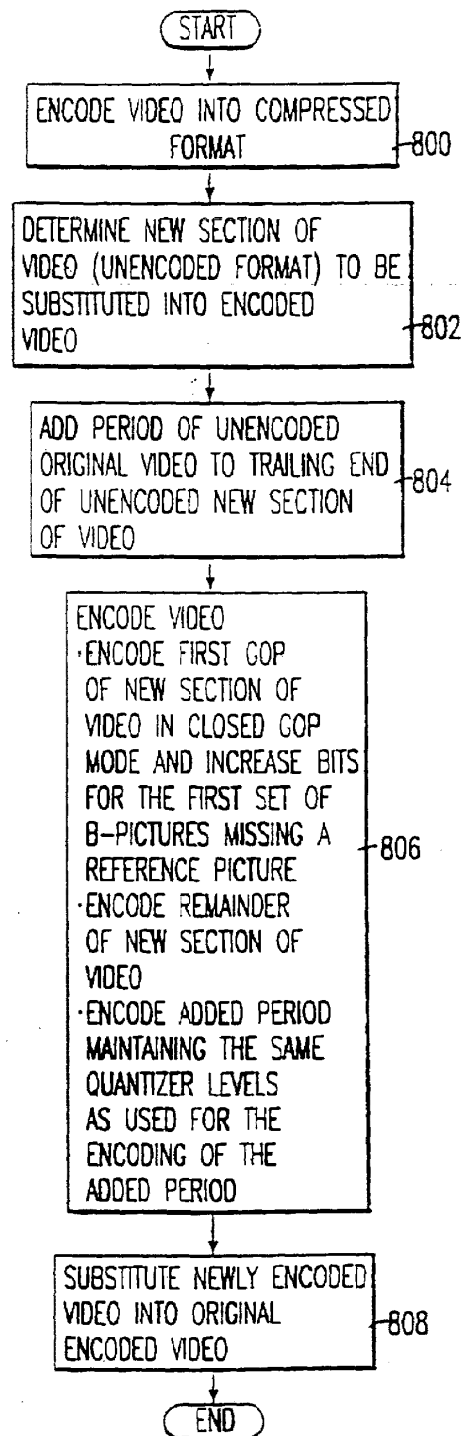


FIG. 36

q1	q2	q1	q2
q3	q4	q3	q4
q1	q2	q1	q2
q3	q4	q3	q4

FIG. 37C

q1	q2	q3	q4
q2	q3	q4	q1
q3	q4	q1	q2
q4	q1	q2	q3

FIG. 37B

q1	q2	q1	q2
q2	q1	q2	q1
q1	q2	q1	q2
q2	q1	q2	q1

FIG. 37A

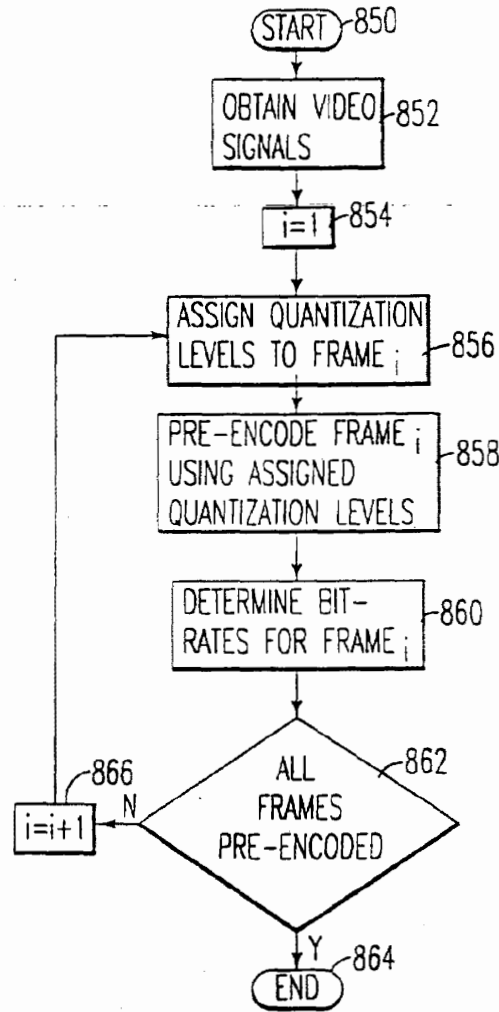


FIG. 38

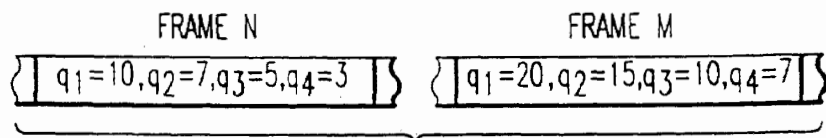


FIG. 39

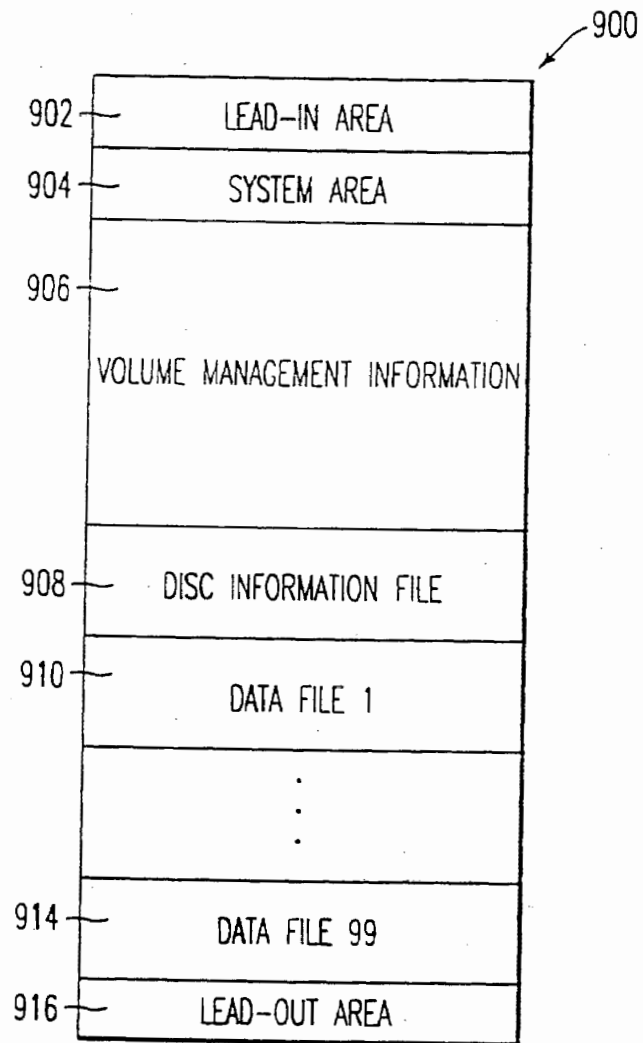


FIG. 40

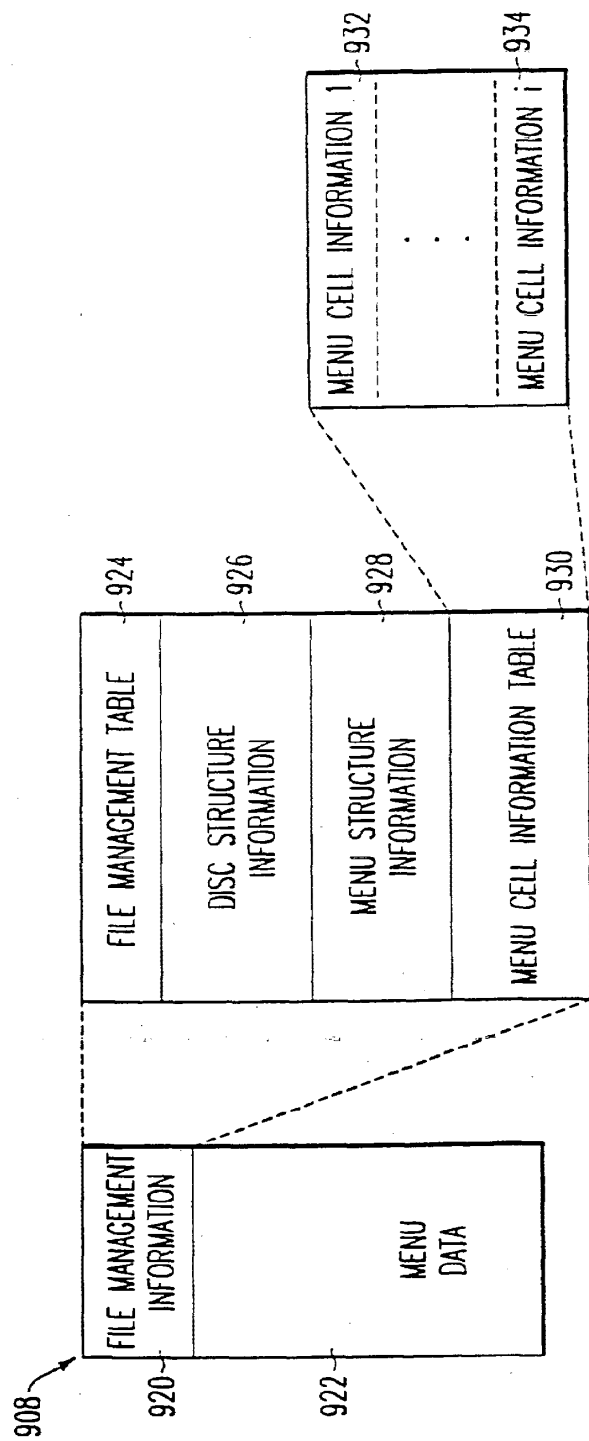


FIG. 41

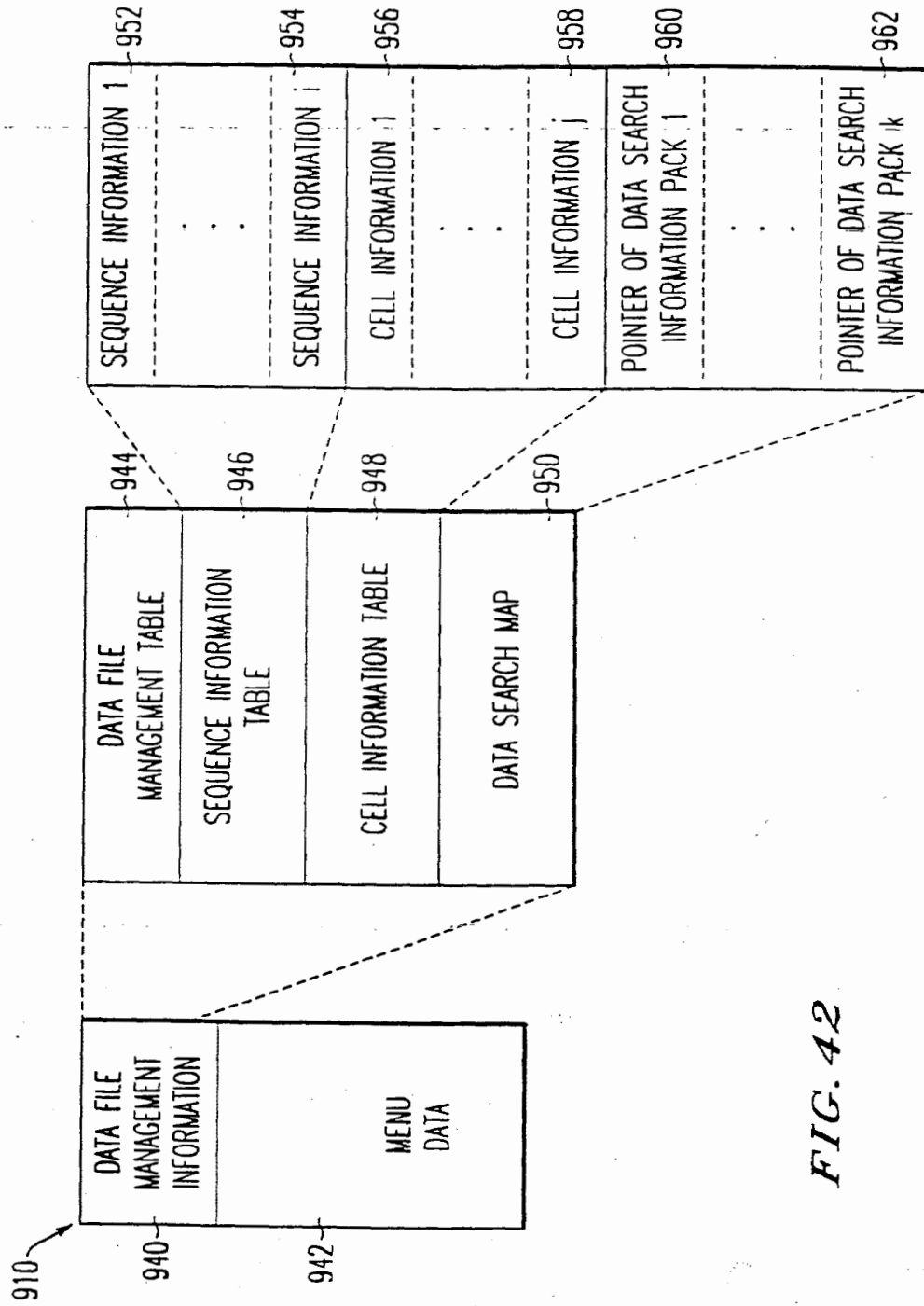


FIG. 42

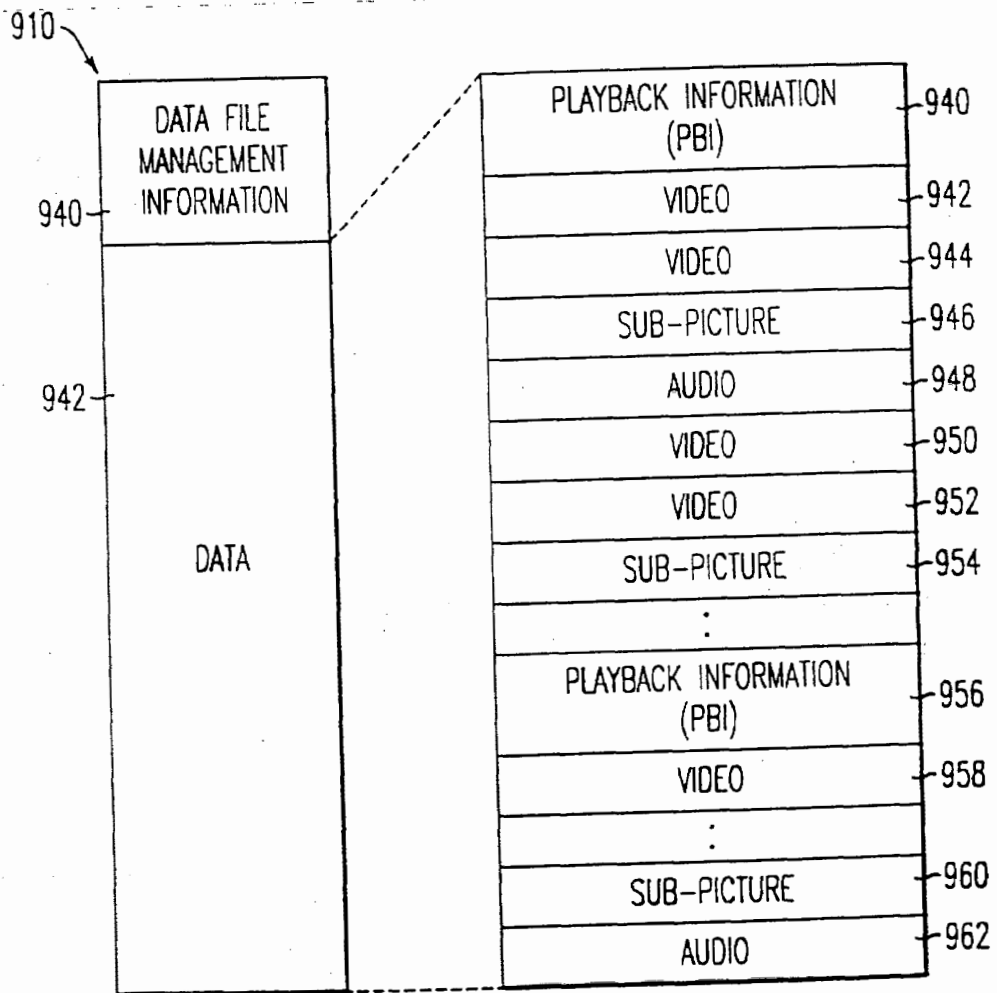
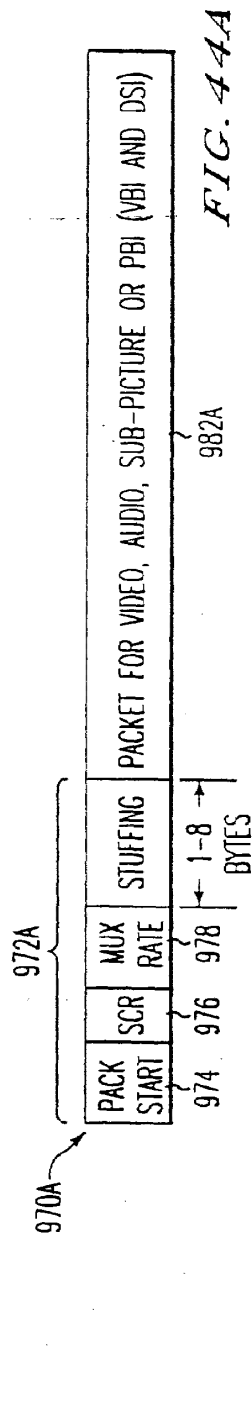
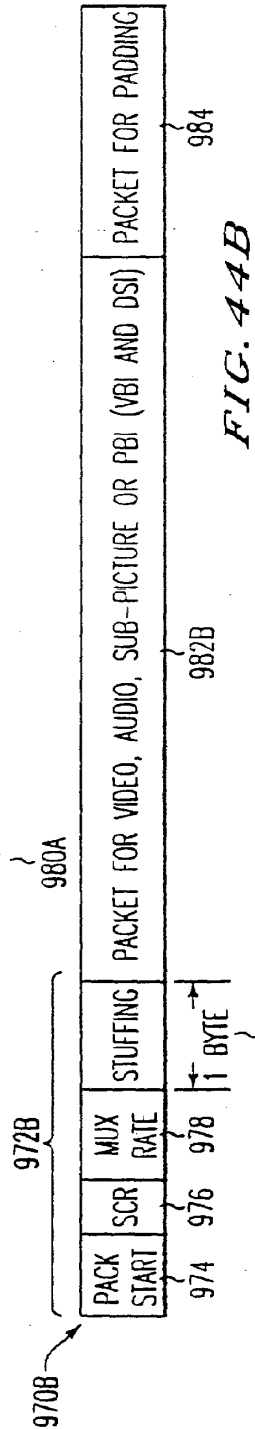
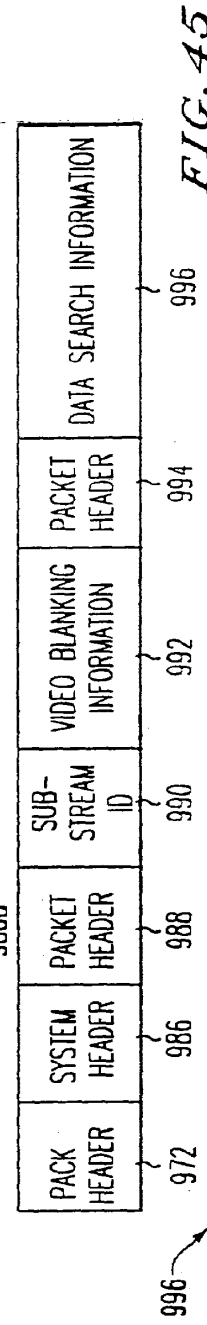
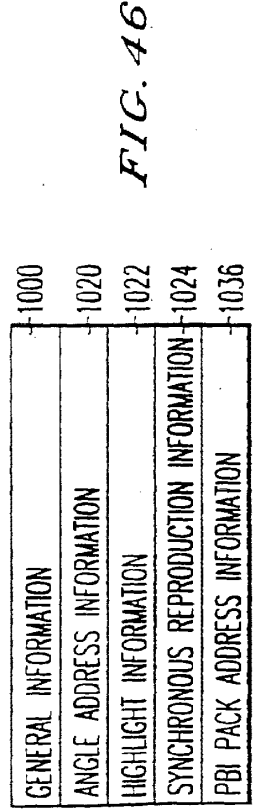
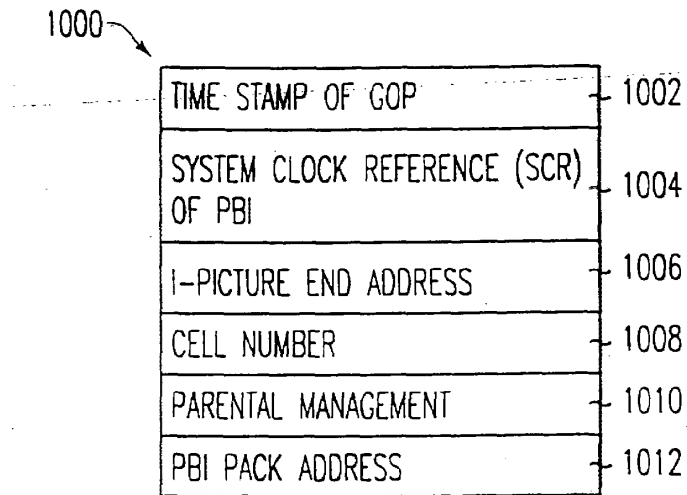
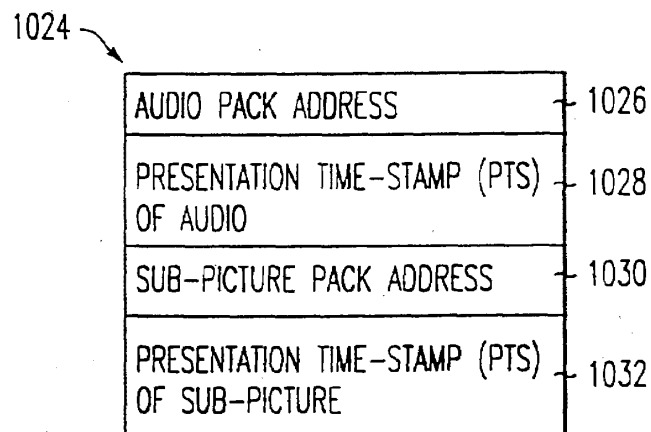
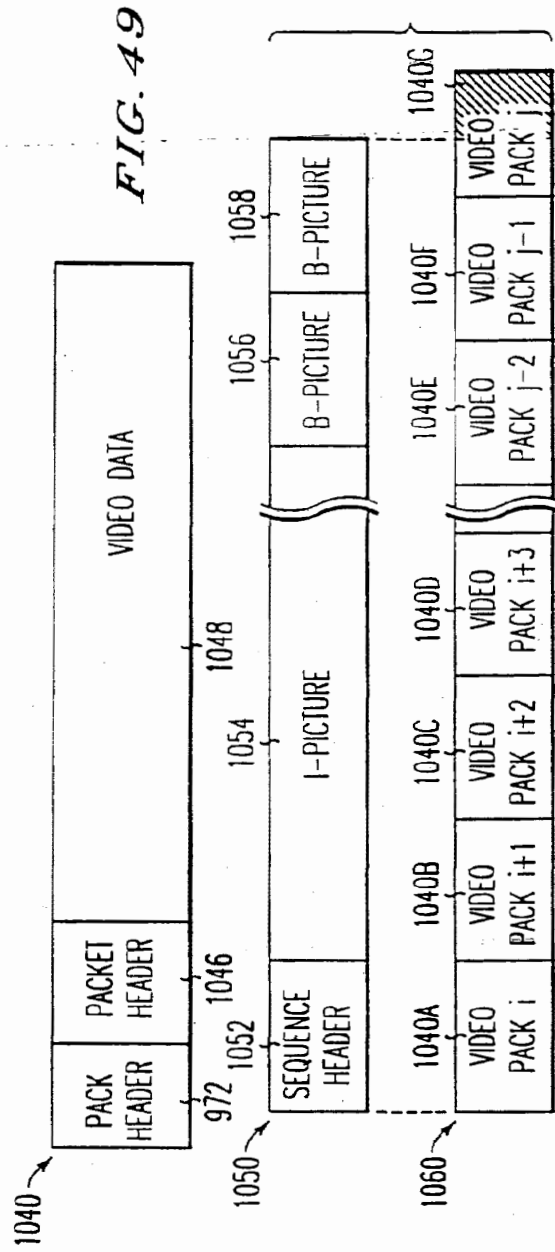
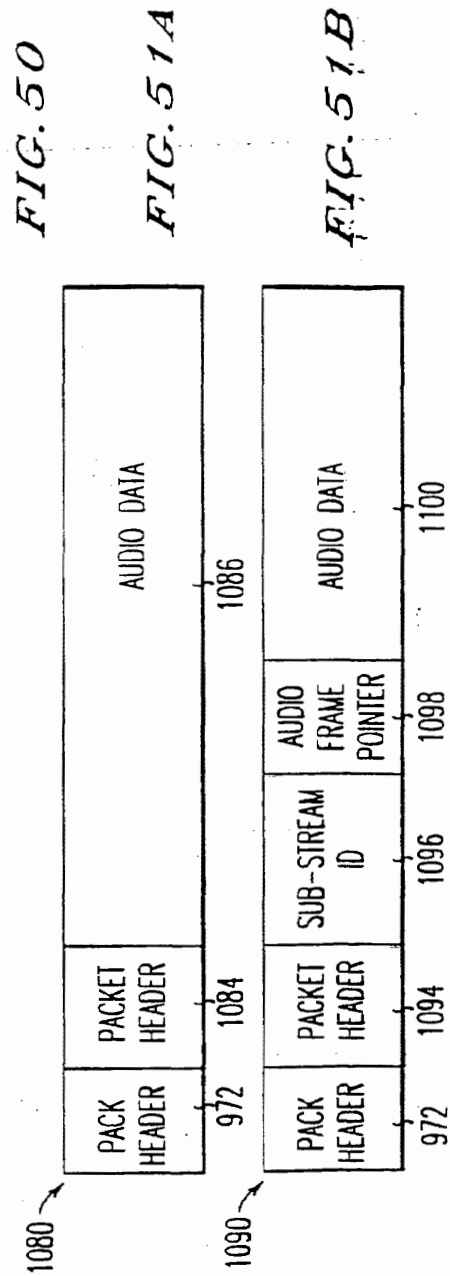


FIG. 43



*FIG. 47**FIG. 48*



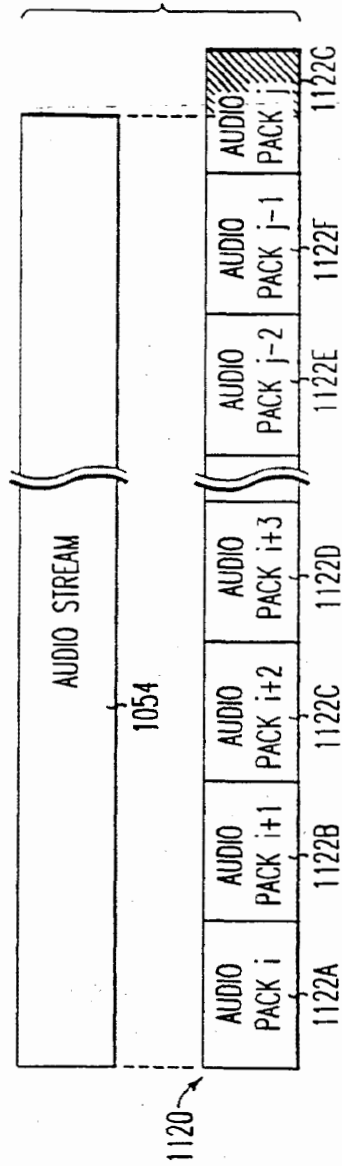
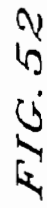
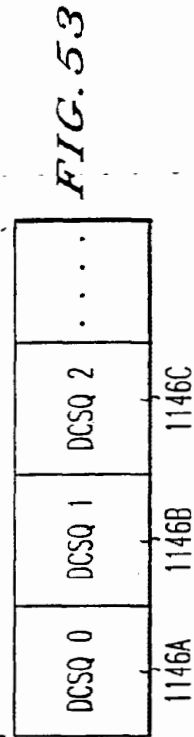
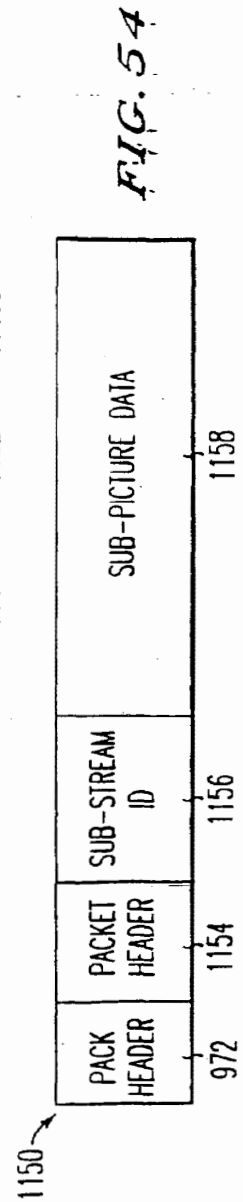


FIG. 54

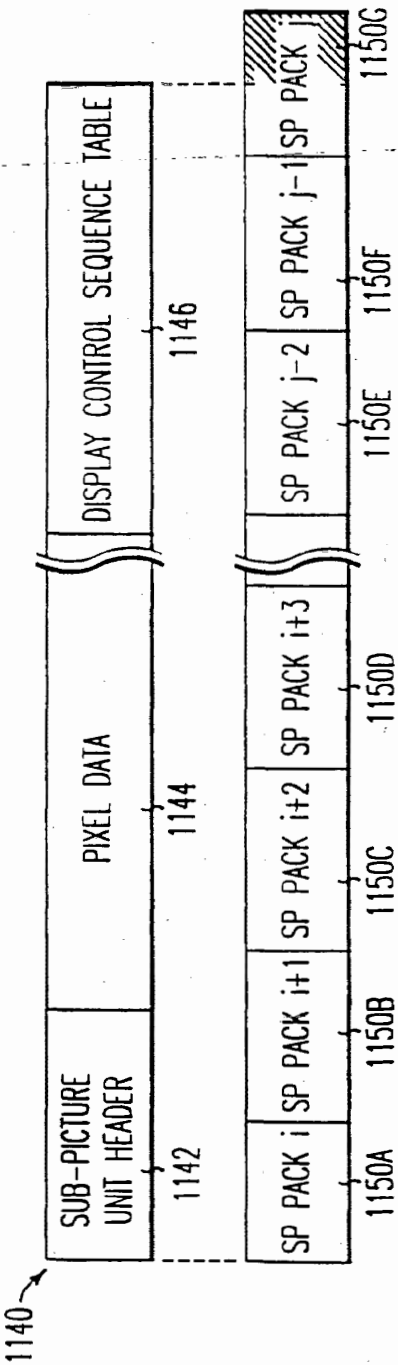


FIG. 55

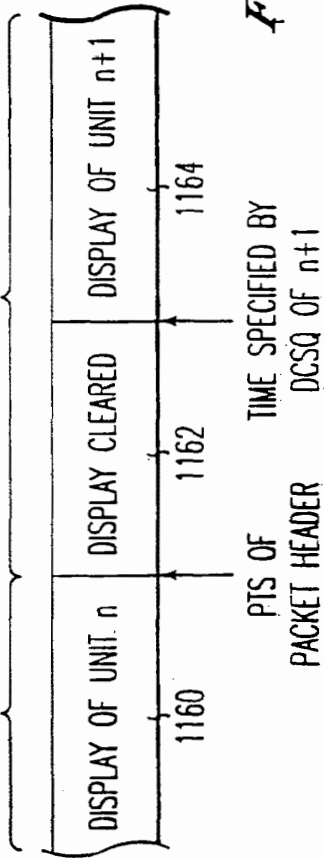


FIG. 56

REFERENCES CITED IN THE DESCRIPTION

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|---|---|
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|---|---|

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|---|--|



(11) **EP 1 312 188 B1**

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(54) **AUDIO DATA PROCESSING**
AUDIODATENVERARBEITUNG
TRAITEMENT DE DONNEES AUDIO

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(43) Date of publication of application:
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(73) Proprietor: **BRITISH TELECOMMUNICATIONS**
public limited company
London EC1A 7AJ (GB)

(72) Inventors:
• **RADENKOVIC, Milena**
East Midlands NG1 5DU (GB)
• **GREENHALGH, Christopher, Michael**
Nottingham NG11 6DR (GB)

(74) Representative: **Nash, Roger William et al**
BT Group Legal
Intellectual Property Department
PP C5A
BT Centre
81 Newgate Street
London EC1A 7AJ (GB)

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- **KYEONG-YEOL YU, JONG-HOON PARK, JONG-HYEONG LEE: "Linear PCM Signal Processing for Audio Processing Unit in Multipoint Video Conferencing System" IEE, July 1998 (1998-07), pages 549-553, XP002159070 KOREA**

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EP 1 312 188 B1

Description

[0001] This invention relates to audio data processing and in particular concerns a method, system and software application program for a real-time audio service for processing audio data streams transmitted over a communications network.

[0002] Developers of real-time audio services have for some time recognised the need for scalability in terms of the number of simultaneous listeners to an audio stream. Applications such as video on demand, on-line lectures and Internet radio require an audio stream to be broadcast from a single source to potentially many listeners. Various techniques have been proposed for reducing the bandwidth required for such broadcasts, most notably network multicasting including Internet multicasting, for example. By comparison however, support for many simultaneous speakers has not been fully developed. Some telephony and CSCW (Computer Supported Cooperative Work) applications such as audio and video-conferencing do support several simultaneous speakers, but the focus has been mostly on small groups. In addition, where the possibility of simultaneous speaking has been acknowledged applications usually prevent it through the use of floor control and "push-to-talk" concepts that enforce turn-taking and ensure that only one participant speaks at a time.

[0003] Communication protocols often discount the possibility of multiple simultaneous speakers. The Internet standard Real Time Protocol (RTP) (IETF RFC 1889) is the standard packet format used for continuous media traffic, such as audio streams, on the Internet. RTP includes sophisticated algorithms to control the amount of management traffic placed on the network, but assumes that audio traffic will not be a problem: "For example, in any audio conference the data [audio] traffic is inherently self-limiting because only one or two people will speak at a time ...", (RFC 1889, section 6.1). Similarly, the multicast Internet backbone (MBone), which provides the Internet's wide-area multicast capabilities, does not actively support simultaneous speakers. The Mbone guidelines for use and resources assume that each audio session will not have more than one active speaker at a time. Telecommunication applications are now emerging that support on-line events involving large groups of people. For example, collaborative virtual environments (CVEs) support interactive social events such as multi-player games and inhabited television for large on-line communities. Speech is the primary communication medium for on-line social interaction and so real-time audio services are vital for these new applications. Indeed, speech is arguably the most critical aspect of any real-time collaborative application. Research has shown that problems with other aspects of collaborative systems such as video, shared tools, or 3D graphics, can be resolved or compensated for by speech.

[0004] Analysis of patterns of audio activity in CVE applications, for example virtual teleconferencing, has re-

vealed significant periods when several participants are simultaneously generating audio traffic. In relatively focused applications such as tele-conferencing, audio activity is best approximated by a model of people transmitting audio at random, rather than deliberately avoiding overlapping speech. Indeed, overlapping audio, including speech and other sounds, is likely to be a basic requirement for many CSCW applications

[0005] Unfortunately, there are many disadvantages associated with real-time audio services that allow many simultaneous speakers. Real-time audio is bandwidth intensive, particularly for applications that support large numbers of users. In addition, packet-loss due to network congestion can severely reduce the intelligibility of received audio streams.

[0006] Providing an audio service for many simultaneous speakers is a significant problem since each speaker independently introduces a new audio stream that has to be accommodated by the network and that also has to be processed by each recipient's audio receiver. While multicasting reduces the bandwidth required to distribute a single audio stream to many listeners, it does not address the problem of many simultaneous speakers.

[0007] Audio services are known in which peer-to-peer architectures provide each user with an independent audio stream from each respective speaker. By receiving independent audio streams each listener can create their own personal audio mix. The mix can be tailored to the user's own audio equipment capabilities; the audio streams can be spatialised according to associated speaker locations within a shared virtual space; and the streams can be individually controlled, for example allowing the volume of particular speakers to be raised or lowered according to their relative importance, for example. Peer-to-Peer audio services can be implemented using unicast or multicast protocols.

[0008] The peer-to-peer approach is very demanding in terms of network resources, particularly bandwidth, and can easily flood the network with traffic. With underlying unicast protocols, which are typically used today for wide-area communication, the resulting number of audio streams is of the order n^2 , where n is the number of simultaneous users, that is to say for the case in which all users are sending audio data simultaneously. With underlying multicast protocols, which are still experimental over wide-area networks, this reduces to the order n , but no lower.

[0009] With conventional networks such as the Internet, once the traffic in any part of the network exceeds the local capacity then that part of the network becomes congested. This congestion causes speech to become delayed (due both to direct delay and also jitter) and increasingly broken and disjointed (due to audio data packets being lost in transit). The peer-to-peer approach is also very demanding in terms of the processing to be done by each listener's terminal, which must be capable of receiving, decoding, processing, mixing and playing out all of the available audio streams. User terminals may

become overloaded, causing problems similar to an overloaded network.

[0010] Audio services which use total mixing are also known. In the total mixing approach each audio stream is sent to a central audio server that mixes the audio streams into a single stream that is then redistributed to each listener. This approach requires that each listener and surrounding network handles only one audio stream. Total mixing, however, prevents each listener from creating their own personal mix and the central server becomes a potential bottleneck in the system, since it and its nearby network still have to receive, mix and distribute n audio streams. Other drawbacks associated with total mixing include increased delay due to additional processing, reduced audio fidelity, additional hardware requirements and management complexity. In this respect total mixing is only appropriate for relatively simple applications where resources are limited. Total mixing is not appropriate for high fidelity applications such as home entertainment and tele-conferencing where more resources are generally available and where audio spatialisation is more important, that is to say where separate audio streams are required. An example of total mixing is described by one of the present inventors in "Inhabited TV: Multimedia Broadcasting from Large Scale Collaborative Virtual World", Fracta Universitatis, Ser. Electronics and Energetics, 13(1) IEEE, ISBN 0-7803-5678-X.

[0011] Another example of an audio service that supports many simultaneous speakers is described in "Diamond Park and Spine: A Social Virtual Reality System With 3D Animation, Spoken Interaction, and Runtime Extensibility", PRESENCE, 6(4), 461-481, 1997 MIT. This paper describes a system which allow users with low bandwidth connections to access an audio-graphical multi-user virtual world. In this approach one or more low-bandwidth users connect to a specialised access server that interfaces to a main system which is configured for peer-to-peer multicast audio. The access server or servers deal with and mix all of the available audio streams. It is also known from EP-A-1 024 638 to direct terminals to adjust their coding algorithms based on system bandwidth usage. According to an aspect of the present invention there is provided a method of processing audio data streams transmitted in a communications network; said method comprising the steps of:-

- i) receiving a plurality of audio data streams transmitted from one or more audio data stream transmitters distributed in the network; and,
- ii) processing data relating to at least one respective network resource parameter to determine respective network resources available for subsequent communication of said audio data streams to at least one respective audio data stream receiver in the network; and,
- iii) comparing said available resources with respective network resource requirements necessary for communicating said audio streams to at least one

respective audio data stream receiver in the network; and,

- iv) determining whether to mix selected audio data streams prior to transmission in response to said comparison.

[0012] The term "network resources" used herein refers to any component in a communications network necessary for communicating audio data streams to potential recipients, including but not limited to network communication links, network audio processors or mixers, audio stream transmitters and receivers and user terminals, for example.

[0013] By comparing available network resources with network resource requirements necessary for communicating audio streams the above method enables audio mixing decisions to be made dynamically in response to changing network conditions or application requirements. In this way the number of data streams transmitted can be controlled so that network traffic can be optimised according to available network resources. This aspect of the invention is particularly relevant for dynamic virtual environment applications involving varying numbers of active participants engaged in various activities and running over dynamic networks where congestion and delay may change rapidly. The method enables mixing decisions to be made that adapt to changing network conditions and application requirements so as to optimise the conflicting requirements of audio quality and traffic management.

[0014] Preferably the method further comprises the step of:-

- v) processing two or more audio streams in response to said comparison to provide at least one mixed audio data stream for subsequent transmission in said network. Thus, if a decision is made to mix selected audio data streams the traffic introduced into the network can be reduced. The amount of processing required by neighbouring data stream receivers can also be reduced. Mixing enables two or more selected audio data streams to be combined to reduce overall network congestion without significantly affecting the intelligibility of the mixed audio streams when received by a user, for example in a similar way that stereo audio signals can be combined for playback on a non-stereo enabled output device such as a hand held radio receiver having a single loudspeaker.

[0015] Conveniently, steps ii) and iii) comprise the steps of:-

- determining a current value for the or each respective network resource parameter; and,
- comparing the or each respective current resource parameter value with a respective minimum resource threshold value necessary for communicat-

ing said unmixed audio data streams to the or each respective receiver. In this way the available network resources can be compared with pre-determined minimum resource requirements necessary for transmitting or processing all the unmixed data streams.

[0016] In preferred embodiments, said minimum resource threshold value is determined according to at least one pre-defined quality of service parameter. In this way mixing decisions can be made according to pre-determined quality of service requirements.

[0017] Preferably, said network is a packet switched network and said pre-defined quality of service parameter is defined by a maximum packet loss rate. Thus, a minimum threshold value may be determined according to an acceptable packet loss rate associated with a codec used to encode respective audio streams. For instance, a maximum acceptable packet loss rate for an audio codec may be 15%.

[0018] Conveniently, one network resource parameter relates to available network bandwidth for transmission of said audio data streams to the or each respective receiver. In this way mixing decisions can be determined according to the bandwidth available for transmitting the audio streams to a next audio stream receiver in the network. This provides for efficient use of network bandwidth and readily enables a maximum number of mixed or unmixed audio data streams to be transmitted without causing congestion in the network.

[0019] Preferably, said available bandwidth capacity is determined by user specific quality of service requirements. In this way bandwidth resources can be allocated or reserved for use according to user specified quality of service requirements. The allocation or reservation of bandwidth may be controlled by different charging tariffs associated with the quality of service required. In this way a user may specify a quality of service requirement of say 3 x 64kb/s audio channels in which case selected audio streams will be mixed by the network when more than three separate audio streams are to be transmitted by the network. In this way available bandwidth may be considered as allocated or reserved bandwidth.

[0020] Conveniently, one network resource parameter relates to receiver processing characteristics. Thus, mixing decisions can be determined according to the characteristics of respective receivers. In this way separate audio streams being sent to a receiver having a low processing capability or capacity can be mixed in the network so that the number of audio streams the receiver receives is reduced.

[0021] In preferred embodiments, said audio data streams are selected for mixing according to predetermined criteria. This enables selection criteria to be used to determine which of the received audio data streams should be mixed.

[0022] In one way, said audio streams are mixed according to audio stream content. For instance, in a virtual

environment one or more audio streams may be more significant in terms of audio content than others and the loss of audio spatialisation experienced by a recipient will be less if the less significant audio streams are mixed in preference to the more significant ones.

[0023] In another way, said audio streams are mixed according to recipient requirements. In this way mixing can be determined by the recipient's own requirements, for instance the extent of audio spatialisation required.

[0024] In a further way, said audio streams are mixed according to respective audio stream sources. In this way audio streams from related sources can be mixed, for instance it may be desirable to mix all audio streams associated with a particular group of participants in an audio conference or virtual environment.

[0025] In a yet further way, said audio streams are mixed according to receiver capabilities. In this way audio streams may be mixed according to the capabilities of the receiver. For example, a receiver may comprise a full 3-D audio system capable of recreating fully spatialised studio quality audio where mixing considerations are important for recreating spatialised audio. Alternatively a receiver may comprise a simple stereo audio system where mixing considerations are less important.

[0026] According to another aspect of the invention there is provided a software program arranged to process audio data streams according to the above mentioned method.

[0027] According to a further aspect of the invention there is provided a system for processing audio data streams transmitted in a communications network; said system comprising:-

- i) a receiver for receiving a plurality of audio data streams transmitted from one or more audio data stream transmitters distributed in the network; and,
- ii) a processor for processing data relating to at least one respective network resource parameter to determine respective network resources available for subsequent transmission of said audio data streams in the network; and,
- iii) a comparator for comparing said available resources with respective network resource requirements necessary for transmission of said audio streams to at least one respective audio data stream receiver in the network; and,
- iv) determining means for determining whether to mix selected audio data streams prior to transmission in response to said comparison.

[0028] The invention will now be described by way of example only with reference to the accompanying drawings in which:-

Figure 1 is a schematic representation of a network used for implementing an embodiment of the present invention;

Figure 2 is a schematic representation of a logical

network topology for part of the network of Figure 1; Figure 3 shows a modular block diagram of an audio data processor for processing audio data streams transmitted in the network of Figures 1 and 2; Figure 4 is a flowchart showing steps involved in implementing an embodiment of the invention; Figure 5 is a schematic representation of a distributed processing system for simulating network conditions; Figure 6 is a schematic representation of one implementation of the distributed processing system of Figure 5 used for determining network performance characteristics; Figure 7 is a graphical representation of simulated network performance characteristics showing packet loss rates for different audio distribution strategies; and, Figure 8 is a graphical representation similar to Figure 7 showing the different audio stream distribution characteristics for different audio distribution strategies.

[0029] An example of an IP communications network for implementing one arrangement of the invention is shown in Figure 1. In Figure 1 a plurality of user terminals 100 are connected to the Internet 102 via respective Internet access servers 106 which are each connected to an audio processor 104. The audio processors 104 are each capable of mixing a plurality of audio data streams received over the network. Although only two audio processors 104 are shown in the network of Figure 1 in practice any number of audio processors may be provided and distributed throughout the network for receiving, mixing or transmitting respective audio data streams. The audio data streams are typically transmitted using the Internet standard data stream transfer protocol Real Time Protocol (RTP) and underlying multicast transport protocols although unicast could be used also.

[0030] Referring to Figure 2, logically the network of Figure 1 can be considered to comprise a plurality of audio data stream sources 200, audio data stream sinks 202 and audio mixer 204 type components. Thus, each terminal 100 and audio processor 104 may comprise at least one source, sink or mixer component and the network of Figure 1 may comprise a plurality of these components linked together by audio data streams 206. In Figure 2 a simple logical network topology is shown in which the source and sink components are positioned at the terminal nodes of the network and the mixer components are positioned at the non-terminal nodes. In the network of Figure 2 each source 200 transmits one or more audio data streams to a respective parent mixer node 204. The mixer nodes may then forward the received audio data stream or streams directly to the other connected nodes, mix selected ones of the received audio data streams and then forward the respective mixed data streams, or perform a combination of these two actions. Each sink eventually receives each of the data

streams either in the original un-mixed state or part of a new mixed stream. As will be described in greater detail later, each mixer component may additionally create multiple mixes from arbitrary subsets of the received data streams and transmit these mixes instead of or in addition to the audio data streams it would otherwise forward to other nodes in the network. For example, each mixer component may select to mix only some of its incoming audio data streams and forward the resultant partial mix on to other connected nodes.

[0031] In the arrangement of Figure 3 the audio data stream processors are integrated with selected IP multicast enabled routers 300 distributed in the network of Figure 1. Each audio processor 104 comprises an audio data stream receiver 302 for receiving audio data streams transmitted over the IP network, an audio mixer component 304 for mixing selected audio data streams and an audio data stream transmitter 306 for forwarding mixed and unmixed audio data streams to other audio processors 104 or terminals 100 in the IP network. The audio data stream receiver 302 and transmitter 306 comprises software components of the type implemented in audio applications such as Real Inc's Real Player or Microsoft Inc's Media Player, for example IP sockets, packet buffers, packet assemblers etc. The audio mixer component 306 is provided for mixing two or more audio data streams received at the processor 104. The audio mixer component is arranged to average selected audio data streams to create a single combined data stream comprising the same number of bits as each unmixed data stream.

[0032] Each audio processor is further provided with a congestion monitor 308 for monitoring congestion on the respective communication transmission links connecting the audio processor to other parts of the network. In one arrangement the congestion monitor 308 utilises Real Time Control Protocol (RTCP) control messages or the like received from respective audio stream receivers in the network to determine congestion levels on respective transmission links. These control messages provide the audio processors with information relating to data stream congestion on respective neighbouring transmission links and audio processors. Selected audio processors, typically those on the edge of the network connecting terminals 100 to the Internet, are also provided with a database 310 containing data relating to recipient terminal equipment characteristics including audio data processing and playback capabilities, network connection types and speeds, other recipient specific data including current tariff data for determining an appropriate quality of service to be provided and user specific audio data stream mixing profiles comprising user defined mixing preferences and other data relating to user specific mixing policies.

[0033] Mixing may occur at many of the audio processor mixer nodes in the network of Figure 1. When an audio processor mixes selected audio streams the number of audio data streams to be transmitted to neigh-

boursing audio processor or terminal nodes is reduced and this also reduces the amount of processing required at the neighbouring nodes. Distributing the audio processor mixer nodes throughout the network as described enables the overall audio mixing task to be shared and mixing bottlenecks avoided. Distributed processing not only provides for scalability, say in terms of the maximum number of simultaneous speakers allowed in an audio conference, but also enables audio processors to monitor and respond to changing local network conditions in heterogeneous environments such as the Internet. Since mixing reduces the quality of the audio streams being mixed the audio processors are arranged to limit the amount of mixing they perform within the constraints of the available network resources so that end user recipients receive as many separate audio streams as possible within prevailing network resource constraints. Thus, the audio data processors are arranged to adapt to changing network conditions so that at one extreme peer-to-peer communication is possible, that is when available network resources are adequate for communicating all the received audio data streams to respective neighbouring nodes. At the other extreme, where available network resources are severely restricted, say due to network congestion, all audio data streams are mixed together to provide a single data stream for communication to the neighbouring nodes.

[0034] Mixing decisions are made dynamically by the audio data processors according to software program logic stored in audio processor memory and executed by the audio processors.

[0035] Referring now to the flowchart of Figure 4, network conditions such as available bandwidth, packet loss rate and delay are monitored by the congestion monitor 308 in step 400 to determine current values for respective network resource parameters associated with transmission links and neighbouring processor and terminal nodes located in the network. Current values for variable network resource parameters including but not limited to available bandwidth on respective transmission links and processing capacity and delay of respective processor nodes are stored in the data base 310 in step 402. Current values for other more stable network resource parameters including but not limited to terminal node processing capability and capacity and user specific mixing preferences and quality of service requirements are also stored in the processor databases 310. The more stable resource parameters may be monitored in step 400 in the same way as the variable resource parameters or by periodic polling the resources for current parameter values.

[0036] In step 404 the receiver 302 monitors relevant unicast or multicast communication channels for incoming audio data streams and determines in step 406 whether any streams are being received at the audio processor node. If audio data streams are being received the audio processor proceeds to step 408 where an appropriate algorithm determines the network resources necessary for transmitting the received data streams to

the next relevant node or nodes in the network. If audio data streams are not being received monitoring continues and control passes back to step 400. In step 408 network resources necessary for forwarding all the received data streams to each respective node in the relevant network distribution tree are determined and in step 410 the respective available resources for transmitting the data streams are determined. In step 412 the current values of the relevant network resource parameters are compared with respective resource requirements necessary for communicating the received audio streams to the respective next network nodes comprising audio data stream receivers. If the available resources match those required for subsequent communication of the data streams all the data streams are transmitted onwards in step 420 to the relevant next nodes. However, if the audio processors determine in step 412 that there are insufficient resources selected mixing occurs. For instance, if there are only three communications channels available on a particular transmission link and four separate data streams are received, at least one pair of data streams are mixed prior to subsequent transmission by the respective audio processor. Similarly, if a recipient terminal node is only capable of processing two data streams simultaneously only two sets of mixed streams are provided for transmission to that terminal. Further, if a terminal node is designated as having a pre-determined quality of service as defined by a user selected tariff there may be insufficient bandwidth allocated for forwarding all the streams without mixing so that mixing will occur even if the network has sufficient bandwidth resources available on the relevant link or links to the user terminal node. In step 414 the difference between available and required resources is determined so that the number of streams to be mixed can be determined. The audio data stream processors select appropriate data streams to be mixed in step 416. The selected streams are mixed in step 418 and then transmitted by the appropriate audio processor transmitter 306 in step 420. Audio data stream selection in step 416 may be based on any number of considerations relevant to network, application or user requirements, for example.

[0037] The following discussion concerns mixing considerations that may be relevant in CSCW environments which support many simultaneous speakers.

[0038] One consideration for mixing may concern the roles of speakers. Some real-time CSCW applications assign participants to different roles within an event. For example, early experiments in inhabited television differentiated between performers, inhabitants and viewers. Performers are part of the core content of an on-line TV show whereas inhabitants are active within the virtual world, but typically receive a broadcast mix created by a director. These roles are complemented by differences in the technologies used to access the real-time event. Performers typically use professional studio-quality equipment, with fully spatialised 3D audio. Inhabitants may use commodity PCs, equipped with headphones.

Viewers on the other hand may use conventional television sets, equipped with multiple loudspeaker surround sound audio systems. Roles or so called "layers of participation" can determine mixing policy. For instance, it may be appropriate to ensure that performers are heard with the maximum possible audio quality. Thus, as network congestion increases the audio streams for inhabitants might be mixed together first, with the performers streams being kept separate for as long as possible. Other CSCW applications may also benefit from defining layers of participation and using these to prioritise audio sources for mixing.

[0039] Another consideration for mixing may concern the roles of listeners or recipients. Mixing can also be prioritised according to listener requirements. Roles or layers of participation can also define the different ways in which listeners take part in an event, although many participants will be both speakers and listeners in an event. For example, an active inhabitant may benefit from fully spatialised audio that provides clues to support navigation and conversation management. A passive viewer with a surround-sound audio system may benefit from a mix that clearly separates the key performers, but where their accurate location in the world is less important. In the case of inhabitants it may be important to maintain the separation of streams from nearby participants, whereas for viewers it may be appropriate to maintain the separation of key performers only.

[0040] A further mixing consideration may concern the grouping of audio sources. CSCW applications often group participants in some pre-determined way. It is often appropriate to mix audio streams from one coherent group. For example, avatars in a CVE may have gathered together to form definable and separate groups. Audio streams from each group could be mixed to form a single stream that could be spatialised to the average position of the group as a whole in the CVE. In addition, some CSCW applications calculate levels of mutual awareness among participants which may provide a more dynamic basis for mixing respective audio streams.

[0041] Another mixing consideration may concern voice characteristics. The timbre of voices or other audio sources may be useful for determining which streams to mix. For example, it may be appropriate to mix a high and a low pitch voice into a single stream so that a listener can readily separate them when hearing the mixed stream.

[0042] Patterns of activity within a multiple speaker environment may also determine mixing decisions. For example, audio data streams from participants whose speech rarely overlaps could be mixed together

[0043] A number of more practical concerns may also shape affect mixing decisions. Mixing may depend upon aspects of the available communication network, including its topology (i.e. shape and structure), underlying bandwidth, regional variations, or transitory congestion. Mixing decisions may also depend on the available computing resources, for example the number and capability

of available mixer components, how many are positioned within the network, and how heavily loaded they are. Mixing decisions may also consider the current and past states of the system. For example, the transition from one choice of mixed streams to another may be noticeable to users, and potentially undesirable.

[0044] It is clear from the above discussion that the process of selecting data streams for optimal mixing is a complex task and will very often be application specific. For instance, different applications may have different mixing requirements. In particular applications may have different ways of assigning priorities to audio streams to determine the order in which streams are gradually mixed together as network resources become scarce. Low priority streams will be mixed before higher priority streams. These requirements may also vary between different phases of the same application as a session progresses. In particular, priorities may change as participants take on different roles or move to different locations. For example, a virtual football game with the crowd (audience) will not have the same mixing requirements as a virtual shopping mall, a virtual education application (on-line demonstration, lecture etc), or an on-line television drama in a virtual world.

[0045] In a virtual football game for example, audio priority will be higher for the referee say, than the other players. In this respect, the system will avoid mixing the referee's audio stream with the respective players audio streams unless this becomes absolutely necessary due to network resource limitations. On the other hand audio streams from a crowd of spectators will have a lower priority since each spectator will not need to be heard individually. It may be sufficient to mix all the streams from the same "stand" or group of collocated spectators and spatialise only the resulting stream at the receiver. In this way mixing is based on the roles of the speakers, that is to say, audio streams from more important speakers are forwarded and less important speakers are mixed

[0046] In a virtual shopping mall, mixing policy may be based on the collocation and mutual awareness of the participants. For example, participants may want to receive separate audio streams from other collocated participants or other more important participants of whom they are more aware of than others. The remainder of the of the group can be mixed together or divided into smaller groups which are mixed separately, similar to the different stands in the football game example.

[0047] In a virtual lecture environment there is likely to be one lecturer, may be a few demonstrators and many mutually aware students. In this environment mixing can be based on speaker roles. For instance, the lecturer's and demonstrators' audio streams may be forwarded so that the other participants can process the individual streams on receipt, where the respective students audio streams will be mixed. The present inventors have implemented and tested the invention in a distributed processing system simulator 500 shown in Figure 5. In the distributed processing system two end user systems

502a and 502b are shown on a first local area network 506 and two end systems 502c and 502d on a second local area network 512. The system 500 may comprise any number of end user systems 502 depending on the networks being simulated. Each end user system comprises a respective virtual world client 514 for accessing a shared virtual world generated by a virtual world server 516 on LAN 512. Each end user system is also provided with a local audio server 518 that is interfaced to respective audio hardware (not shown) so that users can speak to each other within the virtual world environment. Each client 514 controls the local audio server 518 for the respective end system and uses information in the virtual world to determine how the audio server should transmit, receive and process audio streams, for example according to the positions of other users in the virtual world.

[0048] For peer-to-peer audio, each user's audio server 518 sends an audio data stream directly to all the other audio servers in the system using underlying unicast or multicast protocols. An audio processor 104 is provided on each LAN for mixing selected data streams received from the connected audio servers. The audio processors are both controlled directly by the virtual world server 516 and are connected together by means of a WAN simulator 520. The local audio servers 518 and audio processors 104 together define an audio distribution tree as shown by the dashed lines 522 in Figure 5. Each audio processor is capable of receiving audio data streams from the audio servers on its respective LAN and transmitting these streams to the remote audio processor on the other LAN. The audio processors are arranged to adapt to changing network conditions so that instead of three separate audio streams being forwarded from say end user systems 502a, 502b and 502c to 502d the audio processors 104 can mix respective data streams so that end user systems 502c and 502d receive a single stream comprising a mixture of the streams from end user systems 502a and 502b and a separate audio stream from each other, for example.

[0049] The inventors evaluated the effectiveness of dynamic mixing by investigating the effect of network congestion on audio quality. Two quantifiable aspects of audio quality were considered, the level of packet loss experienced and the degree of audio stream spatialisation, that is the number of separate audio streams delivered to an end system. These two criteria were chosen since they both relate to the end user's perceived experience of the system and they can also be readily determined from measurements of the system 500, for example the number of packets being sent per second. The first measure, the level of packet loss experienced, is the primary determinant of whether a network audio stream will be intelligible to the user and therefore of any use at all. Audio codecs that encode audio streams in 40ms to 80ms packets and utilise silence substitution for packet loss recovery typically become unintelligible if 15% or more of the packets are lost during transmission. Other factors such as delay or jitter are of secondary importance

by comparison. The second measure, the degree of spatialisation, is the primary distinguishing feature between the peer-to-peer, fully mixed and partially mixed approaches. For instance, research has shown that spatialised audio is a key factor in providing users with a sense of presence in a virtual environment.

[0050] Referring to Figure 6 which shows a system 600 configured from the system of Figure 5 for evaluating the effect of network congestion on audio quality. In Figure 6 the LANs 506 and 512 are assumed to be generally congestion free high bandwidth networks connected via a lower bandwidth shared WAN 520 which is prone to congestion. Six end

[0051] user systems 502 are provided on LAN 506 for simulating network usage. A single end user system is provided on LAN 512. A WAN simulation tool is provided for simulating network delays and bandwidth restrictions for a limited bandwidth WAN connection. An additional application 602 is provided for introducing controlled levels of competing traffic onto the simulated WAN connection 520 in order to create network congestion.

[0052] All packets on the system 600 are monitored and analysed to classify the packets and to measure the number of audio streams in transit and the amount of competing traffic. Packet loss experienced by the audio streams in transit is measured by matching the number of packets leaving LAN 506 to those arriving on LAN 512.

[0053] The following strategy was used to evaluate the system of Figure 6:-

- Six simulated users on LAN 506 continuously sending audio data to a single user on LAN 512.
- Virtual WAN bandwidth limit of 500 Kbit/second corresponding to just over seven audio streams for 8KHz, 8bit, Ulaw, mono audio encoding. A WAN buffer size of 250 Kbits.
- Eight levels of competing (congestion-inducing) traffic: 0, 70, 140, 210, 280, 350, 420, and 490 Kbits/s.
- Three audio distribution strategies: forward all audio streams without mixing (equivalent to peer-to-peer multicast), mix all audio streams before forwarding (equivalent to total mixing at LAN 1), and mix a dynamic subset of audio streams (partial mixing).

[0054] In the final distribution strategy dynamic or partial mixing was used to keep the packet loss rate below 15% whilst maintaining the maximum number of separate audio streams.

[0055] The experimental results are shown in Figures 7 and 8. Figure 7 shows the effect that increasing levels of congestion has on the packet loss rate experienced for each of the three audio distribution strategies. The peer-to-peer approach (line 700 in the graph of Figure 7) experiences increasing levels of packet loss as competing traffic increases. The packet loss rate exceeds 15%

at 210 Kbits/s of additional traffic. Full-mixing (line 702 in the drawing) uses the minimum bandwidth throughout, and only starts to experience congestion when the competing traffic reaches 490 Kbits/s. Distributed partial mixing (line 704 in the drawing) gives higher loss rates than full mixing, but much lower rates than all-forwarding peer-to-peer, and maintains its loss rate below 15% even with 490 Kbit/s of competing traffic (as for full mixing).

[0056] Figure 8 shows the number of separate audio streams being transmitted to a listener on the end user system on LAN 512. For all-forwarding (line 800 in the graph of Figure 8), six streams are always transmitted by the LAN 506, however none of these arrive in any useful form when competing traffic levels exceed 210 Kbits/s. For total mixing (line 802 in Figure 8), one stream is always sent. Dynamic mixing (line 804 in Figure 8) lies between these two extremes. With no congestion, six distinct streams are transmitted over the WAN connection from the WAN 506 to the WAN 512. As competing traffic, and hence congestion increases, dynamic mixing reduces the number of distinct streams by mixing more audio streams together. When competing traffic levels reach 490 Kbits/s dynamic mixing falls back to total mixing, with only a single stream sent over the WAN.

[0057] The above described investigation demonstrates that dynamic or distributed partial mixing combines the benefits of both peer-to-peer and total mixing audio services. With sufficient bandwidth, the system operates like a peer-to-peers system, delivering independent audio streams to each listener, giving maximum individual flexibility and control over what users hear. As bandwidth becomes restricted the distributed partial mixing scheme moves incrementally towards a totally mixed (minimum bandwidth) service, thereby preserving a useful level of audio communication under a wide range of network conditions.

[0058] More generally, the distributed partial mixing approach has the following distinctive benefits: it is adaptive, reacting to network congestion in a way that peer-to-peer systems cannot; it supports dynamic load balancing between different distributed components of the audio service; it readily supports heterogeneous networks and different end user terminal capabilities; and it is adaptive to varying application requirements.

Claims

1. A method of processing audio data streams transmitted in a communications network; said method comprising the steps of:-

- i) receiving a plurality of audio data streams transmitted from one or more audio data stream transmitters distributed in the network (404); and **characterized by**
- ii) processing data relating to at least one respective network resource parameter to deter-

mine respective network resources available for subsequent communication of said audio data streams to at least one respective audio data stream receiver in the network (408); and,

- iii) comparing said available resources with respective network resource requirements necessary for communicating said audio streams to at least one respective audio data stream receiver in the network (410, 412); and,
- iv) determining whether to mix selected audio data streams prior to transmission in response to said comparison (418, 420).

2. A method according to claim 1 further comprising the step of:-

- v) processing two or more audio streams in response to said comparison to provide at least one mixed audio data stream for subsequent transmission in said network.

3. A method according to claim 1 or claim 2 wherein steps ii) and iii) comprise the steps of:-

- determining a current value for the or each respective network resource parameter, and comparing the or each respective current resource parameter value with a respective minimum resource threshold value necessary for communicating said unmixed audio data streams to the or each respective receiver.

4. A method according to claim 3 wherein said minimum resource threshold value is determined according to at least one pre-defined quality of service parameter.

5. A method according to claim 4 wherein said network is a packet switched network and said pre-defined quality of service parameter is defined by a maximum packet loss rate.

6. A method according to any preceding claim wherein one network resource parameter relates to available network bandwidth for transmission of said audio data streams to the or each respective receiver.

7. A method according to claim 6 wherein said available bandwidth capacity is determined by user specific quality of service requirements.

8. A method according to any preceding claim wherein one network resource parameter relates to receiver processing characteristics.

9. A method according to any preceding claim wherein said audio data streams are selected for mixing according to predetermined criteria.

10. A method according to claim 9 wherein said audio streams are mixed according to respective audio stream content.

11. A method according to claim 9 wherein said audio streams are mixed according to respective recipient requirements.

12. A method according to claim 9 wherein said audio streams are mixed according to respective audio stream sources.

13. A method according to claim 9 wherein said audio streams are mixed according to respective receiver audio data stream processing capabilities.

14. A software program component for processing audio data streams transmitted in a communications network; said program being arranged to execute the following steps when run on a computer:

- i) receive a plurality of audio data streams transmitted from one or more audio data stream transmitters distributed in the network; and
- characterized by :**
- ii) process data relating to at least one respective network resource parameter to determine respective network resources available for subsequent transmission of said audio data streams in the network; and,
- iii) compare said available resources with respective network resource requirements necessary for transmission of said audio streams to at least one respective audio data stream receiver in the network; and,
- iv) determine whether to mix selected audio data streams prior to transmission in response to said comparison.

15. A system for processing audio data streams transmitted in a communications network; said system comprising:-

- i) a receiver (301) for receiving a plurality of audio data streams transmitted from one or more audio data stream transmitters distributed in the network; and,
- characterized by**
- ii) a processor (104) for processing data relating to at least one respective network resource parameter to determine respective network resources available for subsequent transmission of said audio data streams in the network; and,
- iii) a comparator for comparing said available resources with respective network resource requirements necessary for transmission of said audio streams to at least one respective audio data stream receiver in the network; and,
- iv) determining means for determining whether

to mix selected audio data streams prior to transmission in response to said comparison.

5 Patentansprüche

1. Verfahren zur Verarbeitung von Audiodatenströmen, die in einem Kommunikationsnetzwerk übertragen werden, wobei das Verfahren die folgenden Schritte umfasst:

- i) Empfangen von mehreren Audiodatenströmen (404), die von einem oder von mehreren in dem Netzwerk verteilten Audiodatenstromsendern gesendet werden, und
- gekennzeichnet ist durch**
- ii) Verarbeiten von Daten (408), die sich auf zumindest einen jeweiligen Netzwerkressourcenparameter zum Ermitteln entsprechender Netzwerkressourcen beziehen, die für eine nachfolgende Übertragung der Audiodatenströme an zumindest einen jeweiligen Audiodatenstromempfänger des Netzwerks verfügbar sind, und
- iii) Vergleichen der verfügbaren Ressourcen mit dem jeweiligen Netzwerkressourcenbedarf (410, 412), der für die Übertragung der Audiodatenströme an zumindest einen jeweiligen Audiodatenstromempfänger des Netzwerks erforderlich ist, und
- iv) Bestimmen, ob ausgewählte Audiodatenströme vor der Übertragung als Reaktion auf den Vergleich zu mischen sind (418, 420).

2. Verfahren nach Anspruch 1, das ferner den Schritt umfasst zum

- v) Verarbeiten von zwei oder mehr Audiodatenströmen als Reaktion auf den Vergleich, um für die nachfolgende Übertragung in dem Netzwerk zumindest einen gemischten Audiodatenstrom herzustellen.

3. Verfahren nach Anspruch 1 oder 2, worin die Schritte ii) und iii) die Schritte zum

Bestimmen eines aktuellen Werts für den oder für jeden der jeweiligen Netzwerkressourcenparameter und zum Vergleichen des oder eines jeden der jeweiligen aktuellen Netzwerkressourcenparameterwerte mit einem entsprechenden Minimalressourcenschwellwert umfassen, der zur Übertragung der nicht gemischten Audiodatenströme an den oder jeden der jeweiligen Empfänger erforderlich ist.

4. Verfahren nach Anspruch 3, worin der Minimalressourcenschwellwert zumindest einem vorgegebenen Dienstgüteparameter entsprechend festgelegt wird.

5. Verfahren nach Anspruch 4, worin das Netzwerk als Datenpaketvermittlungsnetzwerk ausgebildet ist und der vorgegebene Dienstgüteparameter durch eine maximale Paketverlustrate bestimmt ist.

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6. Verfahren nach einem der vorhergehenden Ansprüche, worin ein Netzwerkressourcenparameter auf die verfügbare Netzwerkbandbreite zur Übertragung der Audiodatenströme an den oder jeden der jeweiligen Empfänger bezogen ist.

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7. Verfahren nach Anspruch 6, worin die verfügbare Bandbreitenkapazität über benutzerspezifische Dienstgütevorgaben bestimmt wird.

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8. Verfahren nach einem der vorhergehenden Ansprüche, worin ein Netzwerkressourcenparameter auf eine Empfängerverarbeitungseigenschaft bezogen ist.

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9. Verfahren nach einem der vorhergehenden Ansprüche, worin die Audiodatenströme nach einem vorgegebenen Kriterium zum Mischen ausgewählt werden.

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10. Verfahren nach Anspruch 9, worin die Audiodatenströme nach dem jeweiligen Inhalt der Audiodatenströme gemischt werden.

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11. Verfahren nach Anspruch 9, worin die Audiodatenströme nach den jeweiligen Empfängervoraussetzungen gemischt werden.

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12. Verfahren nach Anspruch 9, worin die Audiodatenströme entsprechend den jeweiligen Audiodatenstromquellen gemischt werden.

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13. Verfahren nach Anspruch 9, worin die Audiodatenströme entsprechend dem jeweiligen Vermögen der Empfänger zur Verarbeitung von Audiodatenströmen gemischt werden.

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14. Datenverarbeitungsprogrammkomponente zum Verarbeiten von Audiodatenströmen, die in einem Kommunikationsnetzwerk übertragen werden, wobei das Programm dazu ausgebildet ist, bei einer Ausführung auf einer Datenverarbeitungsanlage die folgenden Schritte auszuführen:

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i) Empfangen von mehreren Audiodatenströmen, die von einem oder von mehreren in dem Netzwerk verteilten Audiodatenstromsendern gesendet werden, und

gekennzeichnet ist durch

ii) Verarbeiten von Daten, die sich auf zumindest einen jeweiligen Netzwerkressourcenparameter zum Ermitteln entsprechender Netzwerkressourcen beziehen, die für eine nachfolgende

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Übertragung der Audiodatenströme in dem Netzwerk verfügbar sind, und

iii) Vergleichen der verfügbaren Ressourcen mit dem jeweiligen Netzwerkressourcenbedarf, der für die Übertragung der Audiodatenströme an zumindest einen jeweiligen Audiodatenstromempfänger des Netzwerks erforderlich ist, und
iv) Bestimmen, ob ausgewählte Audiodatenströme vor der Übertragung als Reaktion auf den Vergleich zu mischen sind.

15. Vorrichtung zur Verarbeitung von Audiodatenströmen, die in einem Kommunikationsnetzwerk übertragen werden, wobei die Vorrichtung umfasst:

i) einen Empfänger (301) zum Empfangen von mehreren Audiodatenströmen, die von einer oder von mehreren in dem Netzwerk verteilten Audiodatenstromsendern gesendet werden, und

gekennzeichnet ist durch

ii) einen Prozessor (104) zum Verarbeiten von Daten, die sich auf zumindest einen jeweiligen Netzwerkressourcenparameter zum Ermitteln entsprechender Netzwerkressourcen beziehen, die für eine nachfolgende Übertragung der Audiodatenströme in dem Netzwerk verfügbar sind, und

iii) einen Komparator zum Vergleichen der verfügbaren Ressourcen mit dem jeweiligen Netzwerkressourcenbedarf, der für die Übertragung der Audiodatenströme an zumindest einen jeweiligen Audiodatenstromempfänger des Netzwerks erforderlich ist, und

iv) eine Bestimmungseinrichtung zum Bestimmen, ob ausgewählte Audiodatenströme vor der Übertragung als Reaktion auf den Vergleich zu mischen sind.

Revendications

1. Procédé de traitement de trains de données audio transmis dans un réseau de communication, ledit procédé comprenant les étapes suivantes :

i) la réception d'une pluralité de trains de données audio émis depuis un ou plusieurs émetteurs de trains de données audio répartis dans le réseau (404), et **caractérisé par** :

ii) le traitement de données se rapportant à au moins un paramètre de ressource de réseau respective pour déterminer des ressources de réseau respectives disponibles pour une communication ultérieure desdits trains de données audio à au moins un récepteur de trains de données audio respectif dans le réseau (408), et
iii) la comparaison desdites ressources disponi-

- bles aux exigences de ressources de réseau respectives nécessaires pour communiquer lesdits trains de données audio à au moins un récepteur de trains de données audio respectif dans le réseau (410, 412), et
iv) la détermination du mélange ou non de trains de données audio sélectionnés avant la transmission, en réponse à ladite comparaison (418, 420).
2. Procédé selon la revendication 1, comprenant en outre l'étape suivante :
- v) le traitement de deux trains de données audio ou plus en réponse à ladite comparaison pour procurer au moins un train de données audio mélangé en vue d'une transmission ultérieure dans ledit réseau.
3. Procédé selon la revendication 1 ou la revendication 2, dans lequel les étapes ii) et iii) comprennent les étapes consistant à :
- déterminer une valeur en cours pour le ou chaque paramètre de ressource de réseau respectif, et
comparer la ou chaque valeur de paramètre de ressource en cours respectif à une valeur de seuil de ressource minimale respective nécessaire pour communiquer lesdits trains de données audio non mélangés au récepteur ou à chaque récepteur respectif.
4. Procédé selon la revendication 3, dans lequel ladite valeur de seuil de ressource minimale est déterminée conformément à au moins un paramètre de qualité de service prédéfini.
5. Procédé selon la revendication 4, dans lequel ledit réseau est un réseau commuté par paquets et ledit paramètre de qualité de service prédéfini est défini par un taux de perte de paquets maximal.
6. Procédé selon l'une quelconque des revendications précédentes, dans lequel un paramètre de ressource de réseau se rapporte à une bande passante de réseau disponible pour la transmission desdits trains de données audio au récepteur ou à chaque récepteur respectif.
7. Procédé selon la revendication 6, dans lequel ladite capacité de bande passante disponible est déterminée par les exigences de qualité de service spécifiques à un utilisateur.
8. Procédé selon l'une quelconque des revendications précédentes, dans lequel un paramètre de ressource de réseau se rapporte à des caractéristiques de traitement de récepteur.
9. Procédé selon l'une quelconque des revendications précédentes, dans lequel lesdits trains de données audio sont sélectionnés en vue d'un mélange conformément à des critères prédéterminés.
10. Procédé selon la revendication 9, dans lequel lesdits trains de données audio sont mélangés conformément à des contenus des trains de données audio respectifs.
11. Procédé selon la revendication 9, dans lequel lesdits trains de données audio sont mélangés conformément à des exigences de destinataires respectives.
12. Procédé selon la revendication 9, dans lequel lesdits trains de données audio sont mélangés conformément à des sources de trains de données audio respectives.
13. Procédé selon la revendication 9, dans lequel lesdits trains de données audio sont mélangés conformément à des capacités de traitement de trains de données audio de récepteurs respectifs.
14. Composant de programme logiciel destiné à traiter des trains de données audio transmis dans un réseau de communication, ledit programme étant conçu pour exécuter les étapes suivantes quand il est exécuté sur un ordinateur :
- i) la réception d'une pluralité de trains de données audio émis depuis un ou plusieurs émetteurs de trains de données audio répartis dans le réseau, et **caractérisé par** :
- ii) le traitement de données se rapportant à au moins un paramètre de ressource de réseau respectif pour déterminer des ressources de réseau respectives disponibles en vue d'une transmission ultérieure desdits trains de données audio dans le réseau, et
iii) la comparaison desdites ressources disponibles à des exigences de ressources de réseau respectives nécessaires à la transmission desdits trains de données audio à au moins un récepteur de trains de données audio respectif dans le réseau, et
iv) la détermination du mélange ou non de trains de données audio sélectionnés avant la transmission, en réponse à ladite comparaison.
15. Système de traitement de trains de données audio transmis dans un réseau de communication, ledit système comprenant :
- i) un récepteur (302) destiné à recevoir une pluralité de trains de données audio émis depuis

un ou plusieurs émetteurs de trains de données audio répartis dans le réseau, et **caractérisé par** :

- ii) un processeur (104) destiné à traiter des données se rapportant à au moins un paramètre de ressource de réseau respectif pour déterminer des ressources de réseau respectives disponibles en vue d'une transmission ultérieure desdits trains de données audio dans le réseau, et
- iii) un comparateur destiné à comparer lesdites ressources disponibles auxdites exigences de ressources de réseau respectives nécessaires pour la transmission desdits trains de données audio à au moins un récepteur de trains de données audio respectif dans le réseau, et
- iv) un moyen de détermination destiné à déterminer s'il faut mélanger des trains de données audio sélectionnés avant la transmission, en réponse à ladite comparaison.

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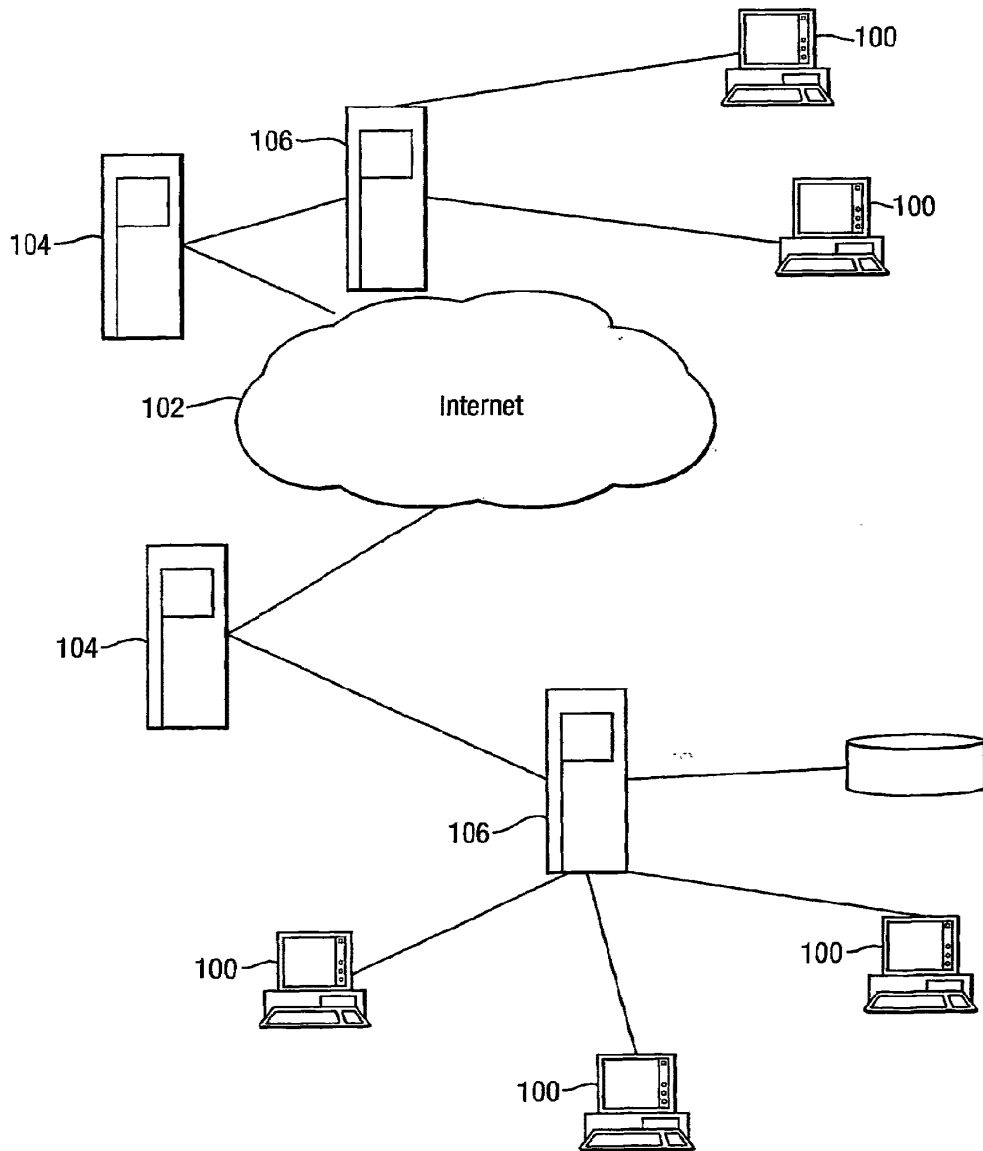
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Fig.1.



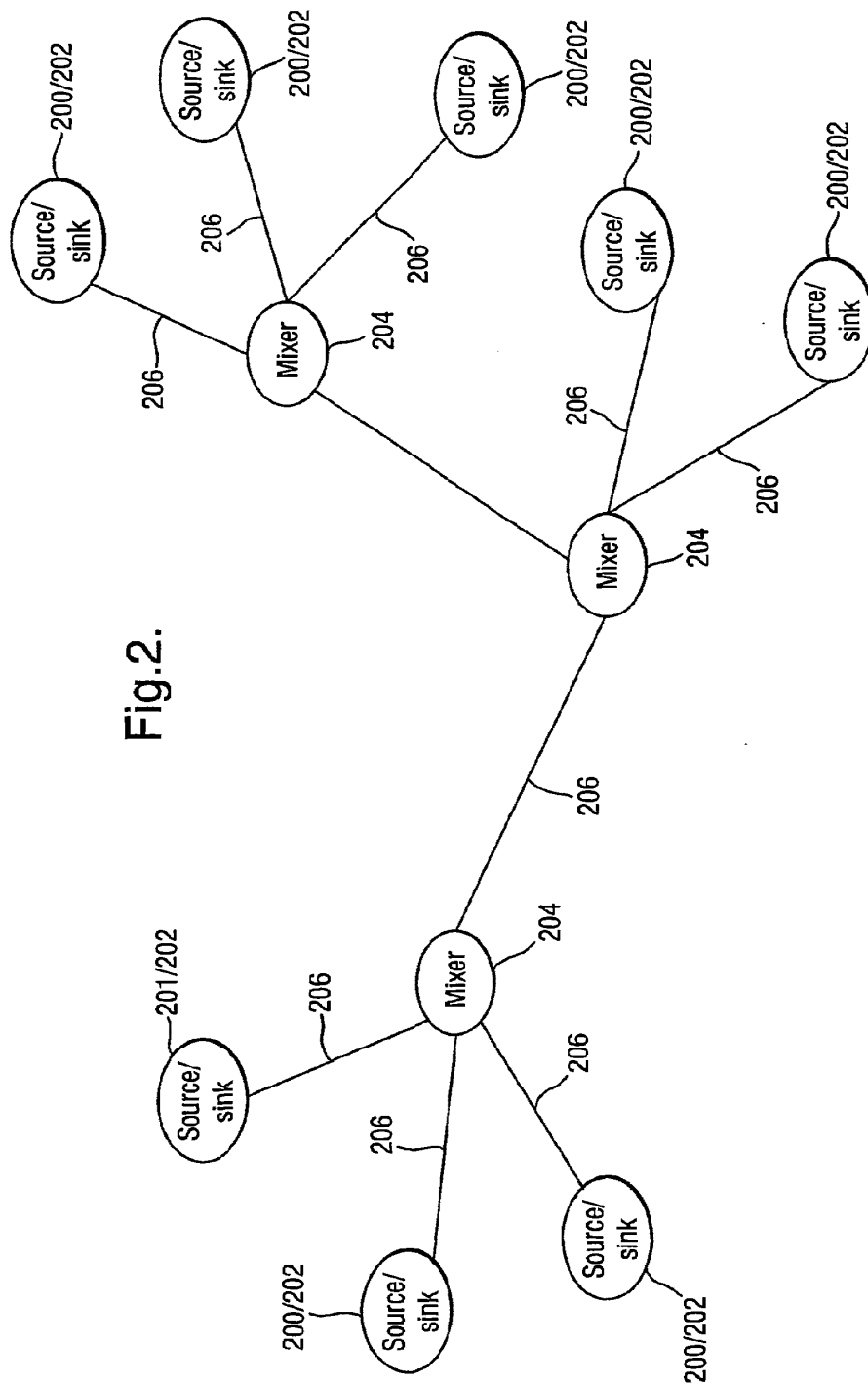
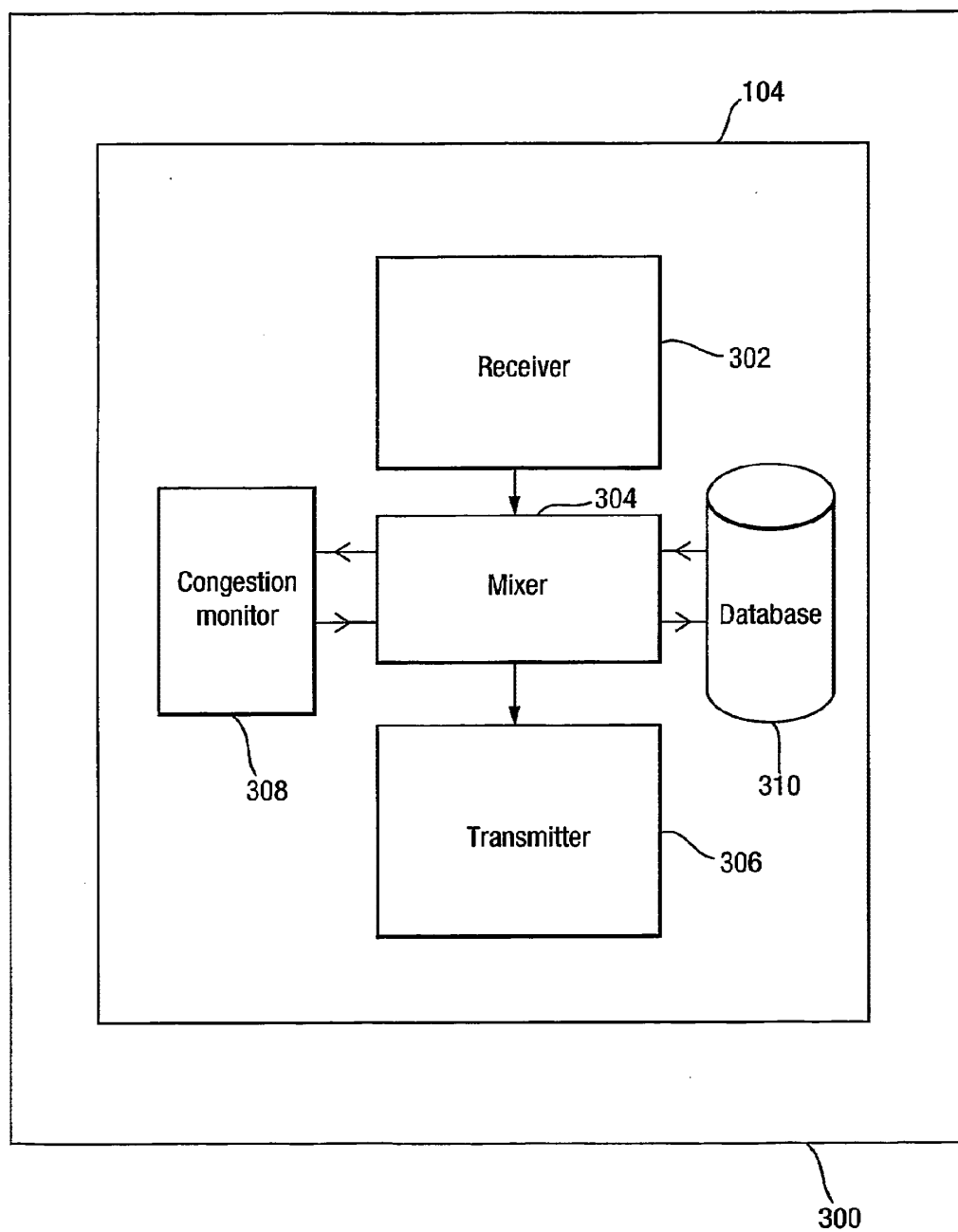


Fig. 2.

Fig.3.



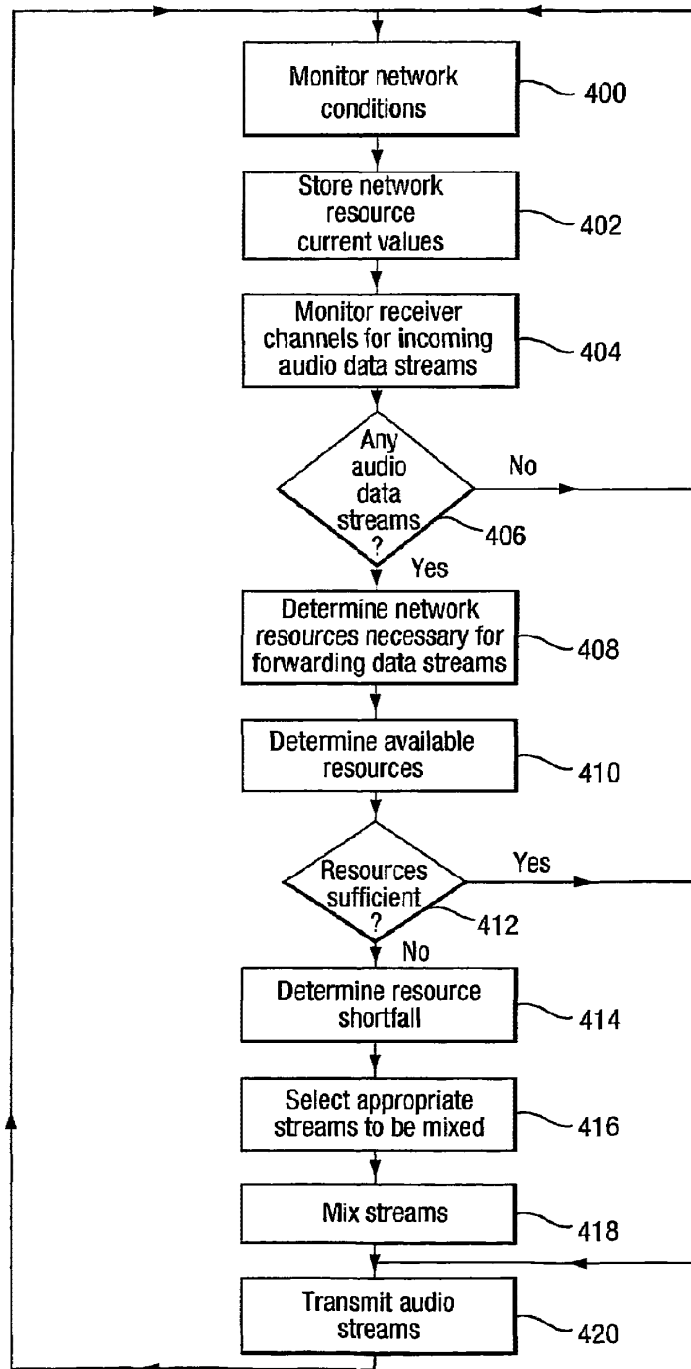
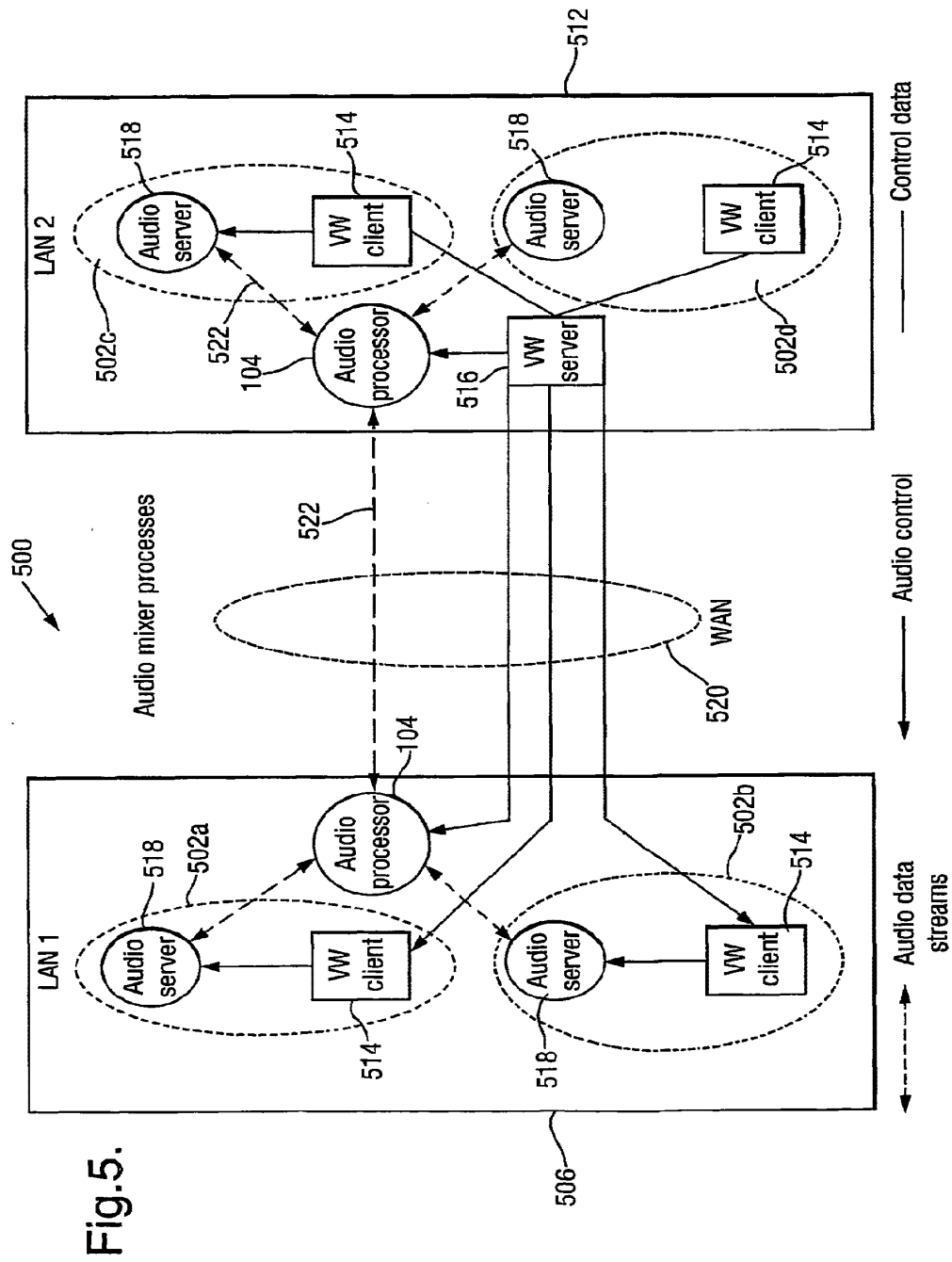


Fig.4.



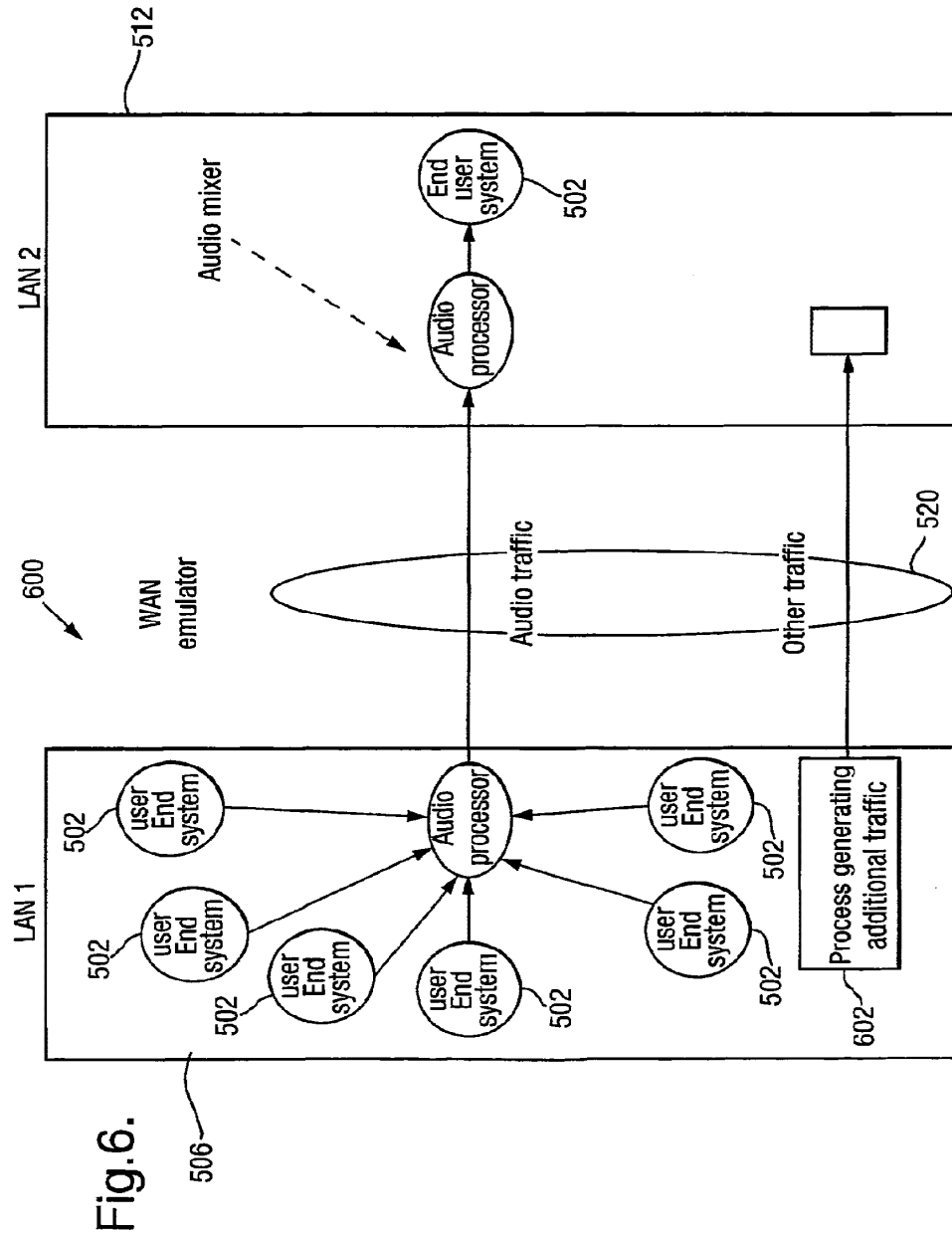
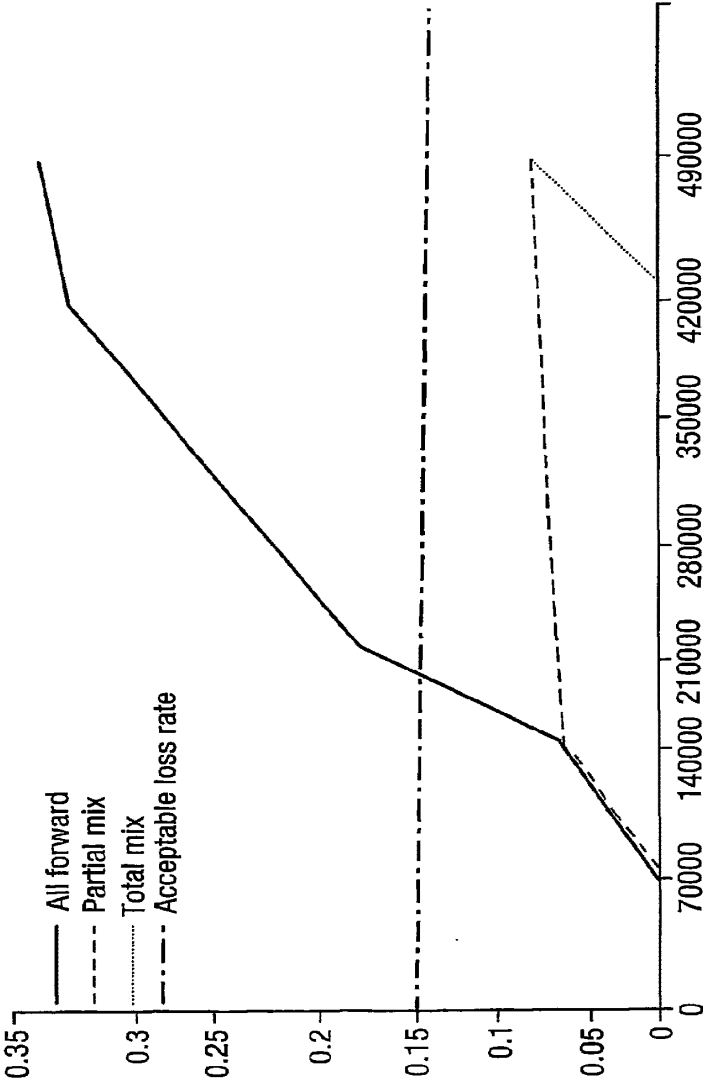
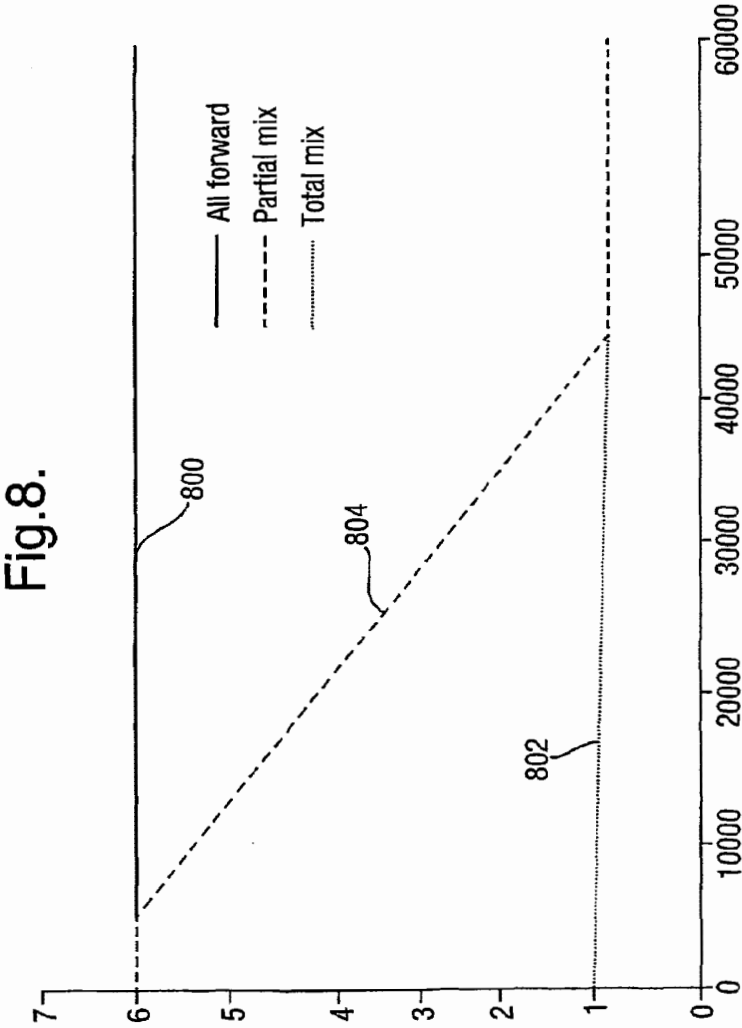


Fig.7.





REFERENCES CITED IN THE DESCRIPTION

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(54) **AUDIO CHANNEL TRANSLATION**
AUDIOKANALÜBERSETZUNG
MODULATION DE CANAL AUDIO

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(73) Proprietor: **DOLBY LABORATORIES LICENSING
CORPORATION**
San Francisco
California 94103 (US)

(72) Inventor: **DAVIS, Mark, Franklin**
San Francisco, CA 94103 (US)

(74) Representative: **Hoffmann, Eckart**
Patentanwalt,
Bahnhofstrasse 103
82166 Gräfelfing (DE)

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EP 1 410 686 B1

Description**TECHNICAL FIELD**

5 **[0001]** The invention relates to audio signal processing. More particularly the invention relates to translating M audio input channels representing a soundfield to N audio output channels representing the same soundfield, wherein each channel is a single audio stream representing audio arriving from a direction, M and N are positive whole integers, and M is at least 2.

10 **BACKGROUND ART**

[0002] Although humans have only two ears, we hear sound as a three dimensional entity, relying upon a number of localization cues, such as head related transfer functions (HRTFs) and head motion. Full fidelity sound reproduction therefore requires the retention and reproduction of the full 3D soundfield, or at least the perceptual cues thereof. 15 Unfortunately, sound recording technology is not oriented toward capture of the 3D soundfield, nor toward capture of a 2D plane of sound, nor even toward capture of a 1D line of sound. Current sound recording technology is oriented strictly toward capture, preservation, and presentation of zero dimensional, discrete channels of audio.

[0003] Most of the effort on improving fidelity since Edison's original invention of sound recording has focused on ameliorating the imperfections of his original analog modulated-groove cylinder/disc media. These imperfections included limited, uneven frequency response, noise, distortion, wow, flutter, speed accuracy, wear, dirt, and copying generation loss. Although there were any number of piecemeal attempts at isolated improvements, including electronic amplification, tape recording, noise reduction, and record players that cost more than some cars, the traditional problems of individual channel quality were arguably not finally resolved until the singular development of digital recording in general, and specifically the introduction of the audio Compact Disc. Since then, aside from some effort at further extending the quality 25 of digital recording to 24bits/96 kHz sampling, the primary efforts in audio reproduction research have been focused on reducing the amount of data needed to maintain individual channel quality, mostly using perceptual coders, and on increasing the spatial fidelity. The latter problem is the subject of this document.

[0004] Efforts on improving spatial fidelity have proceeded along two fronts: trying to convey the perceptual cues of a full sound field, and trying to convey an approximation to the actual original sound field. Examples of systems employing the former approach include binaural recording and two-speaker-based virtual surround systems. Such systems exhibit a number of unfortunate imperfections, especially in reliably localizing sounds in some directions, and in requiring the use of headphones or a fixed single listener position. 30

[0005] For presentation of spatial sound to multiple listeners, whether in a living room or a commercial venue like a movie theatre, the only viable alternative has been to try to approximate the actual original sound field. Given the discrete channel nature of sound recording, it is not surprising that most efforts to date have involved what might be termed conservative increases in the number of presentation channels. Representative systems include the panned-mono three-speaker film soundtracks of the early 50's, conventional stereo sound, quadraphonic systems of the 60's, five channel discrete magnetic soundtracks on 70mm films, Dolby surround using a matrix in the 70's, AC-3 5.1 channel sound of the 90's, and recently, Surround-EX 6.1 channel sound. "Dolby", "Pro Logic" and "Surround EX" are trademarks of Dolby Laboratories Licensing Corporation. To one degree or another, these systems provide enhanced spatial reproduction compared to monophonic presentation. However, mixing a larger number of channels incurs larger time and cost penalties on content producers, and the resulting perception is typically one of a few scattered, discrete channels, rather than a continuum soundfield. Aspects of Dolby Pro Logic decoding are described in U.S. Patent 4,799,260. Details of AC-3 are set forth in "Digital Audio Compression Standard (AC-3)," Advanced Television Systems Committee (ATSC), Document A/52, December 20, 1995 (available on the World Wide Web of the Internet at www.atsc.org/Standards/A52/a_52.doc). See also the Errata Sheet of July 22, 1999 (available on the World Wide Web of the Internet at www.dolby.com/tech/ATSC_err.pdf). 45

Insights Underlying Aspects of the Present Invention

50 **[0006]** The basis for recreating an arbitrary distribution in a source-free wave medium is provided by a theorem by Gauss that stipulates that a wave field within some region is completely specified by the pressure distribution along the boundary of the region. This implies that re-creation of the sound field in a concert hall within the confines of a living room is possible by conceptually placing the living room, walls impermeable to sound, within the concert hall, then electronically rendering the walls sonically transparent by festooning the outside of the walls with an infinite number of infinitesimal microphones, each connected with suitable amplification to a corresponding loudspeaker just inside the wall. By interposing a suitable recording medium between microphones and speakers, a complete, if impractical, system of accurate 3D sound reproduction is realized. The only remaining design task is to render the system practical. 55

[0007] A first step toward practicality can be taken by noting the signal of interest is bandlimited, at about 20 kHz, permitting the application of the Spatial Sampling theorem, a variant of the more common Temporal Sampling theorem. The latter holds that there is no loss of information if a continuous bandlimited temporal waveform is discretely sampled at a rate at least twice the highest frequency of the source. The former theory follows from the same considerations to stipulate that the spatial sampling interval must be at least twice as dense as the shortest wavelength in order to avoid information loss. Since the wavelength of 20 kHz in air is about 3/8", the implication is that an accurate 3D sound system can be implemented with an array of microphones and loudspeakers spaced no more than 3/16" apart. Extended over all surfaces of a typical 9'x12' room, this works out to about 2.5 million channels, a considerable improvement over an infinite number, but still impractical at this time. Still, it establishes the basic approach of using an array of discrete channels as spatial samples, from which the sound field can be recovered by application of appropriate interpolation.

[0008] Once the sound field is characterized, it is possible in principle for a decoder to derive the optimal signal feed for any output loudspeaker. The channels supplied to such a decoder will be referred to herein variously as "cardinal," "transmitted," and "input" channels, and any output channel with a location that does not correspond to the position of one of the cardinal channels will be referred to as an "intermediate" channel. An output channel may also have a location coincident with the position of a cardinal input channel.

[0009] It is therefore desirable to reduce the number of discrete channel spatial samples, or cardinal channels. One possible basis for doing so is the fact that, above 1500 Hz, the ear no longer follows individual cycles, only the critical band envelope. This might allow channel spacing commensurate with 1500 Hz, or about 3". This would reduce the total for the 9'x12' room to about 6000 channels, a useful saving of about 2.49 million channels compared to the previous arrangement.

[0010] In any case, further reduction in the number of spatial sampling channels is theoretically possible by appeal to psychoacoustic localization limits. The horizontal limit of resolution, for centered sounds, is about 1 degree of arc. The corresponding limit of vertical resolution is about 5 degrees. If this density is extended appropriately around a sphere, the result will still be a few hundred to a few thousand channels.

[0011] EP 1 054 575 A discloses a process for translating two audio input channels representing a soundfield to eight audio output channels representing the same soundfield, wherein each channel is a single audio stream representing audio arriving from a direction. A matrix generates, from the two input channels the eight output channels by a process that includes determining a measure of the correlation of the two input channels and the level interrelationships of the two input channels.

[0012] EP 1 001 549 discloses an encoding/decoding system having, on the encoding side, a mix and matrix circuit for translating six audio input channels representing a soundfield, wherein each input channel is a single audio stream representing audio arriving from a direction, to six intermediate signals. Each intermediate signal is associated with two or three input channels. On the decoding side this known system has another mix and matrix circuit for translating the six intermediate signals to six audio output channels respectively corresponding to the six input channels. The mix and matrix circuit on the encoding side includes: an adder which adds a first and a second one of the input channels to a first intermediate signal, a subtracter which subtracts the second input channel from the first input channel to generate a second intermediate signal, a combination of an adder, a 1/2 divider, and a subtracter which processes a third input channel, a fourth input channel, and a fifth input channel into a third intermediate signal, another adder which adds the fourth input channel and the fifth input channel to a fourth intermediate signal, a subtracter which subtracts the fifth input channel from the fourth input channel to generate a fifth intermediate signal, and a combination of a multiplier and a subtracter which processes the third input channel and the sixth input channel into the sixth intermediate signal.

[0013] It is an object of the invention to solve the problem of how to approximate, in a practical way, an actual original sound field for presentation to multiple listeners in a living room or a commercial venue like a movie theatre.

DISCLOSURE OF THE INVENTION

[0014] This object is achieved by a process as claimed in claim 1 and its preferred embodiment as claimed in claim 2.

[0015] In accordance with the present invention, a process translates M audio input channels representing a soundfield to N audio output channels representing the same soundfield, wherein each channel is a single audio stream representing audio arriving from a direction, M and N are positive whole integers, and M is at least two. One or more sets of output channels are generated, each set having one or more output channels. Each set is associated with two or more spatially adjacent input channels and each output channel in a set is generated by a process that includes determining a measure of the correlation of the two or more input channels and the level interrelationships of the two or more input channels.

[0016] In one aspect of the present invention, multiple sets of output channels are associated with more than two input channels and the process determines the correlation of input channels with which each set of output channels is associated according to a hierarchical order such that each set or sets is ranked according to the number of input channels with which its output channel or channels are associated, the greatest number of input channels having the highest ranking, and the processing processes sets in order according to their hierarchical order. Further according to an aspect of the

present invention, the processing takes into account the results of processing higher order sets.

[0017] The playback or decoding aspects of the present invention assume that each of the M audio input channels representing audio arriving from a direction was generated by a passive-matrix nearest-neighbor amplitude-panned encoding of each source direction (i.e., a source direction is assumed to map primarily to the nearest cardinal channel or channels), without the requirement of additional side chain information (the use of side chain or auxiliary information is optional), making it compatible with existing mixing techniques, consoles, and formats. Although such source signals may be generated by explicitly employing a passive encoding matrix, most conventional recording techniques inherently generate such source signals (thus, constituting an "effective encoding matrix"). The playback or decoding aspects of the present invention are also largely compatible with natural recording source signals, such as might be made with five real directional microphones, since, allowing for some possible time delay, sounds arriving from intermediate directions tend to map principally to the nearest microphones (in a horizontal array, specifically to the nearest pair of microphones).

[0018] A decoder or decoding process according to aspects of the present invention may be implemented as a lattice of coupled processing modules or modular functions (hereinafter, "decoding modules"), each of which is used to generate one or more output channels (or, alternatively, control signals usable to generate one or more output channels) from the two or more of the closest spatially adjacent cardinal channels associated with the decoding module. The output channels represent relative proportions of the audio signals in the closest spatially adjacent cardinal channels associated with the particular decoding module. As explained in more detail below, the decoding modules are loosely coupled to each other in the sense that modules share nodes and there is a hierarchy of decoding modules. Modules are ordered in the hierarchy according to the number of cardinal channels they are associated with (the module or modules with the highest number of associated cardinal channels is ranked highest). A supervisory routine function presides over the modules so that common node signals are equitably shared and higher-order decoder modules may affect the output of lower-order modules.

[0019] Each decoder module may, in effect, include a matrix such that it directly generates output signals or each decoder module may generate control signals that are used, along with the control signals generated by other decoder modules, to vary the coefficients of a variable matrix or the scale factors of inputs to or outputs from a fixed matrix in order to generate all of the output signals.

[0020] Decoder modules emulate the operation of the human ear to attempt to provide perceptually transparent reproduction. Each decoder module may be implemented as either a wideband or multiband structure or function, in the latter case with either a continuous filterbank, or a block-structure, for example, a transform-based processor, using, for example, the same essential processing in each band.

[0021] Although the basic invention relates generally to the spatial translation of M input channels to N output channels, wherein M and N are positive whole integers and M is at least two, another aspect of this invention is that the quantity of speakers receiving the N output channels can be reduced to a practical number by judicious reliance upon virtual imaging, that is the creation of perceived sonic images at positions in space other than where a loudspeaker is located. The most common use of virtual imaging is in the stereo reproduction of an image part way between two speakers, by panning a mono signal between the channels. Virtual imaging is not considered a viable technique for group presentation with a sparse number of channels, because it requires the listener to be equidistant from the two speakers, or nearly so. In movie theatres, for example, the left and right front speakers are too far apart to obtain useful phantom imaging of a center image to much of the audience, so, given the importance of the center channel as the source of much of the dialog, a physical center speaker is used instead.

[0022] However, as the density of the speakers is increased, a point will be reached where virtual imaging is viable between any pair of speakers for much of the audience, at least to the extent that pans are smooth; with sufficient speakers, the gaps between the speakers are no longer perceived as such. Such an array has the potential to be nearly indistinguishable from the 2 million array derived earlier.

[0023] In order to test aspects of the present invention, we deployed a horizontal array of 5 speakers on each wall, 16 total allowing for common corner speakers, plus a ring of 6 speakers above the listener at a vertical angle of about 45 degrees, plus a single speaker directly above, total 23, plus a subwoofer/LFE channel, total 24, all fed from a PC set up for 24-channel playback. Although by current parlance this system might be referred to as a 23.1 channel system, for simplicity it will be referred to as a 24-channel system herein.

[0024] FIG. 1 is a top plan view showing schematically an idealized decoding arrangement in the manner of the just-described test arrangement. Five wide range horizontal cardinal channels are shown as squares 1', 3', 5', 9' and 13' on the outer circle. A vertical channel, perhaps derived from the five wide range cardinals via correlation or generated reverberation, or separately supplied, is shown as the broken square 23' in the center. The twenty-three wide range output channels are shown as numbered filled circles 1-23. The outer circle of sixteen output channels is on a horizontal plane, the inner circle of six output channels is forty-five degrees above the horizontal plane. Output channel 23 is directly above one or more listeners. Five two-input decoding modules are illustrated as arrows 24-28 around the outer circle, connected between each pair of horizontal cardinal channels. Five additional two-input vertical decoding modules are illustrated as arrows 29-33 connecting the vertical channel to each of the horizontal cardinals. Output channel 21, the

elevated center rear channel, is derived from a three-input decoding module illustrated as arrows between output channel 21 and cardinal channels 9, 13 and 23. Thus, each module is associated with a respective pair or trio of closest spatially adjacent cardinal channels. Although the decoding modules represented in FIG. 1 have three, four or five output channels, a decoding module may have any reasonable number of output channels. An output channels may be located intermediate to one or more cardinal channels or at the same position as a cardinal channel. Thus, in the FIG. 1 example, each of the cardinal channel locations is also an output channel. Each input channel is shared by multiple decoding modules.

[0025] As will be discussed, a design goal of this invention is that the playback processor should be capable in concept of working with an arbitrary number and arrangement of speakers, so the 24-channel array will be used as an illustrative but non-unique example of the density and arrangement required to achieve a convincing continuum perceived soundfield according to one aspect of the invention.

[0026] The desire to be able to use a large, and possibly user-selectable, number of presentation channels raises the question of the number of discrete channels, and/or other information, that must be conveyed to the playback processor in order for it to derive, at least as one option, the twenty four channels described above. Obviously, one possible approach is simply to transmit twenty four discrete channels, but aside from the fact that it would likely be onerous for content producers to have to mix that many separate channels, and for a transmission medium to convey as many channels, it is preferred not to do so, as the 24-channel arrangement is merely one of many possible, and it is desired to allow for more or fewer presentation channels from a common transmitted signal array.

[0027] One way to recover output channels is to use formal spatial interpolation, a fixed weighted sum of the transmitted channels for each output, assuming the density of such channels is sufficiently great to allow for that. However, this would require from thousands to millions of transmitted channels, analogous to the use of a multihundred-tap FIR filter to perform temporal interpolation of a single signal. Reduction to a practical number of transmitted channels requires the application of psychoacoustic principles and more aggressive, dynamic interpolation from far fewer channels, still leaving unanswered the question of just how many channels are needed to impart the percept of a complete soundfield.

[0028] This question was addressed by an experiment performed by the present inventor some years ago, and recently replicated by another. The basis for the earlier experiment, at least, was the observation that conventional 2-channel binaural recording is capable of reproducing a realistic left/right image spread, but results in erratic front/back localization, owing in part to the imperfection of any HRTF employed, and the lack of head motion cues. To circumvent this drawback, a dualbinaural (4-channel) recording was made, using two pairs of directional microphones spaced commensurate to the size of the human head. One pair faced forward, the other to the rear. The resulting recording was played over four speakers spaced close to the head, to mitigate acoustic cross coupling effects. This arrangement provided realistic left/right timing and amplitude localization cues from each pair of speakers, plus unambiguous front/back information from the corresponding discrete positions of the microphones and speakers. The result was a singularly compelling surround sound presentation that lacked only a viable representation of height information. A recent experiment of another added a center front channel and two height channels, and was reported to be similarly realistic, perhaps even enhanced by the addition of height information.

[0029] Therefore, from both psychoacoustic considerations and empirical evidence, it appears that the relevant perceptual information can be conveyed in perhaps 4 to 5 "binaural-like" horizontal channels, plus perhaps one or more vertical channels. However, the signal crossfeed characteristic of binaural channel pairs makes them unsuitable for direct playback to a group via loudspeakers, since there is very little separation at midrange and low frequencies. So rather than introducing the crossfeed at the encoder (as is done for a binaural pair) only to have to undo it in the decoder, it is simpler and more direct to keep channels isolated, and to mix output channel signals from the nearest transmitted channels. Not only does this allow for direct playback through a like number of speakers without a decoder, if desired, plus optional downmix to fewer channels with a passive matrix decoder, but it essentially corresponds to the existing standard arrangement of 5.1 channels, at least in the horizontal plane. It is also largely compatible with natural recordings, such as might be made with five real directional microphones, since, allowing for some possible time delay, sounds arriving from intermediate directions will tend to map principally to the nearest microphones (in a horizontal array, specifically to the nearest pair of microphones).

[0030] Thus, from a perceptual standpoint, it should be possible for a channel translation decoder to accept a standard 5.1 channel program and convincingly present it through an arbitrary number of horizontally arrayed speakers, including the sixteen horizontal speakers of the twenty-four-channel array described earlier. With the addition of a vertical channel, such as is sometimes proposed for a digital cinema system, it should be possible to feed the entire twenty-four-channel array with individually derived, perceptually valid signals that together impart a continuum soundfield percept at most listening positions. Of course, if there is access to the fine grain source channels at the encoding site, additional information about them might be used to actively alter the encode matrix scale factors to pre-compensate for decoder limitations, or might simply be included as additional side-chain (auxiliary) information, perhaps similar to the coupling coordinates used in AC-3 (Dolby Digital) multichannel coding, but perceptually, such extra information should not be necessary; and practically, requiring the inclusion of such information is undesirable. The intended operation of the channel translation decoder is not limited to operation with 5.1 channel sources, and may use fewer or more, but there is at least some

justification to the belief that credible performance can be obtained from 5.1 channel sources.

[0031] This still leaves unanswered the question of just how to extract the intermediate output channels from a sparse array of transmitted channels. The solution proposed by one aspect of the present invention is to exploit again the notion of virtual imaging, but in a somewhat different way. It was previously noted that virtual imaging is not viable for group presentation with sparse speaker arrays because it required the listener to be nearly equidistant from each speaker. But it will work, after a fashion, for a listener who is fortuitously so placed, allowing the percept of intermediate phantom channels for signals that have been amplitude panned between the nearest real output channels. It is therefore proposed in one aspect of the present invention that the channel translation decoder consist of a series of modular interpolating signal processors, each in effect emulating an optimally placed listener, and each functioning in a manner analogous to the human auditory system to extract what would otherwise be virtual images from amplitude-panned signals, and feed them to real loudspeakers; the speakers preferably arrayed densely enough that natural virtual imaging can fill in the remaining gaps between them.

[0032] In general, each decoding module derives its inputs from the nearest transmitted cardinal channels, which, for example, for a canopy (overhead) array of speakers may be three or more cardinal channels. One way of generating output channels involving more than two cardinal channels might be to employ a series of pair-wise operations, with, e.g., outputs of some pair-wise decoding modules feeding the inputs of other modules. However, this has two drawbacks. One is that cascading decoding modules introduces multiple cascaded time constants, resulting in some output channels responding more quickly than others, causing audible position artifacts. The second drawback is that pair-wise correlation alone can only place intermediate or derived output channels along the line between the pair; use of three or more cardinals removes this restriction. Consequently, an extension to common pair-wise correlation has been developed to correlate three or more output signals; this technique is described below.

[0033] Horizontal localization in the human ear is predicated primarily upon two localization cues: interaural amplitude differences and interaural time differences. The latter cue is only valid for signal pairs in near time alignment, ± 600 microseconds or so. The practical effect is that phantom intermediate images will only occur at positions corresponding to a particular left/right amplitude difference, assuming the common signal content in the two real channels is correlated, or nearly so. (Note: two signals can have cross correlation values that span from +1 to -1. Fully correlated signals (correlation = 1) have the same waveform and time alignment, but may have different amplitudes, corresponding to off-center image positions.) As the correlation of a signal pair diminishes below 1, the perceived image will tend to spread, until, for two uncorrelated signals, there will be no intermediate image, only separate and distinct left and right images. Negative correlations are usually treated by the ear as similar to uncorrelated signal pairs, although the two images may appear to be spread wider. The correlations are carried out on a critical band basis, and above about 1500 Hz, the critical band signal envelopes are used instead of the signals themselves, to save human computational requirements (MIPS).

[0034] Vertical localization is a little more complex, relying on HRTF pinna cues and dynamic modulation of the horizontal cues with head motion, but the final effect is similar to horizontal localization with respect to panned amplitudes, cross correlation, and corresponding perceived image position and fusion. Vertical spatial resolution is, however, less precise than horizontal resolution, and does not require as dense an array of cardinal channels for adequate interpolation performance.

[0035] An advantage of using directional processors that emulate the operation of the human ear is that any imperfections or limitations of the signal processing should be perceptually masked by like imperfections and limitations of the human ear, allowing for the possibility that the system will be perceived as nearly indistinguishable from the original full continuum presentation.

[0036] Although the present invention is designed to make effective use of however many or few output channels are available (including playback via as many loudspeakers as there are input channels with no decoding, and passive mixdown to fewer channels, including mono, stereo and surround compatible Lt/Rt), it is preferably intended to employ a large and somewhat arbitrary, but nonetheless practical number of presentation channels/loudspeakers, and use as source material a similar or smaller number of encoded channels, including existing 5.1 channel surround tracks, and possible next-generation 11- or 12-channel digital cinema soundtracks.

[0037] Implementations of the present invention desirably should exhibit four principles: error containment, dominant containment, constant power, and synchronized smoothing.

[0038] Error containment refers to the notion that, given the likelihood of decoding errors, the decoded position of each source should be in some reasonable sense near its true, intended direction. This mandates a certain degree of conservatism in decoding strategy. Faced with the prospect of more aggressive decoding accompanied by possibly greater spatial disparity in the event of errors, it is usually preferable to accept less precise decoding in exchange for assured spatial containment. Even in situations in which more precise decoding can confidently be applied, it may be unwise to do so if there is a likelihood that dynamic signal conditions will require the decoder to ratchet between aggressive and conservative modes, resulting in audible artifacts.

[0039] Dominant containment, a more constrained variant of error containment, is the requirement that a single well-defined dominant signal should be panned by the decoder to only nearest neighbor output channels. This condition is

necessary to maintain image fusion for dominant signals, and contributes to the perceived discreteness of a matrix decoder. While a signal is dominant, it is suppressed from other output channels, either by subtracting it from the associated cardinal signals, or by directly applying to other output channels matrix coefficients complementary to those used to derive the dominant signal ("anti-dominant coefficients/signal").

[0040] Constant power decoding requires not only that the total decoded output power be equal to the input power, but also equates the input/output power of each channel and directional signal encoded in the conveyed cardinal array. This minimizes gain-pumping artifacts.

[0041] Synchronized smoothing applies to systems with signal dependent smoothing time constants, and requires that if any smoothing network within a decoding module is switched to a fast time constant mode, all other smoothing networks within the module be similarly switched. This is to avoid having a newly dominant directional signal appear to slowly fade/pan from the previous dominant direction.

DESCRIPTION OF THE DRAWINGS

[0042]

FIG. 1 is a schematic drawing showing a top plan view of an idealized decoder arrangement.

BEST MODE FOR CARRYING OUT THE INVENTION

Decoding Module

[0043] Because encoding any source direction is assumed to map primarily to the nearest cardinal channels, channel translation decoding is based on a series of semi-autonomous decoding modules which in a general sense recover output channels, particularly intermediate output channels, each usually from a subset of all the transmitted channels, in a fashion similar to the human ear.

[0044] In a fashion analogous to the human ear, the operation of the decoding module is based on a combination of amplitude ratios, to determine the nominal ongoing primary direction, and cross correlation, to determine the relative width of the image.

[0045] Using control information derived from the amplitude ratios and cross correlation, the processor then extracts output channel audio signals. Since this is best done on a linear basis, to avoid generation of distortion products, the decoder forms weighted sums of cardinal channels containing the signal of interest. (As explained below, it may also be desirable to include information about non-neighbor cardinals in the calculation of the weighted sum.) This limited but dynamic form of interpolation is more commonly referred to as matrixing. If, in the source, the desired signal is mapped (amplitude panned) to the nearest M cardinal channels, then the problem is one of M:N matrix decoding. In other words, the output channels represent relative proportions of the input channels.

[0046] Especially in the case of two-input decoding modules, this is much like the issue addressed by active 2:N matrix decoders, such as the now classic Dolby Pro Logic matrix decoder, with pairwise decoding module inputs corresponding to the Lt/Rt encoded signals

[0047] Note: The outputs of a 2:N matrix decoder are sometimes referred to as cardinal channels. This document, however, uses "cardinal" to refer to the input channels of the channel translation decoder.

[0048] There is, however, at least one significant difference between prior art active 2:N decoders and the operation of a decoding module according to the present invention. While the former use left/right amplitudes to indicate left/right position, as postulated as well for the channel translation decoder, they also use interchannel phase to indicate front/back position, relying specifically on the ratio of sum/difference of the Lt/Rt encoded channels.

[0049] There are two problems with such active 2:N decoder arrangements. One is that fully correlated (frontal), but off-center signals, for example, will result in a sum/difference ratio of less than infinity, incorrectly indicating a less-than-full-frontal position (similarly for full anti-correlated off-center rear signals). The result is a somewhat warped decoding space. The second drawback is that the positional mapping is many-to-one, introducing inherent decoding errors. For example, in a 4:2:4 matrix system, an uncorrelated Left-In and Right-In signal pair with no Front-In or Rear-In will map to the same net, uncorrelated Lt/Rt pair as will an uncorrelated Front-In/Back-In pair, with no Left-In/Right-In, or for that matter from all four inputs uncorrelated. The decoder, faced with an uncorrelated Lt/Rt signal pair, has no choice but to "relax the matrix", that is use a passive matrix that distributes sound to all output channels. It is incapable of decoding to a simultaneous Left-Out/Right-Out only, or Front-Out/Rear-Out only signal array.

[0050] The underlying problem is that the use of interchannel phase to code front/back position in N:2:N matrixing systems runs counter to the operation of the human ear, which does not use phase to judge front/back position. The present invention works best with at least three non-collinear cardinal channels, so that front/back position is indicated by the assumed directions of the cardinal channels, without assigning different directions depending on their relative

phases or polarities. As such, a pair of uncorrelated or anti-correlated channel translation cardinal signals unambiguously decodes to isolated cardinal-output channel signals, with no intermediate signal and no "rearward" direction indicated. (This, by the way, avoids the unfortunate "center pileup" effect in active 2:N decoders, in which uncorrelated Left-In and Right-In signals are presented with reduced separation because the decoder feeds sum and difference of these signals to center and surround channels.) Of course, it is possible in principle to spatially expand a Lt/Rt signal pair by cascading a 2:N decoder, N = 4, or 5, with an N:M channel translation system, but in that case any limitations of the 2:N decoder, such as center pileup, will be carried over to the channel multiplied outputs. It is also possible to combine these functions into a channel translation decoder configured to accept 2-channel Lt/Rt signals and, in such cases, modify its behavior to interpret negative correlation signals as having rearward orientation, leaving the rest of the processing largely intact. However, even in that case, decoding ambiguities resulting from having only two transmitted channels would remain.

[0051] Thus, each decoding module, especially those with two input channels, resembles a prior art active 2:N decoder, with the front/back detection disabled or modified, and an arbitrary number of output channels. Of course, it is a mathematical impossibility to use matrixing to uniquely extract a larger number of channels from a smaller number, as this basically involves N linear equations with M unknowns, M greater than N. Therefore, it is to be expected that the decoding module may at times exhibit less than perfect channel recovery in the presence of multiple active source direction signals. However, the human auditory system, limited to using just two ears, will tend to be subject to the same limitations, allowing the system to be perceived as discrete, even with all channels operating. Isolated channel quality, with other channels muted, is still a consideration to accommodate listeners that may be situated near one speaker.

[0052] To be sure, the ear is operating on a frequency-dependent basis, but given that most sonic images will be similarly correlated at all frequencies, together with the successful empirical experience with Pro Logic decoders as a wideband system, it is to be expected that a wideband channel translation system may also be capable of satisfactory performance in some applications. Multiband channel translation decoding should also be possible, using similar processing on a band-by-band basis, and using the same encoded signal in each case, so the number and bandwidth of individual bands can be left as a free parameter to the decoder implementer. Although multiband processing is likely to require higher MIPS than wideband processing, the computational demands may not be that much higher if the input signals are divided into data blocks and the process is carried out on a block basis.

[0053] Before describing an algorithm usable by the decoding modules of the present invention, consideration is first given to the problem of shared nodes.

Shared Nodes

[0054] If the cardinal channel groups used by the decoding modules were all independent, then the decoding modules themselves could be independent, autonomous entities. Such is, however, not usually the case. A given transmitted channel will in general share separate output signals with two or more neighboring cardinal channels. If independent decoding modules are used to decode the array, each will be influenced by output signals of neighboring channels, resulting in possibly serious decoding errors. In effect, two output signals of neighboring decoding modules will "pull", or gravitate, toward each other, because of the increased level of the common cardinal node containing both signals. If, as is likely to be the case, the signals are dynamic, so too will be the amount of interaction, leading to signal dependent dynamic positioning errors of a possibly highly objectionable nature. This problem does not arise with Pro Logic and other active 2:N decoding, since they use only a single, isolated channel pair as the decoder input.

[0055] Thus, it is necessary to compensate for the "shared node" effect. One possible way to do so would be to subtract one recovered signal from the common node before trying to recover the output signal of an adjacent decoding module sharing the common node. This is often not possible, so as a fall back, each decoding module estimates the amount of common output signal energy present at its input channels, and a supervisor routine then informs each module of its neighbors' output signal energy estimates.

Pair-wise calculation of common energy

[0056] For example, suppose cardinal channel pair A/B contains a common signal X along with individual, uncorrelated signals Y and Z:

$$A = 0.707X + Y$$

$$B = 0.707X + Z$$

where the scalefactors of $0.707 = \sqrt{0.5}$ provide a power preserving mapping to the nearest neighbor cardinal channels.

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$$RMS\ Energy(A) = \int A^2 dt = \overline{A^2} = \overline{(.707X + Y)^2} = \overline{(0.5X^2 + 0.707XY + Y^2)}$$

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$$= 0.5\overline{X^2} + 0.707\overline{XY} + \overline{Y^2}$$

Because X and Y are uncorrelated,

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$$\overline{XY} = 0$$

So:

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$$\overline{A^2} = 0.5\overline{X^2} + \overline{Y^2}$$

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i.e., Because X and Y are uncorrelated, the total energy in cardinal channel A is the sum of the energies of signals X and Y. Similarly:

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$$\overline{B^2} = 0.5\overline{X^2} + \overline{Z^2}$$

Since X, Y, and Z are uncorrelated, the averaged cross-product of A and B is:

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$$\overline{AB} = 0.5\overline{X^2}$$

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[0057] So, in the case of an output signal shared equally by two neighboring cardinal channels which may also contain independent, uncorrelated signals, the averaged cross-product of the signals is equal to the energy of the common signal component in each channel. If the common signal is not shared equally, i.e., it is panned toward one of the cardinals, the averaged cross-product will be the geometric mean between the energy of the common components in A and B, from which individual channel common energy estimates can be derived by normalizing by the square root of the ratio of the channel amplitudes. Actual time averages are computed with a leaky integrator having a suitable decay time constant, to reflect ongoing activity. The time constant smoothing can be elaborated with nonlinear attack and decay time options, and in a multiband system, may be scaled with frequency.

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Higher order calculation of common energy

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[0058] In order to derive the common energy of decoding modules with three or more inputs, it is necessary to form averaged cross-products of all the input signals. Simply performing pairwise processing of the inputs will fail to differentiate between separate output signals between each pair of inputs and a signal common to all.

[0059] Consider, for example, three cardinal channels, A, B, and C, made up of uncorrelated signals W, Y, Z, and common signal X:

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$$A = X + W$$

$$B = X + Y$$

$$C = X + Z$$

[0060] If the average cross-product is calculated, all terms involving combinations of W, Y, and Z will cancel, as in the second order calculation, leaving the average of X^3 :

$$\overline{ABC} = \overline{X^3}$$

[0061] Unfortunately, if X is a zero mean time signal, as expected, then the average of its cube is zero. Unlike averaging X^2 , which is positive for any nonzero value of X, X^3 has the same sign as X, so the positive and negative contributions will tend to cancel. Obviously, the same holds for any odd power of X, corresponding to an odd number of module inputs, but even exponents greater than 2 can also lead to erroneous results; for example four inputs with components (X, X, -X, -X) will have the same product/average as (X, X, X, X).

[0062] This problem has been resolved by employing a variant of the averaged product technique. Before being averaged, the sign of the each product is discarded by taking the absolute value of the product. The signs of each term of the product are examined. If they are all the same, the absolute value of the product is applied to the averager. If any of the signs are different from the others, the negative of the absolute value of the product is averaged. Since the number of possible same-sign combinations may not be the same as the number of possible different-sign combinations, a weighting factor comprised of the ratio of the number of same to different sign combinations is applied to the negated absolute value products to compensate. For example, a three-input module has two ways for the signs to be the same, out of eight possibilities, leaving six possible ways for the signs to be different, resulting in a scale factor of $2/6 = 1/3$. This compensation causes the integrated or summed product to grow in a positive direction if and only if there is a signal component common to all inputs of a decoding module.

[0063] However, in order for the averages of different order modules to be comparable, they must all have the same dimensions. A conventional second-order correlation involves averages of two-input multiplications and hence of quantities with the dimensions of energy or power. Thus the terms to be averaged in higher order correlations must be modified also to have the dimensions of power. For a kth order correlation, the individual product absolute values must therefore be raised to the power $2/k$ before being averaged.

[0064] Of course, regardless of the order, the individual input node energies of a module, if needed, can be calculated as the average of the square of the corresponding node signal, and need not be first raised to the kth power and then reduced to a second order quantity.

Shared nodes: Neighbor levels

[0065] By using averaged squares and modified cross-products of cardinal channel signals, the amount of common output channel signal energy can be estimated. The above example involved a single interpolation processor, but if one or more of the A/B(C) nodes were common to another module with its own common signal component, uncorrelated with any other signals, then the averaged cross-product computed above would not be affected, making the calculation inherently free of any image pulling effects. (Note: if the two output signals are not uncorrelated, they will tend to pull the decoders some, but should have a similar effect on the human ear, so again system operation should remain faithful to human audition.)

[0066] Once each decoding module has computed the estimated common output channel signal energy at each of its cardinal nodes, the supervisor routine function can inform neighboring modules of each others' common energy, at which point the extraction of the output channel signals can proceed as described below. The calculation of the common energy used by a module at a node must take into account the hierarchy of possibly overlapping modules of different order, and subtract the common energy of a higher order module from the estimated common energy of any lower order module sharing the same nodes.

[0067] For example, suppose there are two adjacent cardinal channels A and B, representing two horizontal directions, plus a cardinal channel C, representing a vertical direction, and further suppose the existence of an intermediate or derived output channel representing an interior direction (i.e. one within the limits of A, B and C), with signal energy X^2 . The common energy of a three-input module, with inputs (A, B, C), will be X^2 , but so will the common energy of two-

input modules (A, B), (B, C), and (A, C). If the common energy of A-connected modules (A, B, C), (A, B), and (A, C) is simply added, the result is $3X^2$, instead of X^2 . In order for the calculation of common node energy to be correct, the common energy of each higher order module is first subtracted from the estimate of the common energy of each overlapping lower-level module, so the common energy X^2 of higher order module (A, B, C) is subtracted from the common energy estimates of the two two-input modules, resulting in 0 in each case, and making the net common energy estimate at node A equal to $X^2 + 0 + 0 = X^2$.

Output Channel Signal Extraction

[0068] As has been noted, the process of recovering the ensemble of output channels from the transmitted channels in a linear fashion is basically one of matrixing, that is forming weighted sums of the cardinal channels to derive output channel signals. The optimal choice of matrix scale factors is generally signal dependent. Indeed, if the number of currently active output channels is equal to the number of transmitted channels (but representing different directions), making the system exactly constrained, it is mathematically possible to compute an exact inverse of the effective encoding matrix, and recover isolated versions of the source signals. Even if the number of active output channels is greater than the number of cardinals, it may still be possible to compute a matrix pseudo-inverse.

[0069] Unfortunately, there are problems with this approach, not the least of which is that it is computationally demanding, especially on a multiband basis, and oriented toward high accuracy floating point implementation. Even though intermediate signals are assumed to be panned to nearest neighbor cardinal channels, a mathematical inverse or pseudo-inverse of the effective encoding matrix will in general involve contributions from all cardinal channels to each output channel, because of the node sharing effect. If there are any imperfections in the decoding, as indeed there inevitably will be, a cardinal channel signal could be reproduced from an output channel far removed from it spatially, which is highly undesirable. In addition, pseudo-inverse calculations tend to produce minimum-RMS-energy solutions, which maximally spread the sound around, providing minimum separation; this is quite the opposite of the intention.

[0070] So, in order to implement a practical, fault-tolerant decoder in which spatial decoding errors are inherently contained, the same modular structure as was used for signal detection is employed for signal extraction.

[0071] Following are details of the extraction process by which output signals are recovered by a decoding module. Note that the effective position of each output channel connected to the module is assumed to be indicated by the amplitude ratio that would otherwise be needed to pan a signal to that physical location, i.e., the ratio of the effective matrix encoding coefficients corresponding to that direction. To avoid divide-by-zero problems, ratios are typically calculated as the quotient of one channel's matrix coefficient over the RMS sum of all of that input channels' matrix coefficients (usually 1). For example, in a two-input module with inputs L and R, the energy ratio used would be the L energy over the sum of the L and R energies ("L-ratio"), which has a well-behaved range of 0 to 1. If the two-input decoding module has five output channels with effective encoding matrix coefficient pairs of (1.0, 0), (0.89, 0.45), (0.71, 0.71), (0.45, 0.89) and (0, 1.0), the corresponding L-ratios are 1.0, 0.89, 0.71, 0.45, and 0, since each scale factor pair has an RMS sum of 1.0.

[0072] From the signal energy of each input node (cardinal channel) of the decoding module is subtracted any node-sharing signal energy claimed by neighboring decoding modules, resulting in normalized input signal power levels used for the remainder of the calculation.

[0073] The dominant direction indicator is calculated as the vector sum of the cardinal directions, weighted by the relative energy. For a two input module, this simplifies to being the L-ratio of the normalized input signal power levels.

[0074] The output channels bracketing the dominant direction are determined by comparing the dominant direction L-ratio of step two, to the L-ratios of the output channels. For example, if the L-ratio of the above five-output-decoding-module inputs is 0.75, the second and third output channels bracket dominant signal direction, since $0.89 > 0.75 > 0.71$.

[0075] Panning scale factors to map the dominant signal onto the nearest bracketing channels are calculated from the ratio of the anti-dominant signal levels of the channels. The anti-dominant signal associated with a particular output channel is the signal that results when the corresponding decoding module's input signals are matrixed with the output channel's anti-dominant matrix scale factors. An output channel's anti-dominant matrix scale factors are those scale factors with RMS sum = 1.0 which result in zero output when a single dominant signal is panned to the output channel in question. If an output channel's encode matrix scale factors are (A, B), then the anti-dominant scale factors of the channel are just (B, -A).

Proof

[0076] If a single dominant signal is panned to an output channel with encode scale factors (A, B), then the signal must have amplitudes (kA, kB), k the overall amplitude of the signal. Then the anti-dominant signal for that channel is $(kA \cdot B - kB \cdot A) = 0$.

[0077] So, if a dominant signal consists of two-input module input signals (x(t), y(t)) with input amplitudes normalized to RMS=1 (X, Y), the extracted dominant signal will be $\text{dom}(t) = Xx(t) + Yy(t)$. If the position of this signal is bracketed

by output channels having matrix scale factors (A, B) and (C, D) respectively, the dominant signal scale factor scaling dom(t) for the former channel will be:

$$SF(A,B) = \text{sqrt} ((DX - CY) / ((DX - CY) + (BX - AY))),$$

while the equivalent dominant signal scale factor for the latter channel will be:

$$SF(C,D) = \text{sqrt} ((BX - AY) / ((DX - CY) + (BX - AY))).$$

[0078] As the dominant direction is panned from one output channel to the other, these two scale factors move in opposite directions between zero and one with constant power sum.

[0079] The anti-dominant signal is calculated and panned with suitable gain scaling to all non-dominant channels. The anti-dominant signal is a matrixed signal lacking any of the dominant signal. If the inputs to a decoding module are (x(t), y(t)) with normalized amplitudes (X, Y), the dominant signal is $Xx(t)+Yy(t)$ and the anti-dominant signal is $Yx(t)-Xy(t)$, irrespective of the positions of the non-dominant output channels.

[0080] In addition to the dominant/anti-dominant signal distribution, a second signal distribution is calculated, using the "passive" matrix, which is basically the output channel matrix scale factors already discussed, scaled to preserve power.

[0081] The cross correlation of the decoding module input signals is calculated as the averaged cross-product of the input signals divided by the square root of the product of the normalized input levels.

[0082] Returning to details of the extraction process, the final output signals are then calculated as a weighted crossfade sum of the dominant and passive signal distributions, using the decoding module's input signal cross-correlation to derive the crossfade factor. For correlation=1, the dominant/anti-dominant distribution is used exclusively. As the correlation diminishes, the output signal array is broadened by cross-fading to the passive distribution, reaching completion at a low positive value of correlation, typically 0.2 to 0.4, depending on the number of output channels connected to the decoding module. As the correlation falls further, toward zero, the passive amplitude output distribution is progressively bowed outward, reducing the output channel levels, emulating the response of the human ear to such signals.

Vertical Processing

[0083] Most of the processing described so far applies to the extraction of output channel signals from neighboring cardinal channels, regardless of the direction of the output and cardinal channels. However, because of the horizontal orientation of the ears, human auditory localization tends to be less sensitive to interchannel correlation in the vertical direction than horizontally. To remain faithful to the operation of the human ear, it may be desirable to relax the correlation constraint in interpolation processors using vertically-oriented input channels, such as processing the correlation signal with a warping function before otherwise applying it. However, it may be that use of the same processing as for horizontal channels will not involve any audible penalty, which will simplify the structure of the overall decoder.

[0084] Strictly speaking, vertical information includes both sound from above and below, and the decoder structure described will work equally well with either, but in practice there is little natural sound normally perceived as coming from below, so such processing and channels can probably be omitted without seriously compromising the perceived spatial fidelity of the system.

[0085] That notion may have practical significance in the application of channel translation to existing 5.1 channel surround material, which, of course, lacks any vertical channel. However, it may contain vertical information, such as fly-overs, which are panned across many or all of the horizontal channels. Thus, it should be possible to extract a virtual vertical channel from such source material, by looking for correlations among non-neighboring channels or groups of channels. Where such correlations exist, they will usually indicate the presence of vertical information from above, rather than below the listener. In some instances, it may also be possible to derive virtual vertical information from a reverberation generator, perhaps keyed to a model of the intended listening environment. Once the virtual vertical channel is extracted or derived from the 5.1 -channel source, the expansion to larger numbers of channels, such as the 24-channel arrangement described earlier, can proceed as if a real vertical channel had been supplied.

Directional Memory

[0086] One respect in which the operation of the decoding module control generation described above is similar to a 2:N active decoder such as a Pro Logic decoder is that the only "memory" in the process is in the smoothing networks, which derive the basic control signals. At any one point in time, there is only one dominant direction and one value of input correlation; and signal extraction proceeds directly from these signals.

[0087] However, particularly in complex acoustical environments (like the archetypal cocktail party), the human ear exhibits a certain degree of positional memory, or inertia, in that a briefly dominant sound from a given direction that is clearly localized will result in other, less distinctly localizable sounds from that general direction to be perceived as coming from the same source.

[0088] It is possible to emulate this effect in the decoding modules (and, indeed, in Pro Logic decoding as well) by adding an explicit mechanism to keep track of recently dominant directions and, during intervals of directionally ambiguous signal conditions, weight the output signal distribution toward recently dominant directions. This can enhance the perceived reproduced discreteness and stability of complex signal arrays.

Modified Correlation and Selective Channel Mixing

[0089] As described, the spreading determination of each decoding module is based on the coincident cross correlation of its input signals. This may underestimate the amount of output signal content under some conditions. This will occur, for example, with a naturally recorded signal in which non-centered directions have slightly different arrival times, along with unequal amplitudes, resulting in a reduced correlation value. The effect may be exaggerated if wide-spaced microphones are used, with commensurately elongated interchannel delays. To compensate, the correlation calculation can be extended to cover a range of interchannel time delays, at the expense of slightly higher processing MIPS requirements. Also, since the neurons on the auditory nerve have an effective time constant of about 1 msec., more realistic correlation values may be obtained by first smoothing the rectified audio with a smoother having a 1 msec. time constant.

[0090] In addition, if a content producer has an existing 5.1 channel program with strongly uncorrelated channels, the evenness of the spread when processed with a channel translation decoder can be increased by slightly mixing adjacent channels, thereby increasing the correlation, which will cause the channel translation decoding module to provide a more even spread among its intermediate output channels. Such mixing can be done selectively, for example leaving the center front channel signal unmixed, to preserve the compactness of the dialog track.

Loudness Compression/Expansion

[0091] When the encoding process involves mixing a larger number of channels to a smaller number, there is a potential for clipping of the encoded signal if some form of gain compensation is not provided. This problem exists as well for conventional matrix encoding, but is potentially of greater significance for channel translation, because the number of channels being mixed to a given output channel is greater. To avoid clipping in such cases, an overall gain scale factor is derived by the encoder and conveyed in the encoded bitstream to the decoder. Normally, this value is 0 dB, but can be set to a nonzero attenuating value by the encoder to avoid clipping, with the decoder providing an equivalent amount of compensating gain.

[0092] If the decoder is used to process an existing multichannel that lacks such a scale factor program (e.g., an existing 5.1 channel soundtrack), it could optionally use a fixed scale factor with an assumed value (presumably 0 dB), or apply an expansion function based on signal level and/or dynamics, or possibly make use of available metadata, such as a dialog normalization value, to adjust the decoder gain.

[0093] The present invention and its various aspects may be implemented in analog circuitry, or more probably as software functions performed in digital signal processors, programmed general-purpose digital computers, and/or special purpose digital computers. Interfaces between analog and digital signal streams may be performed in appropriate hardware and/or as functions in software and/or firmware.

Claims

1. A process for translating M audio input channels (1', 3', 5', 9', 13') representing a soundfield to N audio output channels (1-23) representing the same soundfield, wherein each channel is a single audio stream representing audio arriving from a direction, M and N are positive whole integers, and M is a positive integer equal to two or more, **characterized by** comprising a plurality of decoding modules each associated with two or more spatially adjacent input channels (1', 3', 5', 9', 13'), wherein each input channel is shared among multiple modules, and each module either

- includes a matrix that generates, from the associated two or more input channels (1', 3', 5', 9', 13'), one or more output channels (1-23) each constituting a subset of said N channels, by a process that includes determining a measure of the correlation of the two or more input channels (1', 3', 5', 9', 13') and the level interrelationships of the two or more input channels (1', 3', 5', 9', 13'), or
 - generates, from the associated two or more input channels (1', 3', 5', 9', 13') by a process that includes determining a measure of the correlation of the two or more input channels (1', 3', 5', 9', 13') and the level interrelationships of the two or more input channels (1', 3', 5', 9', 13'), control signals that are used, along with control signals generated by other decoder modules, to vary the coefficients of a variable matrix to generate all of the output channels (1-23), or
 - generates, from the associated two or more input channels (1', 3', 5', 9', 13') by a process that includes determining a measure of the correlation of the two or more input channels (1', 3', 5', 9', 13') and the level interrelationships of the two or more input channels (1', 3', 5', 9', 13'), control signals that are used, along with control signals generated by other decoder modules to vary the scale factors of inputs to or outputs from a fixed matrix to generate all of the output channels (1-23).
2. The process according to claim 1 wherein the modules are hierarchically ordered according to their number of input channels (1', 3', 5', 9', 13') and a supervisor communicates with the modules to control the sharing of input signals in accordance with their hierarchical ordering.

Patentansprüche

1. Verfahren zur Umsetzung von M Audioeingangskanälen (1', 3', 5', 9', 13'), die ein Schallfeld darstellen, auf N Audioausgangskanäle (1-23), die dasselbe Schallfeld darstellen, wobei jeder Kanal ein einzelner Audiostrom ist, der aus einer Richtung ankommendes Audio darstellt, M und N positive ganze Zahlen sind und M eine positive ganze Zahl gleich zwei oder mehr ist, **dadurch gekennzeichnet, dass** es enthält:

eine Vielzahl von Decodiermodulen, von denen jedes mit zwei oder mehr räumlich benachbarten Eingangskanälen (1', 3', 5', 9', 13') verknüpft ist, wobei jeder Eingangskanal von mehreren Modulen gemeinsam benutzt wird, und jedes Modul entweder

- eine Matrix einschließt, die aus den verknüpften zwei oder mehr Eingangskanälen (1', 3', 5', 9', 13') einen oder mehr Ausgangskanäle (1-23), von denen jeder eine Untergruppe der N Kanäle darstellt, durch ein Verfahren erzeugt, das die Bestimmung eines Maßes der Korrelation der zwei oder mehr Eingangskanäle (1', 3', 5', 9', 13') und der Pegelzusammenhänge der zwei oder mehr Eingangskanäle (1', 3', 5', 9', 13') einschließt, oder
 - aus den verknüpften zwei oder mehr Eingangskanälen (1', 3', 5', 9', 13') durch ein Verfahren, das die Bestimmung eines Maßes der Korrelation der zwei oder mehr Eingangskanäle (1', 3', 5', 9', 13') und der Pegelzusammenhänge der zwei oder mehr Eingangskanäle (1', 3', 5', 9', 13') einschließt, Steuersignale erzeugt, die zusammen mit von anderen Decodiermodulen erzeugten Steuersignalen benutzt werden, um die Koeffizienten einer variablen Matrix zu verändern, um die Gesamtheit der Ausgangskanäle (1-23) zu erzeugen, oder
 - aus den verknüpften zwei oder mehr Eingangskanälen (1', 3', 5', 9', 13') durch ein Verfahren, das die Bestimmung eines Maßes der Korrelation der zwei oder mehr Eingangskanäle (1', 3', 5', 9', 13') und der Pegelzusammenhänge der zwei oder mehr Eingangskanäle (1', 3', 5', 9', 13') einschließt, Steuersignale erzeugt, die zusammen mit von anderen Decodiermodulen erzeugten Steuersignalen benutzt werden, um die Skalierungsfaktoren der Eingaben in oder der Ausgaben aus einer festen Matrix zu verändern, um die Gesamtheit der Ausgangskanäle (1-23) zu erzeugen.
2. Verfahren nach Anspruch 1, bei welchem die Module hierarchisch nach ihrer Anzahl von Eingangskanälen (1', 3', 5', 9', 13') geordnet sind und eine Kontrollinstanz mit den Modulen kommuniziert, um die gemeinsame Benutzung von Eingangssignalen entsprechend ihrer hierarchischen Ordnung zu steuern.

Revendications

1. Procédé permettant de convertir M canaux d'entrée audio (1', 3', 5', 9', 13') représentant un champ acoustique en N canaux de sortie audio (1-23) représentant le même champ acoustique, dans lequel chaque canal est un flux

audio unique représentant des sons provenant d'une direction, M et N sont des entiers positifs et M est un entier positif égal à deux ou plus, **caractérisé en ce qu'il** comprend :

5 une pluralité de modules de décodage chacun étant associé à deux canaux d'entrée adjacents dans l'espace ou plus (1', 3', 5', 9', 13'), dans lequel chaque canal d'entrée est partagé entre de multiples modules, et chaque module :

10 - comporte une matrice qui génère, à partir des deux canaux d'entrée associés ou plus (1', 3', 5', 9', 13'), un canal de sortie ou plus (1-23) chacun constituant un sous-ensemble desdits N canaux, par un procédé qui comporte la détermination d'une mesure de la corrélation entre les deux canaux d'entrée ou plus (1', 3', 5', 9', 13') et le niveau d'interrelation entre les deux canaux d'entrée ou plus (1', 3', 5', 9', 13'), ou
 15 - génère à partir des deux canaux d'entrée associés ou plus (1', 3', 5', 9', 13'), par un procédé qui comporte la détermination d'une mesure de la corrélation entre les deux canaux d'entrée ou plus (1', 3', 5', 9', 13') et le niveau d'interrelation entre les deux canaux d'entrée ou plus (1', 3', 5', 9', 13') des signaux de commande qui sont utilisés, avec des signaux de commande générés par d'autres modules de décodeur, afin de modifier les coefficients d'une matrice variable pour générer tous les canaux de sortie (1-23), ou
 20 - génère, à partir des deux canaux d'entrée ou plus (1', 3', 5', 9', 13') par un procédé qui comporte la détermination d'une mesure de la corrélation entre les deux canaux d'entrée ou plus (1', 3', 5', 9', 13') et le niveau d'interrelation entre les deux canaux d'entrée ou plus des signaux de commande qui sont utilisés, avec des signaux de commande générés par d'autres modules décodeurs pour modifier les facteurs d'échelle des entrées et/ou des sorties d'une matrice fixe pour générer tous les canaux de sortie (1-23).

2. Procédé selon la revendication 1, dans lequel les modules sont classés par ordre hiérarchique en fonction de leur nombre de canaux d'entrée et un superviseur communique avec les modules pour commander le partage des signaux d'entrée en fonction de leur ordre hiérarchique.

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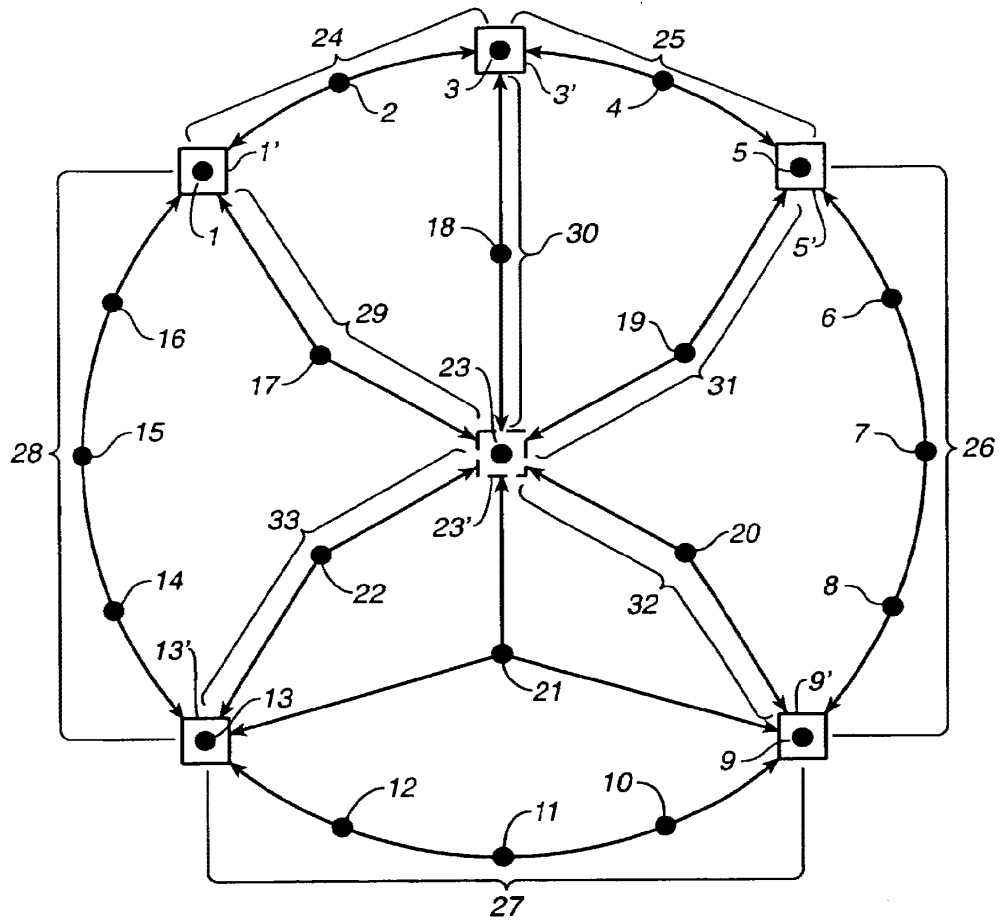


FIG. 1

REFERENCES CITED IN THE DESCRIPTION

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(54) **Communication protocol for controlling transfer of temporal data over a bus between devices in synchronization with a periodic reference signal**

Kommunikationsprotokoll für die Übertragungssteuerung von zeitlichen Daten über einen Bus zwischen Geräten in Synchronisation mit einem periodischen Referenzsignal

Protocole de communication pour contrôler le transfert de données temporelles sur un bus entre dispositifs en synchronisme avec un signal de référence périodique

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(73) Proprietor: **AVID TECHNOLOGY, INC.**
Tewksbury, MA 01876 (US)

(72) Inventors:
• **Wallace, Ron**
Lexington, Massachusetts 02420 (US)

- **Der, Harry**
Westford, Massachusetts 01886 (US)
- **Corbett, Martin**
Westford, Massachusetts 01886 (US)
- **Fetters, Terrence**
Tewksbury, Massachusetts 01876 (US)

(74) Representative: **Kazi, Ilya et al**
Mathys & Squire
120 Holborn
London EC1N 2SQ (GB)

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Description

5 **[0001]** Peripheral devices that process video and audio data commonly are connected to a host computer either to perform functions not performed by the host computer or to convert the video and audio data between digital and analog formats. The connection between the peripheral device and the host computer is typically provided by a bus over which communication is performed using packets. Example types of a bus include, a parallel bus, such as a peripheral component interconnect (PCI) bus, a serial bus, such as an IEEE-1394 compliant bus, or a computer network, such as Ethernet.

10 **[0002]** Video and audio data are typically in a standard format, such as the format described by "Specifications of Consumer-Use Digital VCR's using 6.3mm magnetic tape" of the HD Digital VCR Conference dated December 1994, also called the "Blue Book," or by SMPTE 314M-1999 or by IEC-61834, which describe video and audio data in a format that is commonly called "DV." Video data may be compressed or uncompressed. Audio data typically is uncompressed.

15 **[0003]** An example peripheral device that processes video and audio data is a transcoder. A transcoder typically receives compressed digital video data and audio data, such as DV, over a serial bus, such as an IEEE-1394 compliant bus, and processes the data for output. In particular, the video data is converted into an analog video signal for output to a video device, such as a video monitor or camcorder. The audio data is converted into an audio signal for output to an audio device, such as a set of speakers. Such a transcoder also may receive an input analog video signal and audio signal and generate compressed digital video data and audio data, such as DV, that is transferred to the host computer over a serial bus, such as an IEEE-1394 compliant bus.

20 **[0004]** Many systems currently are available that transfer video and audio data between a peripheral device, such as a camcorder or a transcoder, and a host computer. The host computer may be used for any of a number of purposes, such as video and audio editing. When using DV, DV commonly is transferred between the host computer and a peripheral device using an IEEE-1394 compliant bus. A standard is defined for transporting DV streams over an IEEE-1394 compliant bus, called IEC-61883. In an editing system that edits stored DV and outputs DV over an IEEE-1394 bus using IEC-25 61883, the host first decompresses the DV, performs editing operations on decompressed data, and then compresses the results back into DV before transferring DV over the IEEE-1394 bus. This process requires resources of the host computer to be used for compression, thus limiting the real-time processing capabilities of the host computer. Thus, a DV version of the entire program often is created and stored in a data file before it is transferred from the computer to another device. Further, to produce video in any output format other than DV, such as an analog video signal, the DV version also must be decompressed first. Decompression of DV and encoding of the results in an analog video signal generally is performed by a transcoder.

30 **[0005]** An IEEE-1394 compliant device transfers data based on a local time base using what is called a bus timer clock. As a result, each device attached to an IEEE-1394 bus has its own time base, which generally is faster or slower than the time base on other devices on the same bus. Although the IEC-61883 standard provides a technique for synchronizing these clocks, different time bases generally exist on different devices on the bus due to jitter and phase differences between the clocks. A consequence of the different time bases is that a host either transmits more data packets than a transcoder can consume or transmits too few data packets to the transcoder, causing a frame to be repeated or dropped. To address this problem, a host is required to periodically send empty data packets and to use a presentation time stamp (the SYT field) in a packet of each frame. The presentation time stamp designates to the transcoder a point in time when the frame should be displayed. However, the time stamp is generated by a sender using the sender's bus timer clock, but is used by the recipient using the recipient's bus timer clock. Thus, even if the number and frequency of the transmission of the empty data packets is calculated carefully in advance of transmission, on average, the empty data packets merely extend the period of time before a frame is dropped or repeated. The use of empty data packets does not solve the problem created by the different time bases on sending and receiving devices.45 Another problem with time stamps is that not all commercially available devices support use of time stamps. Some devices do not send timestamps, and some devices do not attempt to process received timestamps. Because frames may be dropped or repeated using these techniques, such transcoders cannot be used to reliably generate a synchronous video signal for playback on a video monitor or for recording to tape, or to synchronize playback of audio and video material with an external synchronized video device.

50 **[0006]** Another method is known from US-A-6418150 which calculates the frequency offset between a receiving and a source device and adjust the frame packets rate accordingly.

[0007] Current transcoder designs also do not enable a host to connect to another device through the transcoder in a manner that allows the host to communicate directly to the other device, yet allow the transcoder to modify data in packets sent to the other device.

55 **[0008]** A communication protocol is provided to allow a device that receives a periodic reference signal to control the flow of temporal data between devices over a bus to maintain synchronization of the temporal data with the periodic reference signal. Flow control messages are sent between devices, allowing a device that receives the periodic reference signal to control how and when the data is sent over the bus. For example, a peripheral device that receives a periodic

reference signal may send flow control messages to a host computer to control the flow of data from the host computer to the peripheral device. Different types of temporal data, such as audio and video data, may be sent as separate data streams. Using the IEEE-1394 protocol, the flow control messages are sent using an asynchronous protocol, whereas temporal data are sent using an isochronous protocol.

[0009] Uncompressed video and audio data are transferred from the host computer over a serial bus to the peripheral device, where the peripheral device performs operations on the data, such as compressing the video data or generating an analog video signal. Video and audio data may be transferred as separate streams. By transferring uncompressed video data, the host computer does not need to consume resources to compress the video data, thus allowing those resources to be used for more creative operations on the video and audio data. Further, by transferring uncompressed video data, the peripheral device can generate video data in many output formats without first decompressing the video data. The peripheral device also may output video and audio data using a standard protocol over a standard bus. For example, the peripheral device may generate DV from the uncompressed audio and video data, and output the DV using IEC-61883 over an IEEE-1394 compliant bus.

[0010] A peripheral device that synchronizes transfer of data from the host computer with a periodic reference signal also can be used to synchronize data transfer to another device, such as a transcoder or camcorder or deck, that is connected to the peripheral device. For example, the peripheral device may encode the uncompressed audio and video data received from the host computer as a video signal. The peripheral device then outputs the video signal to the other device. The other device may provide the periodic reference signal. For example, if the other device is a camera, a composite video output signal from the camera can be used as the periodic reference signal. As a result, transfer of data from the host to the peripheral device and then to the camera is synchronized to the camera.

[0011] Such a peripheral device also allows a real time digital cut to tape to be performed. In particular, by transferring uncompressed data from the host computer to the peripheral device during playback in synchronization with a periodic reference signal, the host computer can process effects in real time. The peripheral device may generate a DV output from the uncompressed data. This DV output can be provided over an IEEE-1394 compliant bus to a recording device such as a DV deck or camcorder. If this DV deck or camcorder has a composite video output signal, then this output signal from the deck or camcorder may be used as the periodic reference signal. The peripheral device also may act as a bridge between the host computer and the other device. In particular, the peripheral device may implement a protocol over the serial bus between the host computer and the peripheral device and a protocol over another serial bus between the peripheral device and the other device. After receiving a packet at one port, the peripheral device determines if the packet is directed to a device connected on the other port or if the packet is directed to the peripheral device or if the packet is to be processed by the peripheral device to create one or more new packets to be output to the device connected on the other port. For example, the peripheral device may decompress compressed video data received from the host and output data packets that include uncompressed video data to the other device. From the perspective of the host, the peripheral device behaves as if it were the other device. From the perspective of other device, the data packets appear to be from a host device.

Fig. 1 is a block diagram of a system in which a host computer is connected to a peripheral device.

Fig. 2 is a block diagram of a system in which a host computer is connected to a peripheral device that is synchronized with another video device.

Fig. 3 is a more detailed block diagram of a mode of operation of the peripheral device as in Fig. 1.

Fig. 4 is a more detailed block diagram of another mode of operation of the peripheral device as in Figs. 1 and 2.

Fig. 5 is a more detailed block diagram of another mode of operation of the peripheral device as in Figs. 1 and 2.

Fig. 6 is a block diagram of a mode of operation using two peripheral devices that both receive data from a host computer.

Fig. 7 is a block diagram of a peripheral device for the mode of operation shown in Fig. 3.

Fig. 8 is a block diagram of a peripheral device for the mode of operation shown in Fig. 4.

Fig. 9 is a block diagram of a peripheral device for the mode of operation shown in Fig. 5.

Figs. 10A-10B are a more detailed block diagram of a peripheral device incorporating the modes of operation illustrated in Figs. 7-9.

Fig. 11 illustrates an example synchronization packet format.

Fig. 12 illustrates an example preamble packet format.

Fig. 13 illustrates an example stream descriptor format.

Fig. 14 is a table of stream identifiers.

Fig. 15 illustrates an example data packet format.

Fig. 16 illustrates an example null packet format.

[0012] Fig. 1 illustrates a system in which a host computer 100 is connected to a peripheral device 102 by a bus 104, such as an IEEE-1394 compliant bus. The host computer typically includes storage for storing video data and audio

data in data files using a file system of the operating system of the computer. An editing system is provided on the host computer to allow editing of a program using the video data and audio data with effects to be performed on the video data. Such an editing system typically represents a program as a sequence of clips, wherein a clip is a reference to a data file containing temporal media, and a range within the data file. A playback system also is provided for playing back the edited program in full motion at full resolution. The playback system allows an editor to view the program that has been edited. If the video data is compressed, the playback system decompresses the video data, and performs any effects on the decompressed video data.

[0013] The bus may be any type of bus over which communication is performed using packets and which can transmit data for temporal data, such as video data and audio data, at a rate faster than the sample rate of the temporal data, such as the frame rate of the video data. The host computer includes a bus interface for sending packets to and receiving packets from the device over the bus. A buffer in memory commonly is connected to transfer data to the bus interface for transmission as a plurality of packets. This buffer will receive, for example, data from the playback application for transfer to the device. A driver, a form of software that acts as a controller, for the bus interface also is on the host computer and allows the host computer to implement communication protocols, such as the protocol described herein, over the bus.

[0014] The host computer may receive data over the bus from the peripheral device. The host computer may send data over the bus to the peripheral device. The host computer and peripheral device may exchange several kinds of data including video data, audio data and other temporal data and other content. Video data may be either compressed or uncompressed, and may be combined with or separate from audio data and other related time-based media. Audio data typically is uncompressed, but may be compressed. The invention is not limited to any particular format of video data or audio data or other temporal data.

[0015] Data is transferred over the bus 104 in the form of packets. A packet is a unit of transmission of data that includes both data and routing information, such as an address of a node on the bus. A packet also may indicate a type of the packet if the communication protocol allows different types of packets to be transmitted. A packet also may include error correction codes and other data. The form and content of each type of packet typically is specified as part of a communication protocol that is described in more detail below.

[0016] The peripheral device may have several inputs 106 to receive data in any of a number of formats from a source, such as video and audio data from a tape deck. The peripheral device also may have several outputs 108 to send data in any of a number of formats to a receiver, such as video and audio data to a monitor and speakers or to a tape deck. For example, the peripheral device may send DV to a DV device, such as a DV camcorder or DV tape deck, or may receive DV from a DV device. Examples of other video and audio input and output formats include, but are not limited to, analog formats such as composite video, component video (such as YCrCb and YUV video) and S-Video, and digital formats, both compressed and uncompressed, both standard and proprietary, such as MPEG-2, SMPTE-125M, SMPTE-260M, SMPTE-264M, SMPTE-265M, SMPTE-267M, SMPTE-274M, and SMPTE-279M, among others. The peripheral device also may internally generate or receive from an external source a periodic reference signal 110 for use in controlling data transfer from the host. This periodic reference signal has a rate that corresponds to the desired rate of the temporal data, such as a field or frame rate of video, that is output by the peripheral device. The periodic reference signal may be generated by an accurate crystal clock generation circuit. Such a circuit may be internal to the peripheral device or external to the peripheral device. The periodic reference signal also may be generated by using a video decoder that decodes a received composite video signal to generate a vertical synchronization signal (VSYNC), a horizontal synchronization signal (HSYNC), a pixel clock (such as 27 MHz for NTSC/PAL standard definition video) and a burst phase signal. Such a composite video signal used for this purpose is commonly called a genlock, RS170, black burst or house sync signal, and may or may not include active video. The signals output from the decoder may be provided as inputs to a video encoder to generate output video that is in synchronization with the received composite video signal.

[0017] Each field boundary or frame boundary, or other designated boundary in time that can be derived from the periodic reference signal, may be used to define a data interval. A data interval is a period of time within the temporal data that corresponds to the periodic reference signal, such as a frame or field of video data, or audio or metadata that corresponds to the frame or field of video data. The communication protocol described below utilizes a data interval as a basis for transfer of data from the host to the device to synchronize the data with the periodic reference signal.

[0018] The peripheral device may be used to capture or playback multiple streams of audio and video data for the host computer. For capture, the peripheral device receives data from one or more of its inputs and transfers the received data to the host computer over the bus. For example, the peripheral device may receive a DV stream, which it transfers directly to the host computer. The peripheral device may receive an analog video and audio stream, which it then converts to a DV stream that is transferred to the host. In the playback mode, the peripheral device receives a stream of video and audio data from the host computer over the bus. This stream may be converted into any of a number of formats for output to another device. For example, the peripheral device may convert the received stream into an analog video and audio signal or into a DV stream for transmission to a recording device or to a monitor and speakers.

[0019] Referring to Fig. 2, the peripheral device 202 can synchronize data transfer over the bus 204, from the host

computer 200 to another device 212 that is connected (via inputs 206 or outputs 208) to the peripheral device 202, to a periodic reference signal 210 that is received by both the peripheral device 202 and the other device 212. The periodic reference signal may be generated by the peripheral device 202, the other device 212, or may come from another source.

[0020] Examples of other devices 212 that may be connected to the peripheral device 202 include but are not limited to a transcoder, video display, tape deck, camera or video processing equipment. For example, the peripheral device may receive uncompressed video and audio data from the host computer. The peripheral device may generate from this data an analog video and audio signal that is synchronized with the periodic reference signal and that is output to the other device.

[0021] The various functions of the peripheral device also may be integrated within the other device to produce a single device that operates to provide the combined functions. Similarly, the various functions of the other device may be integrated within the peripheral device to produce a single device that operates to provide the combined functions. Thus, a transcoder, video display, tape deck, camera or other video processing equipment could incorporate the functions of the peripheral device as described herein. Three example operational modes for such a peripheral device include uncompressed playback mode, digital cut mode, and capture mode, as will be described in connection with Figs. 3-5.

[0022] In uncompressed playback mode, shown in Fig. 3, the host system 300 passes a stream of uncompressed video and audio and/or device control commands to the peripheral device 302 over the bus 304. For playback during editing, the uncompressed stream may be fractional frame (such as 1/4-frame) or full-frame, depending on a user-defined setting. For digital cut mode discussed below, full-frame mode is the default setting. The peripheral device may include a resizer to scale fractional-frame video to full-frame video. The peripheral device may generate an analog video and audio output 306 using the uncompressed stream. A video window on the desktop of the host computer may play in synchronization with the outputs of the peripheral device. The peripheral device may create a DV or other compressed output 308 by encoding the uncompressed stream using an encoder on the peripheral device. By providing sufficient buffering, the compressed output may be kept in synchronization in spite of latencies incurred by encoding the uncompressed stream on the peripheral device.

[0023] In this mode, editing, playback and creation of output video and audio with real-time effects can be provided. In particular, because the host does not compress the video and audio stream after application of a real-time effect, the uncompressed output is transferred in real time during playback to the peripheral device, which in turn can encode the uncompressed stream for output. Such operation also allows a real time digital cut to tape to be performed. In particular, by transferring uncompressed data from the host computer to the peripheral device during playback in synchronization with a periodic reference signal, the host computer can process effects in real time. The peripheral device may generate a DV output from the uncompressed data. This DV output can be provided over an IEEE-1394 compliant bus to a recording device such as a DV deck or camcorder. If this DV deck or camcorder has a composite video output signal, then this output signal from the deck or camcorder may be used as the periodic reference signal.

[0024] Playback of video may occur in one or more of a desktop window on the host computer, composite or S-Video outputs 306 on the peripheral device, and a DV output 308 via a IEEE-1394 compliant bus on the peripheral device, or on other outputs of the peripheral device. Playback of audio may occur through one or more of the analog outputs, DV outputs or digital outputs (not shown).

[0025] In another mode shown in Figure 4, a digital cut of a compressed stream rendered by the host computer 400 and/or device control commands is provided to the peripheral device 402 over the bus 404. In this mode, any effects are rendered and the rendered images are compressed at the host computer and stored in the data files before the transfer. The compressed stream is read by the host computer 400 from its data files and transmitted to peripheral device 402. The compressed stream may be transmitted to the peripheral device using the synchronized protocol described below. The synchronized protocol is used to transfer the DV stream for playback on a synchronized device such as a television monitor.

[0026] Playback of video may occur in one or more of a desktop window on the host computer, composite or S-Video outputs 406 on the peripheral device, and a DV output 408 via the IEEE-1394 compliant bus on the peripheral device, or on other outputs of the peripheral device. Playback of audio may occur through one or more of the analog outputs, DV outputs or digital outputs (not shown). The analog outputs are created by having the peripheral device decode the incoming compressed stream using a decoder on the peripheral device, then encoding the decoded data using a video encoder. To perform a digital cut to a DV tape, the peripheral device performs device control operations on the DV device (not shown) attached to the DV output 408, and outputs the DV data directly as received from the host.

[0027] In this mode the compressed output from the peripheral device plays in synchronization with the video in the desktop window on the host computer. By providing sufficient buffering in the host, the latencies arising from decoding the DV stream and encoding the analog outputs on the peripheral device may be overcome to synchronize the analog outputs with the video in the desktop window.

[0028] A third mode of operation is capture, shown in Figure 5. Capture employs one of several different input sources (506, 508, 510) of the peripheral device 502, such as composite video, S-Video video, and DV video and audio inputs. DV device control by the peripheral device 502 is available if a DV input device is used.

[0029] If the input device is DV, then the DV stream is passed through the peripheral device directly to the host 500 over the bus 504. The DV stream may be transferred to the host system using the synchronized protocol described below. During transfer of DV data to the host computer, the peripheral device may decode the DV stream and encode the decoded data to generate an analog video and audio outputs (not shown).

[0030] If the input signal is composite or S-Video, or some other format, then the input signal is converted into a digital format, such as DV, before being transmitted to the host over the IEEE-1394 compliant bus. In this mode the analog video and audio signals also may be output through the corresponding analog video and audio outputs (not shown). Output of DV format video over the other IEEE-1394 compliant bus output 508 to an external device also may be active or inactive.

[0031] In all of these modes, the peripheral device may act as a bridge between the host computer and the other device. In other words, the peripheral device implements a protocol over the first serial bus for communication between the host computer and the peripheral device. The peripheral device implements the same protocol over the second serial bus for communication between the peripheral device and the other device. The peripheral device communicates information from the other device to the host computer to allow the host computer to communicate packets directed to the other device using the protocol over the first serial bus. The peripheral device also communicates information from the host computer to the other device to allow the other device to communicate packets directed to the host computer using the protocol over the second serial bus.

[0032] In general, the peripheral device receives messages from a host at one port of the peripheral device, translates the message, then send a message to the other device from another port of the peripheral device. The translation involves at least translating any node address in the message so that it is properly directed to the other device through the other port. Translation may also involve other processing to be performed on the data in the packet such as conversion from one data format to another.

[0033] Conversely, the peripheral device receives messages from the other device at one port of the peripheral device, translates the message, then sends a message to the host from another port of the peripheral device. Again, the translation involves at least translating any node address in the message so that it is properly directed to the host through the other port, and may involve converting data formats or other processing of the data in the message.

[0034] To implement such functionality, a separate process for managing communication on each port is executed by a microprocessor on the peripheral device. When a message is received on one port, the process for that port invokes a translator process which translate the message and sends the translated message to the other process for the other port.

[0035] Whether packets flow from the host to the other device or from the other device to the host, packets received by the host, peripheral device or other device are locally acknowledged using a low level protocol, such as the link layer protocol. However, application level acknowledgements are passed from the host to the other device, and from the other device to the host, through the peripheral device using the node translation described above. Thus, the peripheral device appears to be a virtual host from the perspective of the other device, and appears to be a virtual device from the perspective of the host.

[0036] This bridge functionality will now be described in more detail for four types of communication: from host to peripheral device, from peripheral device to other device, from other device to peripheral device and from peripheral device to host.

[0037] If a packet is received by the peripheral device from the host computer on one port, that port locally acknowledges receipt of the packet using, for example, a link layer or other low level protocol. The packet is processed to determine if the packet is directed to the peripheral device or the other device. If the packet is directed to the peripheral device, the packet is processed at the peripheral device. If the packet is directed to the other device, the packet is first processed by the peripheral device to create one or more new packets that are in turn sent to the other device.

[0038] If a packet is sent to the other device by the peripheral device by one port, the other device locally acknowledges receipt of the packet by using a link layer or other low level protocol message.

[0039] If a packet is received by the peripheral device from the other device by one port, that port locally acknowledges receipt of the packet, for example using a link layer or other low level protocol. The packet is processed to determine if the packet is directed to the host computer or to the peripheral device. If the packet is directed to the peripheral device, the packet is processed at the peripheral device. If the packet is directed to the host computer, the packet is processed by the peripheral device to create one or more new packets that are in turn transferred to the host computer.

[0040] If a packet is sent to the host by the peripheral device by one port, the host locally acknowledges receipt of the packet by using a link layer or other low level protocol message.

[0041] The local acknowledgement of packets using the low level protocol is different from acknowledgement of application level messages by the other device or by the host. If the other device is to respond to a message received from the host through the peripheral device, this response is made using an application level protocol, such as AV/C, and is handled as a message from the other device that is directed to the host through the peripheral device. Similarly, if the host is to respond to a message received from the other device through the peripheral device, this response is made using an application level protocol, such as AV/C, and is handled as a message from the host that is directed to

the other device through the peripheral device.

[0042] Two peripheral devices also may be connected to receive data from the same host, as illustrated in Fig. 6. In particular, a first peripheral device 600 and a second peripheral device 610 both receive a periodic reference signal 620. The first peripheral device 600 communicates with a host 630 over a bus 640. The second peripheral device 610 is another node on the same bus 640. Using the command protocol described below the host computer selects one of the peripheral devices and directs it not to send flow control messages. The host computer then replies only to flow control messages from the other peripheral device. In this manner, both of the peripheral devices can provide synchronized outputs from the same data stream sent by the host computer.

[0043] Example hardware that supports each of the modes of operation described above will now be described in connection with Figs. 7-10A-B.

[0044] In uncompressed playback mode, shown in Figure 7, the peripheral device operates as a frame buffer and DV compression device. Uncompressed audio and video data is sent by the bus interface, shown as a IEEE-1394 transceiver 706, on the host 700 over the IEEE-1394 compliant bus 704 to the peripheral device 702. The bus interface, shown as a IEEE-1394 transceiver 708, on the peripheral device 702 receives the data, and the data is stored in the media hub 710, which includes a frame buffer. The video data is then reformatted as an ITU/CCIR 656-video stream by a CCIR 656 formatter 712, which provides the reformatted data to both the DV encoder 714 as well as a video encoder 716. An I2S formatter 718 reformats the audio data into an I2S audio stream for transport to both an audio encoder 720 and the DV encoder 714. The DV stream output by DV encoder 714 may be provided to an IEEE-1394 formatter 722 to produce IEEE-1394 packets that may be output over another IEEE-1394 compliant bus 726 by another IEEE-1394 transceiver 724.

[0045] To prevent dropping or adding video frames in the video provided to the CCIR 656 formatter 712 during playback of video over the IEEE-1394 compliant bus 704, the peripheral device controls the flow of data from the host computer 700 to the peripheral device 702. This flow control also allows the peripheral device to generate and output a synchronous video signal using the analog encoder 716 that is synchronized with an internal or external periodic reference signal. A communication protocol to implement this flow control is described in more detail below.

[0046] In digital capture mode, shown in Fig. 8, the peripheral device 802 operates as both a buffer and a decoder. Digital video and audio data, such as DV, is received from a device (not shown) and is passed to the host computer 800 for processing and storage on a storage device accessed by the host computer, such as a disk system. The received data, if compressed, also may be decompressed to produce output audio and video for monitoring. More particularly, in Fig. 8, data, such as DV, is received through an IEEE-1394 interface 806, is buffered and sent by another IEEE-1394 interface 808 over the bus 804 to the IEEE-1394 interface 810 at the host 800.

[0047] The received data also may be deformatted by IEEE-1394 deformatter 812 and input to a decoder 814, such as a DV decoder, to be decompressed. The decompressed data then can be encoded for output to other devices. For example, the decompressed data may be input to an analog video encoder 816 and the audio data may be input to an audio encoder 818. The outputs of the encoders can be applied to a monitor and speakers for monitoring of the capture process. Because the timing on the analog video encoder 816 is derived from a local time base and is asynchronous to the incoming DV data, the analog encoder 816 may either repeat a frame of video or drop a frame of video depending on whether the incoming DV data stream is running slower or faster than the video encoder.

[0048] In analog capture mode, shown in Fig. 9, the peripheral device 902 converts input analog video and associated audio into digital format, and transfers the digital data to the host 900 over the bus 904. For example, an analog video decoder 906 converts the input analog video into a digital ITU 656-video data stream. The digital video data is sent to an encoder 908, such as a DV encoder, for compression. Received audio is connected to an audio encoder/decoder 910 for conversion to an I2S digital audio format. The audio data also may be provided to encoder 908 for compression. The audio encoder/decoder 910 also may provide audio outputs for monitoring the audio during capture. A video encoder 912 also may provide a video output for monitoring the video during capture. Encoded video data from encoder 908 is formatted by an IEEE-1394 formatter 914, which prepares it for transmission by the bus interface, shown as IEEE-1394 transceiver 916. The host receives the data over the bus 904 through the bus interface, shown as IEEE-1394 transceiver 918.

[0049] A block diagram of a peripheral device that has these modes of operation is shown in Figs. 10A-10B. In Figs. 10A-10B, the peripheral device 1000 operates in two basic modes: capture and playback. In capture mode, digital video and audio data, such as DV, may be received by an IEEE-1394 compliant interface 1002 and sent to the host over an IEEE-1394 interface 1004 to a corresponding interface 1051 on the host. Analog audio and video also may be received and converted to a digital format and then sent to the host over the interface 1004. In playback mode video and audio data is received from the host through interface 1004 and is output as analog video and audio and/or is output as digital video and audio data through the IEEE-1394 compliant interface 1002. The IEEE-1394 compliant interface 1002 provides an interface through port 1045 to external devices, such as cameras and decks, whereas the IEEE-1394 compliant interface 1004 provides an interface to the host computer. The interfaces 1002 and 1004 can be implemented, for example, using a physical layer device and a link layer device. A Texas Instruments TSB12LV32 Link-Layer Controller and a TSB41LV01 Cable Transceiver/Arbiter can be used to achieve a 400 megabit per second nominal transmission

rate over the IEEE-1394 bus.

[0050] Left and right audio inputs and outputs 1040, 1041, 1042, 1043 for audio processing provide inputs to an audio encoder/decoder 1006, which may be implemented, for example, using a Philips UDA1345TS, which supports 32, 44.1 and 48kHz rates. Other chips supporting other rates can be used. To keep the audio locked to the video during capture, the audio clock is derived from the input video decoder 1008. For playback or output the audio clock is derived from the analog video encoder 1010.

[0051] A headphone audio output 1044 may be provided through an encoder 1012, which may be implemented, for example, using a Philips UDA1320ATS audio digital to audio converter (DAC), which supports 32, 44.1 and 48kHz rates. Other chips supporting other rates can be used. A separate DAC may be used for the headphone to allow adjustment to the headphone volume without affecting the main audio outputs.

[0052] The peripheral device 1000 may receive analog video through inputs 1046 and 1047, such as composite and S-video inputs. These inputs are received by a decoder 1008 which converts the input signal into digital video data. The decoder 1008 may be implemented, for example, by a Philips SAA7114H video decoder, which supports NTSC, NTSC-j and PAL analog video formats. Other decoders supporting the same or other formats may be used. If connected to either a composite or S-Video source, this decoder converts the incoming analog signal into a standard ITU 656 YUV 4:2:2 digital video data stream. This decoder also provides support for luminance offset control, luminance contrast control, chrominance saturation control and chrominance hue control.

[0053] The peripheral device 1000 also may output analog video through outputs 1048 and 1049, such as composite and S-video outputs provided by an encoder 1010. The encoder 1010 may be implemented, for example, by an Analog Devices ADV7176 encoder. The encoder 1010 accepts a YUV 4:2:2 digital video stream and real time control stream information from the genlock circuit 1018, buffered through a media hub 1028, to generate synchronized analog composite and S-Video outputs. The ADV7176 device encodes analog video and provides several functions for processing the output video, such as luminance filters (Low-Pass/Notch/Extended) control, subcarrier frequency and phase control and luminance delay control.

[0054] The peripheral device 1000 also may include a compression/decompression device (codec) 1014, such as a DV codec. This codec may be implemented, for example, using a Divio NW701 DV25 codec for both decompressing and compressing a DV data stream. In compression, the codec 1014 receives an ITU 656 YUV 4:2:2 digital video stream and an I2S audio stream and compresses it to produce a DV compressed video stream. In decompression, the codec 1014 receives a DV compressed video stream and produces both an ITU 656 YUV 4:2:2 digital video stream and an I2S audio stream. The NW701 codec also provides NTSC (4:1:1), PAL (4:2:0) and PAL (4:1:1) processing, audio support for 48, 44.1 and 32 kHz (12 and 16 bit), and optional low data-rate modes, including 3.0, 2.4, 1.8, 1.5 and 1.0 Mbytes/sec. The codec 1014 uses a memory 1016 for storing data temporarily during processing. Such a memory 1016 may be implemented, for example, using a 256Kx32 DRAM.

[0055] The peripheral device 1000 also may receive a periodic reference signal (genlock) through an input 1050. This signal is received by a video genlock device 1018, which may be implemented, for example, using a Philips SAA7113 decoder. This decoder supports NTSC, NTSC-j and PAL analog video formats. If connected to a composite genlock source it extracts the vertical synchronization, horizontal synchronization and color burst phase information and provides this information to the media hub 1028, which can be used to lock the output timing of the ADV7176 video encoder 1010 to the genlock source.

[0056] Controls may be provided on the peripheral device to allow an individual to adjust the various controls of the input decoders and output encoders, particularly to adjust levels of the analog video signals, for audio muting and for adjusting gain levels of the audio signals.

[0057] A frame buffer controller, resizer and data router, herein called a "media hub" 1028, and associated buffer 1030 provide four basic functions on the peripheral device 1000. First, they buffer both uncompressed and compressed data streams between the inputs and the outputs. Second, they provide a switching or multiplexing function that routes data among the inputs, outputs, encoders and decoders. Third, they de-format incoming IEEE-1394 compliant data streams and ITU-656 compliant video streams. Fourth, they format out going IEEE-1394 compliant data streams and ITU-656 compliant video streams. The media hub 1028 may be implemented, for example, using a field programmable gate array (FPGA) with a local memory, such as an SDRAM as its associated buffer 1030.

[0058] The functions of the media hub 1028 are performed in connection with a microcontroller subsystem 1020. The primary function of the microcontroller subsystem is to provide an abstraction layer for software commands received over the host IEEE-1394 compliant interface 1004 and for controlling the IEEE-1394 compliant interface 1002. The microcontroller subsystem also provides support for setting up and controlling the video and audio encoders and decoders 1006, 1008, 1010, 1012 and 1018. The microcontroller subsystem 1020 includes a microprocessor 1022, which may be implemented, for example, using a Motorola MPC5307 microprocessor. The microprocessor 1022 operates in accordance with instructions stored in the flash memory 1024 and static random access memory 1026. In particular, an actual operating system with a real-time kernel can be provided for the microcontroller, such as the NUCLEUS operating system from Accelerated Technology, Inc., the embedded systems division of Mentor Graphics and located in Mobile,

Alabama.

[0059] Operation of the microcontroller will now be described in more detail. The microcontroller receives commands from the host through a command protocol described in more detail below. These commands allow the host to set the mode of operation of the peripheral device. These modes include settings for the expected audio and video format for input and output. Example format mode settings include PAL and NTSC for video and 48KHz, 44.1KHz or other rate for audio. For these modes, the host may designate a format to the microcontroller. The microcontroller in turn sets various registers of the input and output devices to handle the desired format.

[0060] Other modes indicate whether the host is receiving data from or providing data to the peripheral device. Some example modes include, but are not limited to:

- DV input to peripheral device and DV output to host
- DV input to peripheral device and uncompressed output to host
- Composite input to peripheral device and DV output to host
- Composite input to peripheral device and uncompressed output to host
- Uncompressed input from host and composite output
- Uncompressed input from host and DV output
- DV input from host and composite output
- DV input from host and DV output.

[0061] In a mode in which DV is input to the peripheral device and DV is output to the host computer, packets are received through interface 1002 and stored in the memory 1030 by media hub 1028. When a packet becomes available in the memory 1030, the media hub 1028 notifies the interface 1004. Interface 1004 then transfers the packet to the host. In this mode the device acts as a bridge.

[0062] In a mode in which DV is input to the peripheral device and uncompressed data is output to the host computer, packets are received through interface 1002 and stored in the memory 1030 by media hub 1028. Packets are allowed to accumulate in the memory 1030 until a frame of image data is received. The media hub 1028 then directs the DV codec 1014 to read the frame of data from memory 1030 into its memory 1016. The DV codec then generates an uncompressed frame. The uncompressed frame is placed in another portion of the memory 1030. When an uncompressed frame is available in the memory 1030, the media hub 1028 notifies the interface 1004. The interface 1004 then transfers the image as multiple packets to the host. In this mode the device acts as a bridge.

[0063] In a mode in which a composite input is received by the peripheral device and DV is output to the host computer, the decoder 1008 produces digital video information which is stored in the memory 1030 by the media hub 1028. When a frame of image data has been received, the media hub 1028 directs the DV encoder 1014 to read the frame of data from memory 1030 into its memory 1016. The DV encoder then generates DV and places the data in another portion of the memory 1030. The media hub 1028 notifies the interface 1004 that data is available. The interface 1004 then transfers the DV image as multiple packets to the host.

[0064] In a mode in which a composite input is received by the peripheral device and uncompressed data is output to the host computer, the decoder 1008 produces digital video information which is stored in the memory 1030 by the media hub 1028. As image data is received, the media hub 1028 directs the interface 1004 that data is available for transfer. The interface 1004 then transfers packets of uncompressed data to the host.

[0065] In a mode in which the peripheral device receives uncompressed data from the host and outputs a composite analog video signal, the media hub 1028 directs packets received through interface 1004 to be stored in the buffer 1030. As data becomes available in memory 1030, the media hub 1028 directs data to the encoder 1010 for output.

[0066] In a mode in which the peripheral device receives uncompressed data from the host and outputs DV to another device, the media hub 1028 directs packets received through interface 1004 to be stored in the buffer 1030. As a frame of uncompressed data becomes available in the memory 1030, the media hub 1028 notifies the DV codec 1014 which reads the data into its buffer 1016. The DV codec then generates DV which is stored into the memory 1030. The media hub 1028 then notifies the interface 1002 that DV data is available. The interface 1002 reads the data and transfers it as packets through the output 1045. In this mode the device acts as a bridge.

[0067] In a mode in which the peripheral device receives DV input from the host and provides a composite analog video output signal, the interface 1004 receives packets of DV data that are stored in the memory 1030. When a frame of data is available, the media hub 1028 notifies the DV codec 1014, which then reads the data into its memory 1016. The DV codec then produces uncompressed video data which is stored in memory 1030. The media hub 1028 then provides the uncompressed video data to the analog encoder 1010 for output.

[0068] In a mode in which the peripheral device receives DV input from the host and outputs DV to another device, the interface 1004 receives packets of DV data that are stored in memory 1030 by media hub 1028. The media hub 1028 notifies the interface 1002 when packets are available for output. The interface 1002 then reads the packets from the memory 1030. In this mode the device acts as a bridge.

[0069] During each of the modes described above, corresponding audio input and output may be provided. Also, a composite or S-video output may be provided by the peripheral device to allow monitoring by an external monitor. Each device 1002, 1014 and 1004 may have a direct memory access (DMA) engine to access the memory 1030.

[0070] In some modes, the uncompressed video data may be received or output in 1/4 frame size. In these modes, the media hub 1028 also performs a resize function.

[0071] In some modes, multiple outputs may be provided by the peripheral device. For example, the peripheral device may output DV, composite and S-video using an uncompressed input from the host. Instead of DV, the peripheral device may input or output other digital video formats, whether compressed or uncompressed. Instead of composite, the peripheral device may input or output other analog video formats. Multiple outputs also may be provided by the peripheral device, in which operations from the above described modes would be performed in parallel.

[0072] The peripheral device also may include a bus connector connected to a data path to the media hub 1028 to allow other devices to connect to the media hub to transfer audio and video data through the peripheral device. This bus connector could be used to allow other processing devices both to provide data to and to receive data from the peripheral device. For example, such a device could receive a high definition video signal and convert it into a compressed data stream which is then transferred to the peripheral device for input to the host computer.

[0073] A communication protocol for controlling communication between the host computer and the peripheral device will now be described. This protocol is divided into two parts, a data transport protocol and a control protocol. The data transport protocol defines how temporal data, such as video, audio and associated metadata, are transferred from the host system to the peripheral device, or from the peripheral device to the host system. The data transport protocol includes both data format and synchronization rules. The control protocol defines how the host system queries and controls the states, modes, and operating parameters of the peripheral device. These protocols are described in more detail below.

[0074] The communication protocol is built on parts of the IEEE-1394 high-speed serial digital bus standard and conforms with the following standards documents, IEEE 1394-1995 Specification IEEE 1394 A Specification. The communication protocol is designed to work with standard IEEE-1394 compliant hardware bus controllers. However, for certain video data formats (e.g.: uncompressed, full-frame, 2-field, 4:2:2) the host and peripheral device controllers support a 400Mb data rate.

[0075] The protocol used to control external devices (such as DV camcorders and VTRs) is known as AV/C, or Audio/Video Control. The specification and command set are defined in the following documents: AV/C Digital Interface Command Set, General Specification, and Version 3.0AV/C Tape Recorder/Player Subunit Specification, Version 2.1. An example data transport protocol will now be described in more detail. In this example, the data-transport protocol is used to transmit streams of video and audio over an IEEE-1394 compliant bus between a host system and a peripheral device. This protocol is sufficiently flexible to allow for expansion for transmission of metadata as well.

[0076] In the following description of the protocol, the host computer is called the host and the peripheral device is called the device. Depending on the mode in use, the host can transmit (for digital cut mode) or receive (for capture mode). The transmitting device is called the talker and the receiving device is called the listener. The term host should be understood to indicate any apparatus that can send or receive data. The term device should be understood to indicate any device that can send or receive data that also processes a periodic reference signal.

[0077] Data transmission is performed in intervals, called data intervals, that correspond to a rate of the periodic reference signal to which temporal data is to be synchronized. For example, if the periodic reference signal corresponds to video, the data interval corresponds to an image, typically a frame, of video data. Example frame intervals include, but are not limited to, 29.97 FPS (for NTSC (525-line) systems) and 25 FPS (for PAL (625-line) systems).

[0078] Data is transmitted in the form of a series of concurrent streams, each identified by a unique stream identifier defined by the protocol. During transmission, streams can be started and stopped at any sample corresponding to a data interval, such as a frame boundary in the video data. The streams of data are broken down into segments based on the data interval in effect. Because of the nature of IEEE-1394 protocol, each data stream is further divided into a series of one or more packets for transmission during a given data interval.

[0079] For each data interval, the talker can choose to transmit any combination of streams that the listening device accommodates. For example, assuming that the listening device can accept two fields of video and two channels of audio, the talker could choose to just send audio or just video, or nothing at all. The listener takes a default action if a stream is missing. For example, if one or more audio streams are missing, which may occur if the audio channel is muted, then the listener fills silence on those channels. If the video data is omitted, which may occur in off speed playback, then the listener might repeat the last frame it received.

[0080] A data transmission includes several types of packets. A "synchronization packet" is sent from the device to the host indicating that a new data interval has begun. The talker then may send data. In one embodiment, a "preamble packet" is sent from the talker to indicate to the listener which streams will be transmitted during the data interval, in what order they will be sent, and how many packets will be used to transport each stream. Zero or more "data packets" then are sent from the talker, based on the information given in the preamble. The talker then waits for the next data

interval to send more data. For example, the talker may send zero or more "null packets" to fill bus cycles on an IEEE-1394 interface until the next data interval begins. This sequence of packets is repeated for each data interval.

[0081] Preamble, data, synchronization, and null packets may use the streaming packet modes of the IEEE-1394 protocol. The IEEE-1394 specification allows for two types of streaming packets: isochronous and asynchronous. For both types, the packet format is the same, utilizing the same transaction code (tcode = 0xA). Asynchronous stream transmissions are similar to isochronous transmissions, but there are the following differences. Asynchronous stream packets are transmitted during the asynchronous bus time, whereas isochronous stream packets are transmitted during the isochronous bus interval. Asynchronous stream packets are limited to a maximum payload of 2048 bytes, whereas isochronous stream packets may contain as many as 4096 bytes.

[0082] The synchronization packet is sent by the peripheral device to indicate that a data interval has begun. The peripheral device generates the synchronization packet in response to the periodic reference signal. More particularly, a synchronization packet is sent at the beginning of each data interval corresponding to the periodic reference signal. The beginning of a data interval may be detected, for example, by monitoring the periodic reference signal. An interrupt or other signal may be issued to the microcontroller of the device when the beginning of a data interval is detected.

[0083] An example format for a synchronization packet 1100, as an asynchronous packet, is shown in Figure 11, and has a format that corresponds to a Write Request packet. The shaded fields indicate those defined by the IEEE-1394 standard. The "destination_ID" field 1102 includes a 16-bit address of the node that is to receive the packet. This field includes a destination_bus_ID (upper 10 bits) and a destination_node_ID (lower 6 bits). To broadcast the synchronization packet to all nodes on the local bus, the destination_bus_ID is set to 3FFh and the destination_node_ID is set to 3Fh. The "tl" field 1104 is the transaction label which is a field that is set by the host. The "rt" field 1106 is the retry code, specifying whether the packet is a first attempt (00b) or a successive retry of the same command (01b through 11b). The "tcode" field 1108 specifies the 1394 transaction code for the packet and is set to "1h", which indicates that this packet is a Write Request packet. The "pri" field 1110 represents a priority code in the IEEE-1394 standard, but is not used. The "source_ID" field 1112 is the address of the node that is sending the packet. The "destination_offset" field 1114 specifies the 48-bit address location on the target that is being accessed. A specific register may be set aside in each possible destination for receiving synchronization packets. This field 1114 shall contain the address of that register. The "device_status" field 1116 is the data sent by the packet and includes information regarding the reception of the previous frame, and the general status of the device. A field that enables error detection and correction ("header_CRC" 1118) also may be included.

[0084] The preamble packet is transmitted by the talker at the beginning of each data interval as a single packet. It may be transmitted as an isochronous packet. This packet specifies which streams will be transmitted during the data interval, and how many packets to expect for each stream. The talker transmits a preamble packet for a data interval even if no stream is to be sent. An example format for a preamble packet 1200 is shown in Figure 12. The shaded fields indicate those defined by the IEEE-1394 standard. The "data_length" field 1202 includes the length in bytes of the payload section of the isochronous packet. The "tag" field 1204 includes the value 00b which indicates that the data is unformatted. The "channel" field 1206 includes a value that indicates the channel assigned to the talker and listener for transmission of this packet. The "tcode" field 1208 includes the value 1010b which indicates that this packet is a streaming packet. The "sy" field 1210 includes the value 0001b to indicate that this packet is a preamble packet. The "transfer rate" field 1212 indicates the rate at which the stream data is arriving in comparison to its real time rate, and is represented by one byte. In other words, it specifies how many frames will be transmitted during this frame time. Example transfer rates are 1x, 2x and 4x real time. The "stream count" field 1214 is an unsigned integer that indicates how many streams will be transmitted per frame during the current data interval. This value is represented by one byte. A number of stream descriptors 1216 follows this quadlet equal to the count specified by the product of the transfer rate field and the stream count field. The list of stream descriptors specifies which streams will be transmitted during this data interval, in what order they will be transmitted, and how many packets are used for each stream. The preamble packet also may include fields that enable error detection and correction ("header_CRC" 1218 and "data_CRC" 1220).

[0085] An example format for stream descriptor 1300 is shown in Figure 13. The "stream ID" field 1302 identifies the type of stream, for example using an 8-bit unsigned integer. The "packet count" field 1304 specifies how many packets will be used to transmit the stream, for example using a 12-bit unsigned integer. Bits in fields marked as "stream-specific-data" in the preamble and stream descriptors are set to zero for the video fields, video frames, system and auxiliary streams. For each audio stream, this value is an unsigned 12-bit integer representing the number of audio samples in the data interval. The total number of data packets which will follow the preamble is the sum of all of the "packet count" fields in the stream descriptor list.

[0086] Example values for stream identifiers for the "stream ID" field are illustrated in the table 1400 of Fig. 14. The stream identifiers in this example do not indicate the format of the data itself (e.g.: uncompressed or DV or JFIF). Instead, the format of the data is established before transmission begins using the control protocol. In this table, each row represents a stream identifier. The "value" column 1402 represents the value that is used as a "stream ID" in a stream descriptor packet. The "name" 1404 and "description" 1404 columns serve to further describe the stream identifier and

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may be used to reference the stream identifier. The following table illustrates example definitions for preambles for different types of video data, specifically DV25, DV50, IMX and uncompressed data, including a list of descriptors for each type.

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name	transfer-rate	stream-count	descriptors
DV25/DV50 @ 1X	1	1	VFRAME
DV25/DV50@ 2X	2	1	VFRAME VFRAME
DV25 @ 4X	4	1	VFRAME VFRAME VFRAME VFRAME
IMX @ 1X	1	10	SYSTEM VFRAME A1 A2 A3 A4 A5 A6 A7 A8
IMX @ 2X	2	10	SYSTEM VFRAME A1 A2 A3 A4 A5 A6 A7 A8 SYSTEM VFRAME A1 A2 A3 A4 A5 A6 A7 A8
uncompressed two	1	10	VFIELD1

(continued)

name	transfer-rate	stream-count	descriptors
field			VFIELD2 A1 A2 A3 A4 A5 A6 A7 A8

[0087] Data packets convey actual audio, video and metadata. An example format for a data packet 1500 is shown in Figure 15. This format corresponds to either an asynchronous packet or an isochronous packet. The data packet format is a basic IEEE-1394 streaming packet. One frame of data for a single stream is broken up into a series of packets of this format. The shaded fields indicate those defined by the IEEE-1394 standard. The "data_length" field 1502 includes the length in bytes of the payload section 1512 of the packet. The "tag" field 1504 includes the value 00b which indicates that the data is unformatted. The "channel" field 1506 includes a value that indicates the channel assigned to the talker and listener for transmission of this packet. The "tcode" field 1508 includes the value 1010b which indicates that this packet is a streaming packet. The "sy" field 1510 includes the value 0000b to indicate that this packet is a data packet. The data packet also may include fields that enable error detection and correction ("header_CRC" 1514 and "data_CRC" 1516).

[0088] The payload 1512 of a data packet includes 1 to 1024 data quadlets (meaning 4 bytes of data) for an isochronous packet. There are 1 to 512 data quadlets for an asynchronous packet. In general, a talker formats the data for a data interval into N packets of data of the same fixed size to fit the payload of the data packet. This size can be the maximum size allowed by the transmission mode being used to efficiently use bandwidth. For example, in the IEEE-1394a standard, this maximum may be either 2,048 or 4,096 bytes. If the length of the data is not evenly divisible by this quantity, then it is permissible to generate a short packet at the end of a stream.

[0089] For example, assume an uncompressed, full frame of NTSC 4:2:2 YUV format video data is transmitted using isochronous mode of IEEE-1394. The frame is transmitted as two fields, there are (240 lines * 720 pixels * 2 bytes-per-pixel =) 345,600 bytes per field. To send a single field in 4096-byte packets, the number of packets used is computed as: 345,600 bytes per field / 4096 bytes per packet = 84 packets with a remainder 1536 bytes, indicating that 84 4096-byte packets and one 1536-byte packet will be transmitted.

[0090] It is also permissible for a talker to transmit a short packet at the beginning of a stream if there are memory page alignment constraints. For instance, if the talker is generating the packets for a stream out of a host buffer that is not aligned on a 4096-byte boundary, then each packet in the stream would cross a page boundary, resulting in poor direct memory access performance. To remedy this, the talker can generate a short packet at the beginning in order to insure that the remaining packets are generated out of buffers that are aligned on 4096-byte boundaries. For example, if the memory buffer for a field to transmit is located at address 119403A4H, then the next 4096-byte boundary is at address 11941000H. That means that there are 119403A4H - 11941000H = C5CH, or 3,164 (decimal) bytes, until the next page boundary. Thus, the first packet would be 3,164 bytes generated out of address 119403A4H, the second packet would be 4,096 bytes generated out of address 11941000H, the third packet would be 4,096 bytes generated out of address 11942000H, and so on.

[0091] The host may transmit null packets for any unused isochronous cycle. During data transmission, one or more null packets may be sent in the data intervals between the transmission of data for each data interval. An example format for a null packet 1600 is shown in Figure 16. This format corresponds to either an asynchronous packet or an isochronous packet. The shaded fields indicate those defined by the IEEE-1394 standard. The "data_length" field 1602 may include any value, including the value 0. If the data_length is zero, then the payload portion 1614 and error correction data 1616 are omitted from the packet. If the data length is non-zero, then dummy data is in the payload 1614 following the header with a number of quadlets corresponding to the value specified in field 1602. The "tag" field 1604 includes the value 00b which indicates that the data is unformatted. The "channel" field 1606 includes a value that indicates the channel assigned to the talker and listener for transmission of this packet. The "tcode" field 1608 includes the value 1010b which indicates that this packet is a streaming packet. The "sy" field 1610 includes the value 0000b. An optional payload portion 1614 of this packet, if present, is ignored by the listener. Fields that enable error detection and correction ("header_CRC" 1612 and "data_CRC" 1616) may be provided.

[0092] Sequencing and timing of packet transmissions for host-to-device and device-to-host transfers using this protocol will now be described.

[0093] As noted above, to transmit data from a host to a device, a host and device first set up parameters of the communication using the control protocol described below. The host then waits, for example by sending null packets, until a data interval begins. These null packets are ignored by the device. At the beginning of the next data interval, a synchronization packet is sent from the device to the host indicating that a new data interval has begun. After receiving the synchronization packet, the host may send data. In one embodiment, a preamble packet is sent from the host to indicate to the device which streams will be transmitted during the data interval, in what order they will be sent, and how many packets will be used to transport each stream. The device processes the preamble packet to set up appropriate parameters of the device. Data packets then are sent by the host. These data packets are then received and processed by the device. After the data for the data interval has been sent, the host then waits for the next data interval. For example, the host may send null packets to fill bus cycles until the next data interval begins. This sequence of packets is repeated for each data interval.

[0094] As noted above, to transmit data from a device to a host, a host and device first set up parameters of the communication using the control protocol described below. The host then waits, for example by sending null packets, until a data interval begins. These null packets are ignored by the device. At the beginning of the next data interval, a synchronization packet is sent from the device to the host indicating that a new data interval has begun. After sending the synchronization packet, the device may send data. In one embodiment, a preamble packet is sent from the device to indicate to the host which streams will be transmitted during the data interval, in what order they will be sent, and how many packets will be used to transport each stream. The host processes the preamble packet to set up appropriate parameters. Data packets then are sent by the device. These data packets are then received and processed by the host. After the data for the data interval has been sent, the device then waits for the next data interval. For example, the device may send null packets to fill bus cycles until the next data interval begins. This sequence of packets is repeated for each data interval.

[0095] In addition to the data transport protocol, a control protocol is provided. Two types of control information are conveyed by the control protocol: commands which control the modes and parameters of the peripheral device hardware (herein called "break-out box (BOB) control"), and commands which control the operation of a DV camcorder or VTR attached to the peripheral device's IEEE-1394 compliant port (called device control).

[0096] BOB control commands may use asynchronous packets in the IEEE-1394 protocol. Device control commands may use the AV/C command set, an industry standard for controlling camcorders and VTRs. AV/C commands intended for the external device are forwarded directly to that device by the peripheral device hardware, and responses are returned to the host in a similar manner. The AV/C commands are defined in several documents, which include: AV/C Digital Interface Command Set General Specification (version 3.0) and AV/C Tape Recorder/Player Subunit Specification. Thus, for device control commands, the communication protocol complies with the existing IEC 61883-1 and IEEE-1394 Trade Association AV/C specifications. The peripheral device relays commands and responses between the host and external device as though the host were directly interacting with the external device.

[0097] BOB control employs a command set that includes command sequences that are transmitted over the IEEE-1394 compliant bus between the host and the peripheral device. Each command sequence includes a command frame transmitted from the host to the peripheral device, and a corresponding response frame transmitted from the peripheral device to the host. Command frames include a command code and optional parameters. Response frames include a success code and optional data values requested by the command.

[0098] Using the IEEE-1394 protocol, a Write Data Block Request format packet is used, having a "tcode" of "1h". The destination offset specified by a packet designates the registers specified for receiving the commands and responses in this protocol. The payload of such a packet includes up to 1024 bytes. The first two bytes of the payload includes a fixed value that represents that the packet includes a BOB control command or response. The next two bytes indicate the packet length. Two more bytes provide the information, such as parameters for a command, status of a command or other information. An additional byte is reserved. The last byte of the payload conveys status information and determines whether the packet is part of a command or a response, and whether it is part of a single- or multi-packet command.

[0099] The BOB control allows the host to control and query the modes, status and parameters of the peripheral device. The various commands depend on the implementation of the peripheral device, its modes and available registers for status information and control settings. To simplify operation of multiple devices connected to the same bus and same periodic reference signal, the BOB control protocol should allow for a command to instruct the peripheral device not to send flow control messages in response to frame boundaries defined by the periodic reference signal.

[0100] The foregoing describes an example device and communication protocol for communication between the device and a host over a serial bus that allows control of the flow of data transferred between the host and the device so that synchronization to a periodic reference signal can be achieved. The protocol involves transferring flow control messages between the peripheral device and the host computer, allowing the peripheral device to control how the host computer sends the uncompressed audio and video data. The audio and video data may be sent as separate streams. Using the

IEEE-1394 protocol, the flow control messages are sent using an asynchronous protocol, whereas uncompressed video and audio data are sent using an isochronous protocol.

[0101] Further, separate streams of uncompressed video and audio data are transferred from the host computer over a serial bus to the peripheral device, where the peripheral device performs operations on the data such as encoding the data into a standard format, such as DV and MPEG, or generating an analog video signal. Decoding of standard format data is not required to generate such an analog video signal, resulting in less processing. Further, when the host computer is used to process digital video and audio data for editing audiovisual programs, the resources on the host computer are not used for encoding video data. Therefore the host computer resources can be used for more creative operations on the video and audio data. The peripheral device also may output standard format data using a standard protocol over a standard bus. For example, the peripheral device may generate DV format data from the uncompressed audio and video data, and output the DV format data using a standard DV transport protocol over an IEEE-1394 compliant serial bus.

[0102] The peripheral device also can be used to synchronize an output of another device, such as a transcoder connected to the peripheral device via a serial bus such as an IEEE-1394 compliant bus, to a periodic reference signal received by the peripheral device. For example, the peripheral device may perform DV encoding of the uncompressed audio and video data received from the host computer. The peripheral device then transfers this DV format data over the serial bus using a standard protocol to the other device. The operation of this other transcoder may be genlocked to a periodic reference signal provided to the peripheral device.

[0103] The host may be configured to receive a stream of audio from the peripheral device during transfer of an audio-video stream to the peripheral device. Such an audio stream would be data received through the peripheral device which may be used to implement audio dubbing or punch-in. In this embodiment, the audio-video stream is sent from the host to the peripheral device as a stream of isochronous packets. The audio data for dubbing or punch-in is sent from the peripheral device to the host as a stream of asynchronous packets. The peripheral device may mix the audio stream with the audio in the received audio video stream for playback through or output to another device.

[0104] For example, referring to Figs. 10A and 10B, audio codec 1006 receives input audio (at 1040, 1041) and produces digital audio data for storage in the media hub 1028. Meanwhile, the IEEE-1394 interface 1004 receives the audio-video data stream and provides it to the media hub 1028. After receipt of a frame of data, the media hub 1028 mixes the received audio data from codec 1006 and from interface 1004 and outputs the mixed data to the codec 1006 and HeadPhone DAC 1012 to generate audio outputs. The microprocessor 1022 is instructed to periodically retrieve audio data from the media hub 1028 to send the data as multiple asynchronous packets to the host over the interface 1051.

[0105] It also should be understood that the host computer and peripheral device as described herein may be any kind of source and recipient connected over a bus to communicate temporal data in synchronization with a periodic reference signal.

[0106] It also should be understood that the host computer may store computer program instructions on a computer readable medium that, when executed by the host computer, cause the host computer to transfer data from the host computer to the peripheral device or to cause the host computer to receive data from the peripheral device. Similarly, the controller 1020 (Figs. 10A-10B) on the device may accessed stored computer program instructions to cause the device to perform a variety of operations.

[0107] Having now described an example embodiment, it should be apparent to those skilled in the art that the foregoing is merely illustrative and not limiting, having been presented by way of example only. Numerous modifications and other embodiments are within the scope of one of ordinary skill in the art and are contemplated as falling within the scope of the invention. Embodiments will be further described and illustrated by way of the following numbered clauses:

1. A method for transferring data from a first device to a second device over a bus, performed by the second device for each data interval defined by a periodic reference signal processed by the second device, and comprising:

- sending a synchronization packet over the bus to the first device for the data interval in response to the periodic reference signal;
 - receiving a plurality of packets over the bus from the first device after receipt by the first device of the synchronization packet, wherein the packets include data for the data interval;
 - repeating the steps of sending and receiving for each data interval defined by the periodic reference signal.

2. The method of clause 1, wherein the plurality of packets comprises:

- a preamble packet indicating a number of data packets to be sent; and
 - one or more data packets.

3. The method of clause 2, wherein the data comprises a plurality of streams of data, and wherein the preamble packet indicates streams to be transmitted during the interval, an order of transmission of the data streams, and a

number of packets used for each stream.

4. The method of clause 2, wherein the plurality of packets further comprises zero or more null packets after the one or more data packets for a remainder of the data interval.
5. The method of clause 2, wherein data packets comprises video data packets.
6. The method of clause 2, wherein data packets comprises audio data packets.
7. The method of clause 2, wherein data packets comprises video and audio data packets.
8. The method of clause 1, wherein the plurality of packets comprises zero or more null packets after the one or more data packets for a remainder of the data interval.
9. The method of clause 1, wherein the data interval corresponds to a frame of video data.
10. The method of clause 1, wherein the data interval corresponds to a field of video data.
11. The method of clause 1, wherein the bus is a serial bus.
12. The method of clause 11, wherein the serial bus is an IEEE-1394 compliant bus.
13. The method of clause 12, wherein the synchronization packet is an asynchronous packet.
14. The method of clause 12, wherein the plurality of packets are isochronous packets.
15. The method of clause 1, wherein the periodic reference signal is received by the second device.
16. The method of clause 1, wherein the periodic reference signal is generated by the second device.
17. The method of clause 1 further comprising:

receiving from the first device a command to start transfer of the data; and receiving from the first device zero or more null packets over the bus until a first synchronization packet for a first data interval is sent by the second device.
18. The method of clause 1, further comprising:

receiving the data in a buffer at the second device; and
outputting the data from the second device in synchronization with the periodic reference signal.
19. The method of clause 18, wherein outputting the data comprises encoding and outputting an analog video signal.
20. The method of clause 18, wherein outputting comprises generating DV format video data.
21. A method for transferring data from a first device to a second device over a bus, performed by the second device for each data interval defined by a periodic reference signal processed by the second device, and comprising:

sending a synchronization packet over the bus to the first device for the data interval in response to the periodic reference signal;
receiving a plurality of packets over the bus from the first device after receipt by the first device of the synchronization packet, wherein the packets include data for the data interval;
after receiving the plurality of packets, waiting for a subsequent data interval defined by the periodic reference signal.
22. A device for connection to source of data over a bus, comprising:

means for processing a periodic reference signal defining a plurality of data intervals,

means for communicating a synchronization packet over the bus to the source of data for each data interval in response to the periodic reference signal;
 means for accepting a plurality of packets over the bus from the source of data after receipt by the source of data of the synchronization packet, wherein the packets include data for the data interval; and
 means for repetitively operating the means for communicating and the means for accepting for each data interval defined by the periodic reference signal.

23. A device for connection to source of data over a bus, comprising:

a bus interface for sending packets to and receiving packets from the source of data over the bus;
 a controller having an input for receiving a signal related to a periodic reference signal defining a plurality of data intervals, and having an output for providing instructing the bus interface to send a synchronization packet over the bus to the source of data for each data interval in response to the periodic reference signal; and
 a buffer connected to receive data for the data interval from a plurality of packets received through the bus interface over the bus from the source of data after receipt by the source of data of the synchronization packet.

24. A method for transferring data from a first device to a second device over a bus, comprising, for each data interval defined by a periodic reference signal processed by the first device:

the first device sending a synchronization packet over the bus to the second device for the data interval;
 the first device sending a plurality of packets over the bus to the second device after sending the synchronization packet, wherein the packets include data for the data interval; and
 the first device waiting until the next data interval according to the periodic reference signal.

25. The method of clause 24, wherein the plurality of packets comprises:

a preamble packet indicating a number of data packets to be sent; and
 one or more data packets.

26. The method of clause 25, wherein the data comprises a plurality of streams of data, and wherein the preamble packet indicates streams to be transmitted during the interval, an order of transmission of the data streams, and a number of packets used for each stream.

27. The method of clause 25, wherein waiting comprises sending zero or more null packets over the bus to the second device for a remainder of the data interval.

28. The method of clause 25, wherein data packets comprises video data packets.

29. The method of clause 25, wherein data packets comprises audio data packets.

30. The method of clause 25, wherein data packets comprises video and audio data packets.

31. The method of clause 24, wherein waiting comprises sending zero or more null packets over the bus to the second device for a remainder of the data interval.

32. The method of clause 24, wherein the data interval corresponds to a frame of video data.

33. The method of clause 24, wherein the data interval corresponds to a field of video data.

34. The method of clause 24, wherein the bus is a serial bus.

35. The method of clause 34, wherein the serial bus is an IEEE-1394 compliant bus.

36. The method of clause 35, wherein the synchronization packet is an asynchronous packet.

37. The method of clause 35, wherein the plurality of packets are isochronous packets.

38. The method of clause 24, wherein the periodic reference signal is received by the first device.

39. The method of clause 24, wherein the periodic reference signal is generated by the first device.

40. The method of clause 24, further comprising:

5 receiving from the second device a command to start transfer of the data; and
sending zero or more null packets over the bus to the second device until a first synchronization packet for a first data interval is sent.

41. The method of clause 24, further comprising:

10 the first device receiving the data in synchronization with the periodic reference signal into a buffer; and the first device transferring the data from the buffer over the bus to the second device.

42. The method of clause 41, wherein receiving the data comprises receiving and decoding an analog video signal.

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43. The method of clause 41, wherein receiving comprises receiving DV format video data.

44. A device for connection to a recipient of data over a bus, comprising:

20 means for receiving a periodic reference signal defining a plurality of data intervals,
means for communicating a synchronization packet over the bus to the recipient of data for each data interval in response to the periodic reference signal;
means for transferring a plurality of packets over the bus to the recipient of data after receipt by the recipient of data of the synchronization packet, wherein the packets include data for the data interval; and
25 means for repetitively operating the means for communicating and the means for transferring for each data interval defined by the periodic reference signal.

45. A device for connection to a recipient of data over a bus, comprising:

30 a bus interface for sending packets over the bus connected to the recipient of data;
a buffer having an input for receiving data and connected to provide data to the bus interface for transmission as a plurality of packets over the bus to the recipient of data; and
a controller having an input for receiving a signal related to a periodic reference signal defining a plurality of data intervals, and having an output for providing an indication to the bus interface to send a synchronization
35 packet over the bus to the recipient of data for each data interval in response to the periodic reference signal.

46. A system for connection to a host computer over a bus, comprising:

40 a peripheral device comprising:
means for receiving a periodic reference signal defining a plurality of data intervals,
means for receiving video data for the plurality of data intervals;
means for communicating a synchronization packet over the bus to the host computer for each data interval in response to the periodic reference signal;
45 means for transferring a plurality of packets over the bus to the host computer after receipt by the host computer of the synchronization packet, wherein the packets include video data for the data interval; and
means for repetitively operating the means for communicating and the means for transferring for each data interval defined by the periodic reference signal; and

50 a video device, comprising:

an input for receiving and synchronizing with the periodic reference signal; and
means for providing the video data to the peripheral device in synchronization with the periodic reference signal.

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47. A system for connection to a host computer over a bus, comprising:

a peripheral device comprising:

- means for receiving a periodic reference signal defining a plurality of data intervals,
means for communicating a synchronization packet over the bus to the host computer for each data interval
in response to the periodic reference signal;
5 means for accepting a plurality of packets over the bus from the host computer after receipt by the host
computer of the synchronization packet, wherein the packets include video data for the data interval; and
means for repetitively operating the means for communicating and the means for transferring for each data
interval defined by the periodic reference signal; and
means for providing video data for the plurality of data intervals; and
- 10 a video device, comprising:
- an input for receiving and synchronizing with the periodic reference signal; and
means for receiving the video data provided by the peripheral device in synchronization with the periodic
reference signal.
- 15 48. A method for transferring data from a first device to a second device over a bus, performed by the first device
for each data interval defined by a periodic reference signal processed by the second device, comprising:
- receiving a synchronization packet over the bus from the second device for the data interval;
20 after receiving the synchronization packet, sending a plurality of packets over the bus to the second device,
wherein the packets include data for the data interval;
after sending the plurality of packets for the data interval, waiting until receipt of a synchronization packet over
the bus from the second device for a subsequent data interval.
- 25 49. The method of clause 48, wherein the plurality of packets comprises:
- a preamble packet indicating a number of data packets to be sent; and one or more data packets.
- 30 50. The method of clause 49, wherein the data comprises a plurality of streams of data, and wherein the preamble
packet indicates streams to be transmitted during the interval, an order of transmission of the data streams, and a
number of packets used for each stream.
- 35 51. The method of clause 49, wherein waiting comprises sending zero or more null packets over the bus to the
second device for a remainder of the data interval.
52. The method of clause 49, wherein data packets comprises video data packets.
53. The method of clause 49, wherein data packets comprises audio data packets.
- 40 54. The method of clause 49, wherein data packets comprises video and audio data packets.
55. The method of clause 48, wherein waiting comprises sending zero or more null packets over the bus to the
second device for a remainder of the data interval.
- 45 56. The method of clause 48, wherein the data interval corresponds to a frame of video data.
57. The method of clause 48, wherein the data interval corresponds to a field of video data.
58. The method of clause 48, wherein the bus is a serial bus.
- 50 59. The method of clause 58, wherein the serial bus is an IEEE-1394 compliant bus.
60. The method of clause 59, wherein the synchronization packet is an asynchronous packet.
- 55 61. The method of clause 59, wherein the plurality of packets are isochronous packets.
62. The method of clause 48, further comprising:

sending to the second device a command to start transfer of the data; and
 sending zero or more null packets over the bus to the second device until a first synchronization packet for a first data interval is received.

5 63. An apparatus for transferring data to a device over a bus, comprising:

means for receiving a synchronization packet over the bus from the device for each data interval defined by a periodic reference signal processed by the device;

10 means, operative after receiving the synchronization packet, for sending a plurality of packets over the bus to the device, wherein the packets include data for the data interval; and

means, operative after sending the plurality of packets for the data interval, for waiting until receipt of a synchronization packet over the bus from the device for a subsequent data interval.

15 64. An apparatus for transferring data to a device over a bus, comprising:

a bus interface for sending packets to and receiving packets from the device over the bus;
 a buffer connected to transfer data to the bus interface for transmission as a plurality of packets; and
 a controller comprising:

20 means for processing synchronization packets received by the bus interface from the device for each data interval defined by a periodic reference signal;

means, operative after receiving the synchronization packet, for causing a plurality of packets to be transferred by the bus interface over the bus between the apparatus and the device, wherein the packets include data for the data interval; and

25 means, operative after transferring the plurality of packets for the data interval, for waiting until receipt of a synchronization packet over the bus from the device for a subsequent data interval.

65. A computer program product, comprising:

30 a computer readable medium;
 computer program instructions stored on the computer readable medium that, when executed by a host computer, cause the host computer to perform a method for transferring data from the host computer to a peripheral device over a bus, performed by the host computer for each data interval defined by a periodic reference signal and comprising:

35 receiving a synchronization packet over the bus from the peripheral device for the data interval;
 after receiving the synchronization packet, sending a plurality of packets over the bus to the peripheral device, wherein the packets include data for the data interval;
 after sending the plurality of packets for the data interval, waiting until receipt of a synchronization packet
 40 over the bus from the peripheral device for a subsequent data interval.

66. A method for transferring data from a first device to a second device over a bus, performed by the second device for each data interval defined by a periodic reference signal processed by the first device, comprising:

45 receiving a synchronization packet over the bus from the first device for the data interval;
 receiving a plurality of packets over the bus from the first device after receiving the synchronization packet, wherein the packets include data for the data interval; and
 after receiving the plurality of packets, waiting until receipt of a synchronization packet over the bus from the first device for a subsequent data interval.

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67. A computer program product, comprising:

a computer readable medium;
 computer program instructions stored on the computer readable medium that, when executed by a host computer, cause the host computer to perform a method for transferring data from a peripheral device to the host computer over a bus, comprising, for each data interval defined by a periodic reference signal processed by the peripheral device:

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receiving a synchronization packet over the bus from the peripheral device for the data interval;
 receiving a plurality of packets over the bus from the peripheral device after receiving the synchronization
 packet, wherein the packets include data for the data interval; and
 after receiving the plurality of packets, waiting until receiving a synchronization packet over the bus from
 the peripheral device for a subsequent data interval.

68. A method for transferring data from a host computer to a peripheral device over a bus, comprising:

the peripheral device sending a synchronization packet over the bus to the host computer for each data interval
 according to a periodic reference signal;
 the host computer sending a plurality of packets over the bus to the peripheral device after receipt of the
 synchronization packet, wherein the packets include data for the data interval; and
 after sending the plurality of packets for the data interval, the host computer waiting until a synchronization
 packet for a subsequent data interval is received.

69. The method of clause 68, wherein the plurality of packets comprises:

a preamble packet indicating a number of data packets to be sent; and
 one or more data packets.

70. The method of clause 69, wherein the data comprises a plurality of streams of data, and wherein the preamble
 packet indicates streams to be transmitted during the interval, an order of transmission of the data streams, and a
 number of packets used for each stream.

71. The method of clause 69, wherein waiting comprises sending zero or more null packets after the one or more
 data packets for a remainder of the data interval.

72. The method of clause 69, wherein data packets comprises video data packets.

73. The method of clause 69, wherein data packets comprises audio data packets.

74. The method of clause 69, wherein data packets comprises video and audio data packets.

75. The method of clause 68, wherein waiting comprises sending zero or more null packets after the one or more
 data packets for a remainder of the data interval.

76. The method of clause 68, wherein the data interval corresponds to a frame of video data.

77. The method of clause 68, wherein the data interval corresponds to a field of video data.

78. The method of clause 68, wherein the bus is a serial bus.

79. The method of clause 78, wherein the serial bus is an IEEE-1394 compliant bus.

80. The method of clause 79, wherein the synchronization packet is an asynchronous packet.

81. The method of clause 79, wherein the plurality of packets are isochronous packets.

82. The method of clause 68, wherein the periodic reference signal is received by the peripheral device.

83. The method of clause 68, wherein the periodic reference signal is generated by the peripheral device.

84. The method of clause 68, further comprising:

the host computer sending a command to start transfer of the data; and
 the host computer sending zero or more null packets over the bus until a first synchronization packet for a first
 data interval is sent by the peripheral device.

85. The method of clause 68, further comprising:
- the peripheral device receiving the data in a buffer; and
 - the peripheral device outputting the data in synchronization with the periodic reference signal.
86. The method of clause 85, wherein outputting the data comprises encoding and outputting an analog video signal.
87. The method of clause 85, wherein outputting comprises generating DV format video data.
88. A system comprising:
- a host computer;
 - a peripheral device connected to the host computer via a bus;
- wherein the peripheral device comprises means for sending a synchronization packet over the bus to the host computer for each data interval according to a periodic reference signal;
- wherein the host computer comprises means for sending a plurality of packets over the bus to the peripheral device after receipt of the synchronization packet, wherein the packets include data for the data interval, and means, operative after sending the plurality of packets for the data interval, for waiting until a synchronization packet for a subsequent data interval is received.
89. A system comprising:
- a host computer comprising:
 - storage for storing compressed video data and associated audio data;
 - an editing system for editing a program using the video data and the audio data and effects to be performed on the video data;
 - a playback system for playing back the edited program in full motion and at full resolution, including means for decompressing the compressed video data and means for processing the effects on the decompressed video data during playback;
 - a peripheral device connected to the host computer via a bus, wherein the peripheral device comprises:
 - means for controlling flow of decompressed video and audio data over the bus from the host computer to synchronize flow with a periodic reference signal;
 - wherein the host computer comprises:
 - means for sending the processed decompressed video with effects and the audio data from the playback system during playback over the bus to the peripheral device in response to flow control by the peripheral device;
 - wherein the peripheral device further comprises:
 - means for compressing the decompressed video and audio data to create a compressed data stream during playback; and
 - means for outputting the compressed data stream to a recording device synchronized with the periodic reference signal during playback.
90. A peripheral device for providing a bridge between a host computer and another device, wherein the peripheral device connects to the host computer with first serial bus and connects to the other device with a second serial bus, wherein the peripheral device comprises:
- means for implementing a protocol over the first serial bus for communication between the host computer and the peripheral device
 - means for implementing the protocol over the second serial bus for communication between the peripheral device and the other device;
 - means for communicating information from the other device to the host computer to allow the host computer to

communicate packets directed to the other device using the protocol over the first serial bus;
means for communicating information from the host computer to the other device to allow the other device to communicate packets directed to the host computer using the protocol over the second serial bus;
means for processing packets received over the first serial bus, including:

- 5 means for determining if the packet is directed to the other device and for transferring the packet to the other device;
- means for determining if the packet is directed to the peripheral device and for processing the packet at the peripheral device; and
- 10 means for determining if the packet is to be processed by the peripheral device but is directed to the other device, and for processing the packet to create one or more new packets and for transferring the one or more new packets to the other device; and
- means for processing packets received over the second serial bus, including:
- 15 means for determining if the packet is directed to the host computer and for transferring the packet to the host computer;
- means for determining if the packet is directed to the peripheral device and for processing the packet at the peripheral device; and
- 20 means for determining if the packet is to be processed by the peripheral device but is directed to the host computer, and for processing the packet to create one or more new packets and for transferring the one or more new packets to the host computer.

Claims

- 25 1. A method for transferring temporal data from a first device (100) to a second device (102) over a bus, performed by the second device for each data interval defined by a periodic reference signal (110) processed by the second device, and comprising:
30 sending a synchronization packet over the bus to the first device for the data interval in response to the periodic reference signal;
receiving a plurality of packets over the bus from the first device after receipt by the first device of the synchronization packet, wherein the packets include data for the data interval;
repeating the steps of sending and receiving for each data interval defined by the periodic reference signal.
35 2. The method of claim 1, wherein the plurality of packets comprises:
a preamble packet indicating a number of data packets to be sent; and
one or more data packets.
40 3. The method of claim 2, wherein the data comprises a plurality of streams of data, and wherein the preamble packet indicates streams to be transmitted during the interval, an order of transmission of the data streams, and a number of packets used for each stream.
45 4. The method of any preceding claim, wherein the plurality of packets further comprises zero or more null packets after the one or more data packets for a remainder of the data interval.
5. The method of any preceding claim, wherein data packets comprises video data packets.
50 6. The method of any preceding claim, wherein data packets comprises audio data packets.
7. The method of any preceding claim wherein data packets comprises video and audio data packets.
8. The method of any preceding claim, wherein the data interval corresponds to a frame of video data.
55 9. The method of any of claims 1 to 7, wherein the data interval corresponds to a field of video data.
10. The method of any preceding claim, wherein the bus is a serial bus.

11. The method of claim 10, wherein the serial bus is an IEEE-1394 compliant bus.
12. The method of any preceding claim, wherein the synchronization packet is an asynchronous packet.
- 5 13. The method of any preceding claim, wherein the plurality of packets are isochronous packets.
14. The method of any preceding claim, wherein the periodic reference signal is received by the second device.
15. The method of any of claims 1 to 13, wherein the periodic reference signal is generated by the second device.
- 10 16. The method of any preceding claim further comprising:
- receiving from the first device a command to start transfer of the data; and
receiving from the first device zero or more null packets over the bus until a first synchronization packet for a
15 first data interval is sent by the second device.
17. The method of any preceding claim, further comprising:
- receiving the data in a buffer at the second device; and
20 outputting the data from the second device in synchronization with the periodic reference signal.
18. The method of claim 17, wherein outputting the data comprises encoding and outputting an analog video signal.
19. The method of claim 17, wherein outputting comprises generating DV format video data.
- 25 20. A device (102), for connection to source of temporal data over a bus, comprising:
- a bus interface (104) for sending packets to and receiving packets from the source (100) of data over the bus;
a controller having an input for receiving a signal related to a periodic reference signal (110) defining a plurality
30 of data intervals, and having an output for providing instructing the bus interface to send a synchronization
packet over the bus to the source of data for each data interval in response to the periodic reference signal; and
a buffer connected to receive data for the data interval from a plurality of packets received through the bus
interface over the bus from the source of data after receipt by the source of data of the synchronization packet.
- 35 21. A system for transferring temporal data comprising:
- a host computer (100);
a peripheral device (102) connected to the host computer via a bus (104);
- 40 wherein the peripheral device comprises means for sending a synchronization packet over the bus to the host
computer for each data interval according to a periodic reference signal (110);
wherein the host computer comprises means for sending a plurality of packets over the bus to the peripheral device
after receipt of the synchronization packet, wherein the packets include data for the data interval, and means,
operative after sending the plurality of packets for the data interval, for waiting until a synchronization packet for a
45 subsequent data interval is received.

Patentansprüche

- 50 1. Verfahren zum Übertragen von zeitlichen Daten von einer ersten Vorrichtung (100) an eine zweite Vorrichtung (102)
über einen Bus, das von der zweiten Vorrichtung für jedes Datenintervall durchgeführt wird, das von einem periodi-
schen Bezugssignal (110) definiert wird, das von der zweiten Vorrichtung verarbeitet wird, und wobei das Verfahren
Folgendes umfasst:
- 55 Senden eines Synchronisationspakets über den Bus an die erste Vorrichtung für das Datenintervall als Antwort
auf das periodische Bezugssignal;
Empfangen von mehreren Paketen über den Bus von der ersten Vorrichtung, nachdem das Synchronisations-
paket von der ersten Vorrichtung empfangen wurde, wobei die Pakete Daten für das Datenintervall enthalten;

Wiederholen der Schritte des Sendens und Empfangens für jedes Datenintervall, das von dem periodischen Bezugssignal definiert wird.

2. Verfahren nach Anspruch 1, wobei die mehreren Pakete Folgendes umfassen:
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ein Präambelpaket, das eine Anzahl zu sendender Datenpakete anzeigt; und
ein oder mehrere Datenpakete.
3. Verfahren nach Anspruch 2, wobei die Daten mehrere Datenströme umfassen und wobei das Präambelpaket wäh-
10
rend des Intervalls zu übermittelnde Ströme, eine Reihenfolge der Übermittlung der Datenströme und eine Anzahl
von Paketen, die für jeden Strom verwendet werden, anzeigt.
4. Verfahren nach einem der vorhergehenden Ansprüche, wobei die mehreren Pakete weiterhin null oder mehr Null-
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pakete nach dem einen oder den mehreren Datenpaketen für einen Rest des Datenintervalls umfasst.
5. Verfahren nach einem der vorhergehenden Ansprüche, wobei Datenpakete Videodatenpakete umfassen.
6. Verfahren nach einem der vorhergehenden Ansprüche, wobei Datenpakete Audiodatenpakete umfassen.
- 20
7. Verfahren nach einem der vorhergehenden Ansprüche, wobei Datenpakete Video- und Audiodatenpakete umfassen.
8. Verfahren nach einem der vorhergehenden Ansprüche, wobei das Datenintervall einem Videodatenrahmen ent-
spricht.
- 25
9. Verfahren nach einem der Ansprüche 1 bis 7, wobei das Datenintervall einem Videodatenfeld entspricht.
10. Verfahren nach einem der vorhergehenden Ansprüche, wobei der Bus ein serieller Bus ist.
- 30
11. Verfahren nach Anspruch 10, wobei der serielle Bus ein IEEE-1394-konformer Bus ist.
12. Verfahren, nach einem der vorhergehenden Ansprüche, wobei das Synchronisationspaket ein asynchrones Paket ist.
13. Verfahren nach einem der vorhergehenden Ansprüche, wobei die mehreren Pakete isochrone Pakete sind.
- 35
14. Verfahren nach einem der vorhergehenden Ansprüche, wobei das periodische Bezugssignal von der zweiten Vor-
richtung empfangen wird.
15. Verfahren nach einem der Ansprfiche 1 bis 13, wobei das periodische Signal von der zweiten Vorrichtung erzeugt
40
wird.
16. Verfahren nach einem der vorhergehenden Ansprüche, das weiterhin Folgendes umfasst:
Empfangen eines Befehls, die Übertragung der Daten zu starten, von der ersten Vorrichtung und
Empfangen von null oder mehr Nullpaketen von der ersten Vorrichtung über den Bus, bis von der zweiten
45
Vorrichtung ein erstes Synchronisationspaket für ein erstes Datenintezvall gesendet wird.
17. Verfahren einem der vorhergehenden Ansprüche, das weiterhin Folgendes umfasst:
Empfangen der Daten in einem Pufferspeicher an der zweiten Vorrichtung und
50
Ausgeben der Daten von der zweiten Vorrichtung in Synchronisation mit dem periodischen Bezugssignal.
18. verfahren nach Anspruch 17, wobei das Ausgeben der Daten das Kodieren und Ausgeben eines analogen Video-
signals umfasst.
- 55
19. Verfahren nach Anspruch 17, wobei das Ausgeben das Erzeugen von Videodaten im DV-Format umfasst.
20. Vorrichtung (102) zur Verbindung mit einer Quelle von zeitlichen Daten über einen Bus, die Folgendes umfasst:

eine Busschnittstelle (104) zum Senden von Paketen an die Datenquelle (100) und Empfangen von Paketen von der Datenquelle über den Bus;
 einen Controller mit einem Eingang zum Empfangen eines Signals, das mit einem periodischen Bezugssignal (110) in Beziehung steht, das mehrere Datenintervalle definiert, und mit einem Ausgang zum Liefern einer
 5 Azweisung an die Busschnittstelle, für jedes Datenintervall als Antwort auf das periodische Bezugssignal ein Synchronisationspaket über den Bus an die Datenquelle zu senden; und
 einen Pufferspeicher, der angeschlossen ist, um Daten für das Datenintervall von mehreren Paketen zu empfangen, die mittels der Busschnittstelle über den Bus von der Datenquelle empfangen wurden, nachdem das
 10 Synchronisationspaket von der Datenquelle empfangen wurde.

21. System zum Übertragen von zeitlichen Daten, das Folgendes umfasst:

einen Hostrechner (100);
 ein Peripheriegerät (102), das über einen Bus (104) an den Hostrechner angeschlossen ist,

wobei das Peripheriegerät Mittel zum Senden eines Synchronisationspakets über den Bus an den Hostrechner für jedes Datenintervall gemäß einem periodischen Bezugssignal (110) umfasst;
 wobei der Hostrechner Mittel zum Senden von mehreren Paketen über den Bus an das Peripheriegerät nach Empfang des Synchronisationspakets, wobei die Pakete Daten für das Datenintervall enthalten, und Mittel, die nach dem
 20 Senden der mehreren Pakete für das Datenintervall funktionsfähig sind, zum Warten, bis ein Synchronisationspaket für ein folgendes Datenintervall empfangen wird, umfasst.

Revendications

1. Procédé servant à transférer des données temporelles à partir d'un premier dispositif (100) vers un second dispositif (102) sur un bus, exécuté par le second dispositif pour chaque intervalle de données défini par un signal de référence
 25 périodique (110) traité par le second dispositif, et comprenant :

l'envoi d'un paquet de synchronisation sur le bus vers le premier dispositif pour l'intervalle de données en
 30 réponse au signal de référence périodique ;
 la réception d'une pluralité de paquets sur le bus à partir du premier dispositif après réception par le premier dispositif du paquet de synchronisation, dans lequel les paquets comprennent des données pour l'intervalle de
 données ;
 35 la répétition des étapes d'envoi et de réception pour chaque intervalle de données défini par le signal de référence périodique.

2. Procédé selon la revendication 1, dans lequel la pluralité de paquets comprend :

40 un paquet de préambule indiquant un nombre de paquets de données à envoyer ; et un ou plusieurs paquets de données.

3. Procédé selon la revendication 2, dans lequel les données comprennent une pluralité de flux de données, et dans lequel le paquet de préambule indique les flux à transmettre pendant l'intervalle, un ordre de transmission des flux
 45 de données et un nombre de paquets utilisés pour chaque flux.

4. Procédé selon l'une quelconque des revendications précédentes, dans lequel la pluralité de paquets comprend en outre zéro ou plusieurs paquets nuls qui suivent le ou les plusieurs paquets de données pour l'intervalle de données
 50 restante.

5. Procédé selon l'une quelconque des revendications précédentes, dans lequel les paquets de données comprennent des paquets de données vidéo.

6. Procédé selon l'une quelconque des revendications précédentes, dans lequel les paquets de données comprennent des paquets de données audio.

7. Procédé selon l'une quelconque des revendications précédentes, dans lequel les paquets de données comprennent des paquets de données audio et vidéo.

8. Procédé selon l'une quelconque des revendications précédentes, dans lequel l'intervalle de données correspond à une trame de données vidéo.
- 5 9. Procédé selon l'une quelconque des revendications 1 à 7, dans lequel l'intervalle de données correspond à un champ de données vidéo.
- 10 10. Procédé selon l'une quelconque des revendications précédentes, dans lequel le bus est un bus en série.
11. Procédé selon la revendication 10, dans lequel le bus en série est un bus FireWire conforme.
12. Procédé selon l'une quelconque des revendications précédentes, dans lequel le paquet de synchronisation est un paquet asynchrone.
- 15 13. Procédé selon l'une quelconque des revendications précédentes, dans lequel la pluralité de paquets sont des paquets isochrones.
14. Procédé selon l'une quelconque des revendications précédentes, dans lequel le signal de référence périodique est reçu par le second dispositif.
- 20 15. Procédé selon l'une quelconque des revendications 1 à 13, dans lequel le signal de référence périodique est généré par le second dispositif.
16. Procédé selon l'une quelconque des revendications précédentes, comprenant en outre :
25 la réception d'une commande à partir du premier dispositif pour commencer le transfert des données ; et
la réception de zéro ou plusieurs paquets nuls à partir du premier dispositif sur le bus jusqu'à ce qu'un premier paquet de synchronisation soit envoyé par le second dispositif pour un premier intervalle de données.
- 30 17. Procédé selon l'une quelconque des revendications précédentes, comprenant en outre :
la réception des données dans une mémoire tampon au niveau du second dispositif ; et
la sortie des données à partir du second dispositif en synchronisation avec le signal de référence périodique.
- 35 18. Procédé selon la revendication 17, dans lequel la sortie des données comprend le codage et la sortie d'un signal vidéo analogue.
19. Procédé selon la revendication 17, dans lequel la sortie comprend la génération de données vidéo dans un format DV.
- 40 20. Dispositif (102) pour la connexion à une source de données temporelles sur un bus, comprenant :
une interface de bus (104) servant à envoyer des paquets vers la source de données (100) et à recevoir des paquets à partir de celle-ci sur le bus ;
un contrôleur ayant une entrée pour recevoir un signal lié à un signal de référence périodique (110) définissant
45 une pluralité d'intervalles de données, et ayant une sortie pour fournir l'instruction à l'interface de bus d'envoyer un paquet de synchronisation sur le bus vers la source de données pour chaque intervalle de données en réponse au signal de référence périodique ; et
une mémoire tampon connectée pour recevoir des données pour l'intervalle de données à partir d'une pluralité de paquets reçus à travers l'interface de bus sur le bus à partir de la source de données après réception par
50 la source de données du paquet de synchronisation.
21. Système servant à transférer des données temporelles comprenant :
un ordinateur hôte (100) ;
55 un dispositif périphérique (102) connecté à l'ordinateur hôte via un bus (104) ;
dans lequel le dispositif périphérique comprend le moyen d'envoyer un paquet de synchronisation sur le bus vers l'ordinateur hôte pour chaque intervalle de données selon un signal de référence périodique (110) ;
dans lequel l'ordinateur hôte comprend le moyen d'envoyer une pluralité de paquets sur le bus vers le dispositif

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périphérique après réception du paquet de synchronisation, dans lequel les paquets comprennent des données pour l'intervalle de données, et le moyen, opérationnel après l'envoi de la pluralité de paquets pour l'intervalle de données, d'attendre jusqu'à ce qu'un paquet de synchronisation soit reçu pour un intervalle de données ultérieur.

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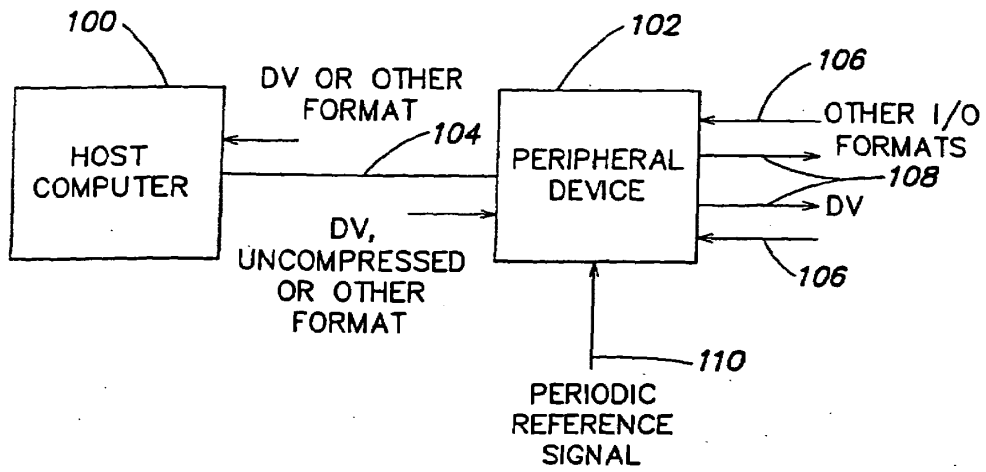


FIG. 1

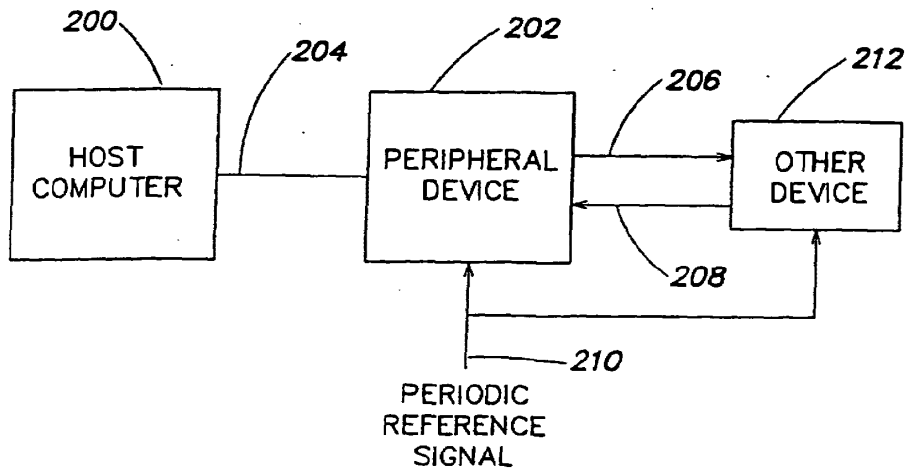


FIG. 2

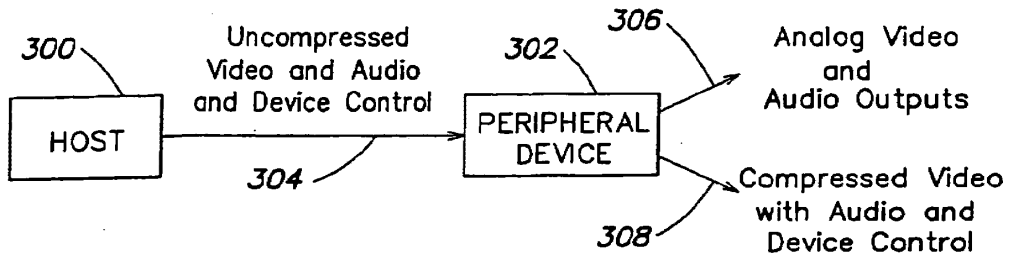


FIG. 3

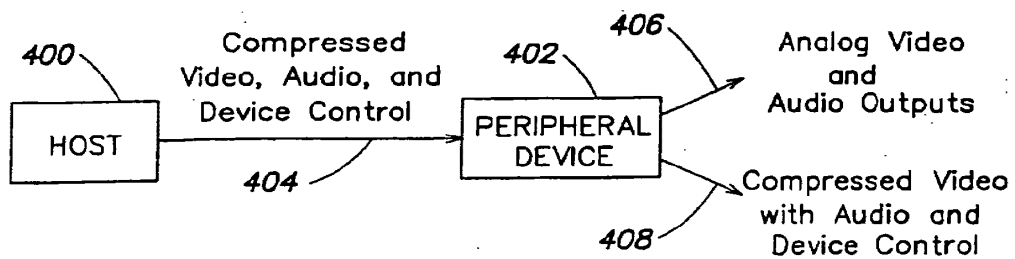


FIG. 4

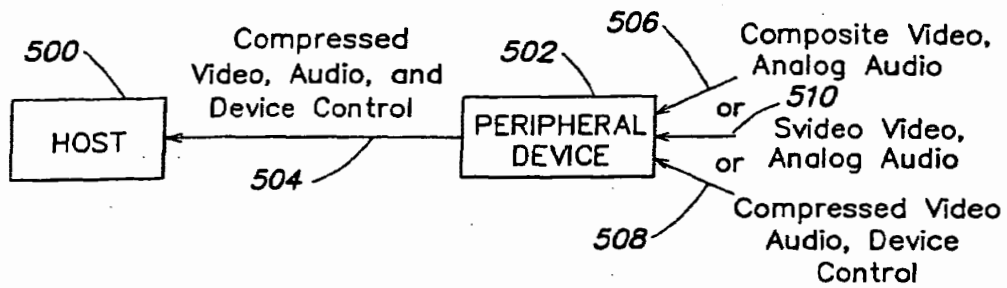


FIG. 5

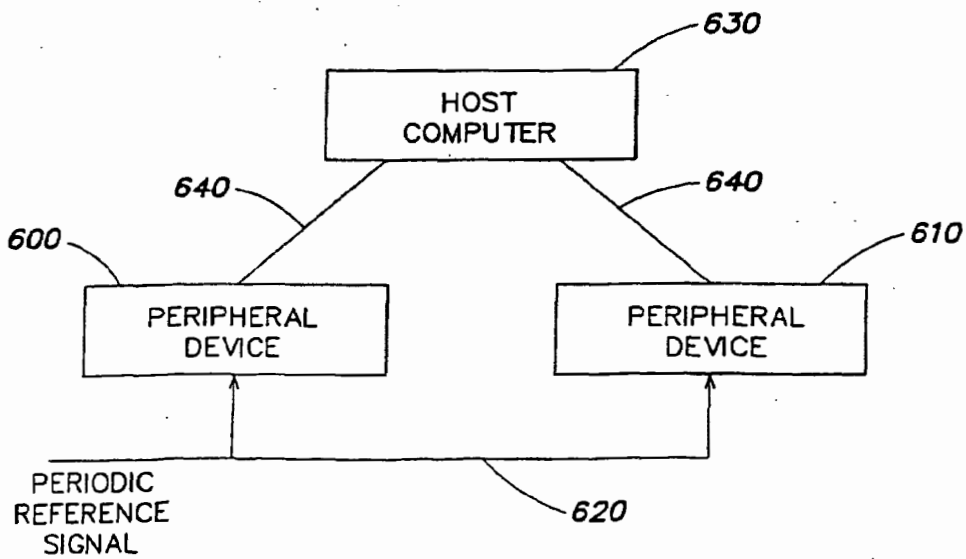


FIG. 6

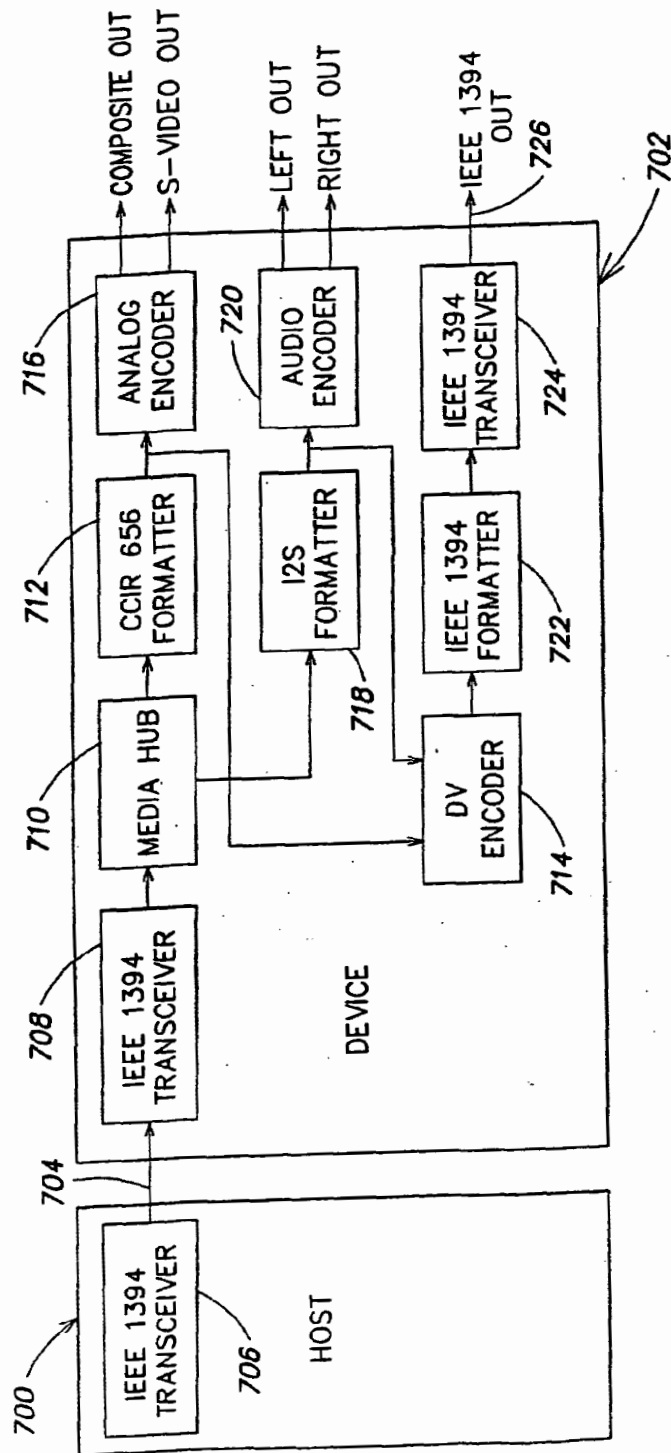


FIG. 7

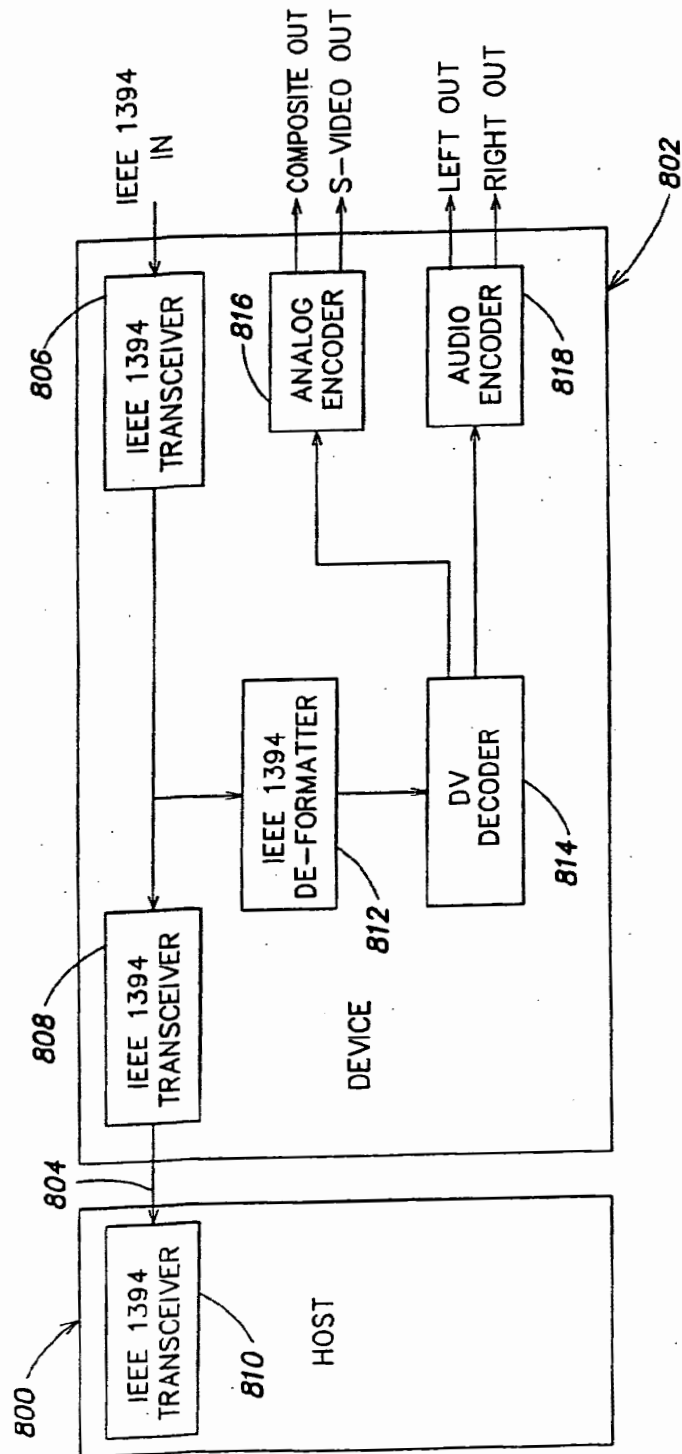


FIG. 8

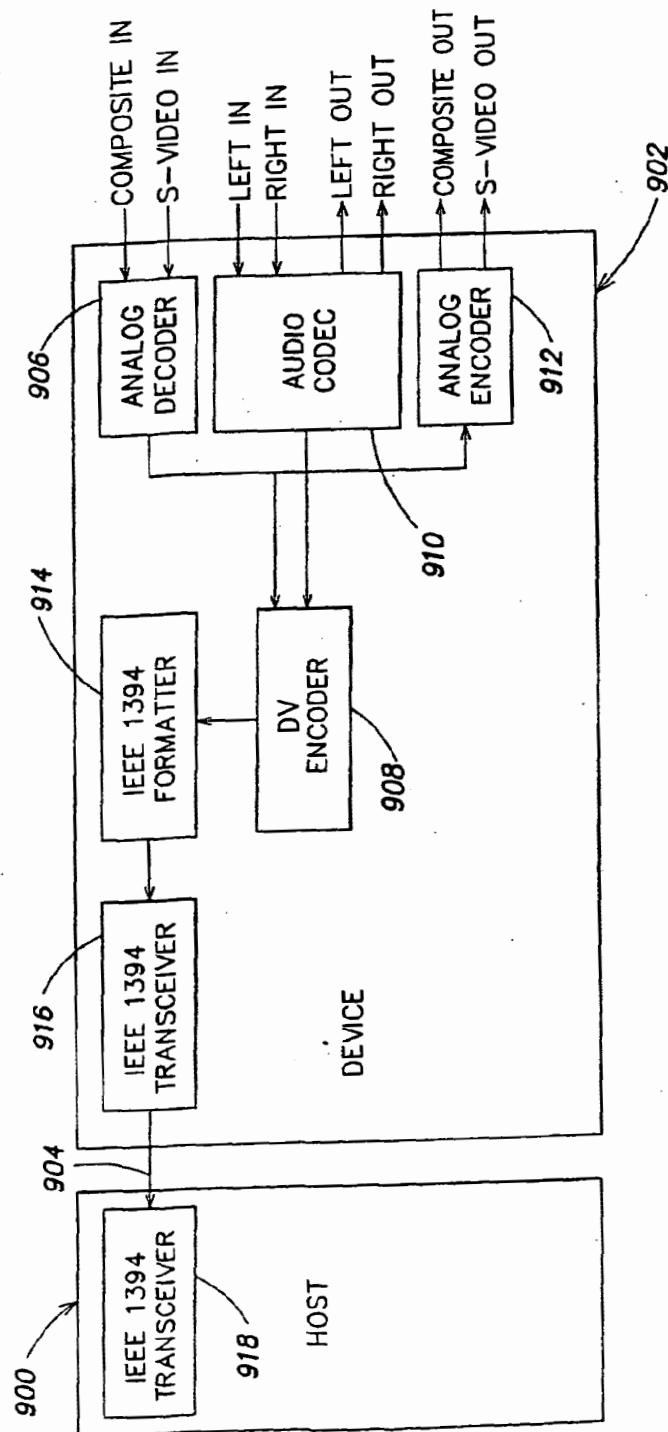


FIG. 9

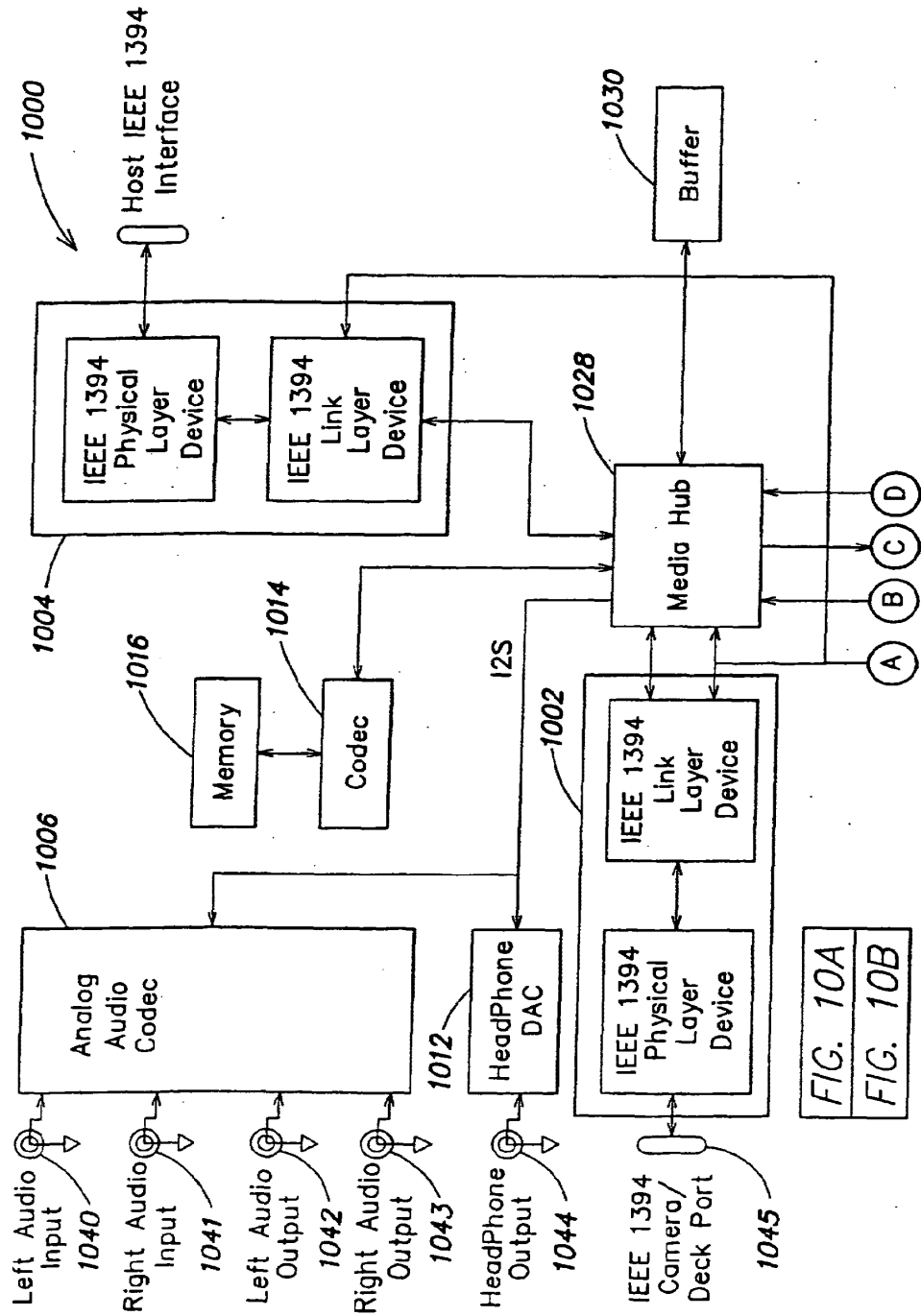


FIG. 10A

FIG. 10A
FIG. 10B

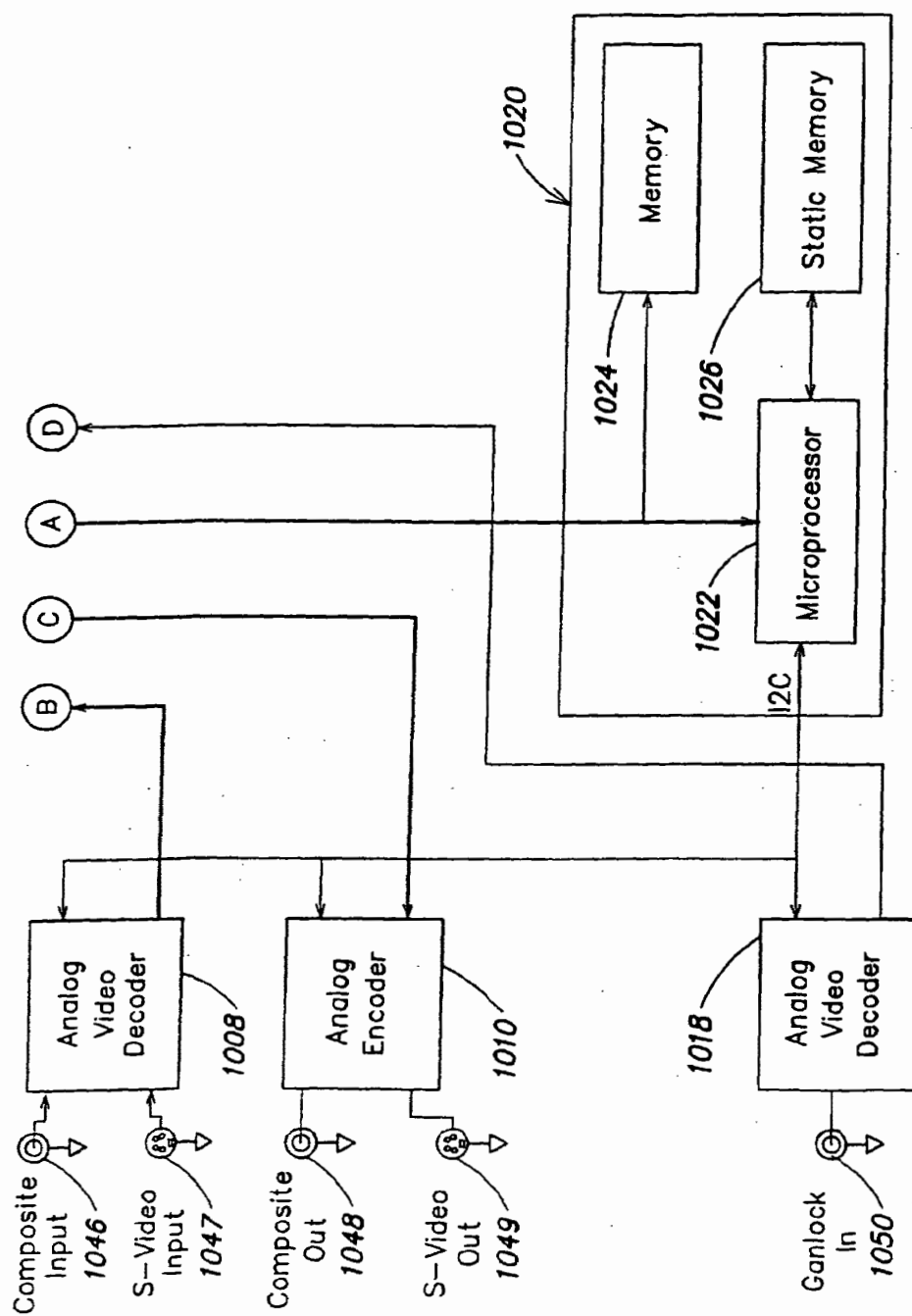


FIG. 10B

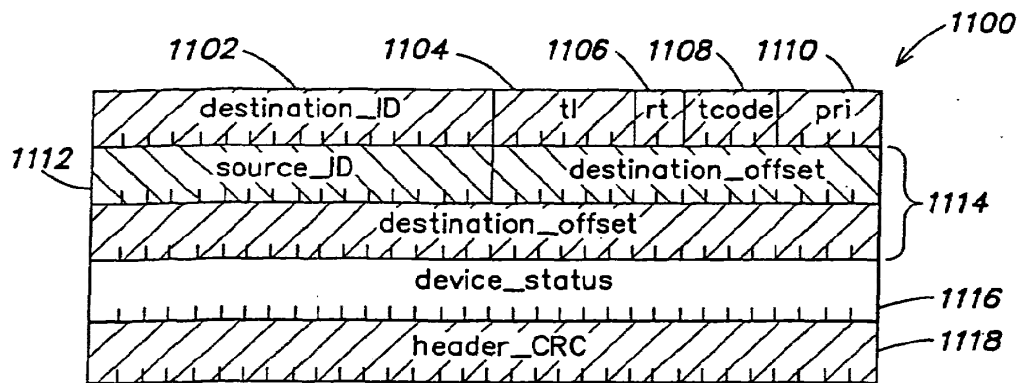


FIG. 11

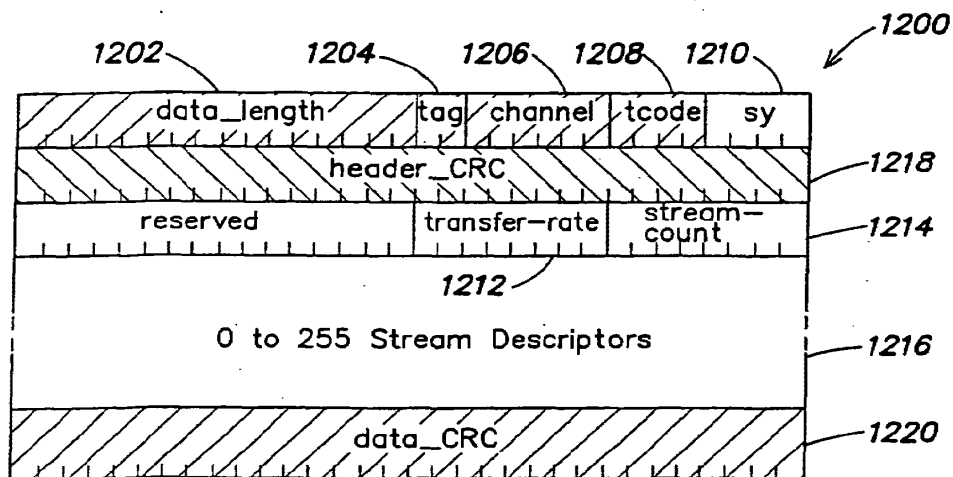


FIG. 12

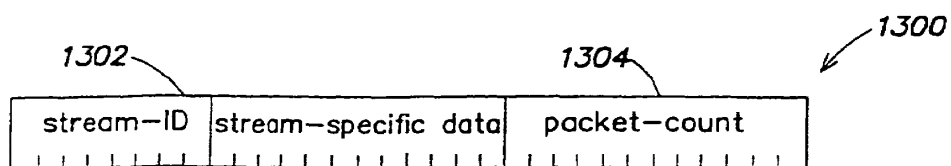


FIG. 13

NAME	VALUE	DESCRIPTION
VFIELD1	1	field 1 video data
VFIELD2	2	field 2 video data
VFRAME	3	full frame video data
SYSTEM	32	system data
AUXILIARY	33	auxiliary data
A1	64	A1 audio data
A2	65	A2 audio data
A3	66	A3 audio data
A4	67	A4 audio data
A5	68	A5 audio data
A6	69	A6 audio data
A7	70	A7 audio data
A8	71	A8 audio data

FIG. 14

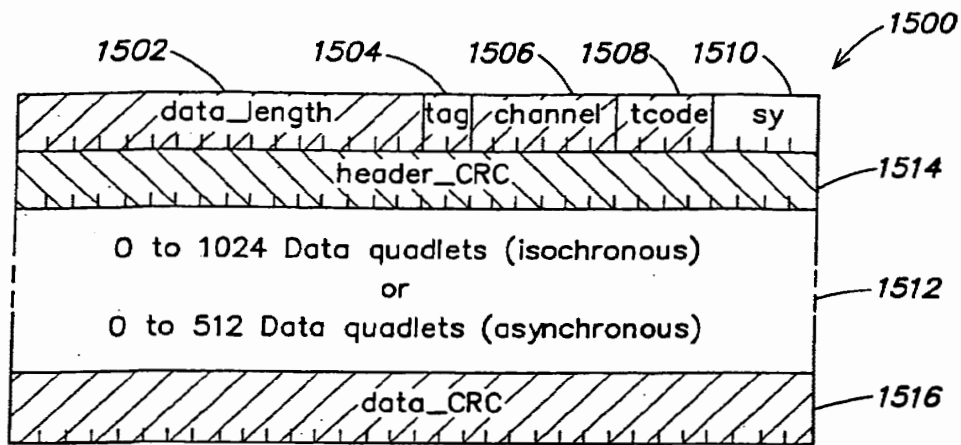


FIG. 15

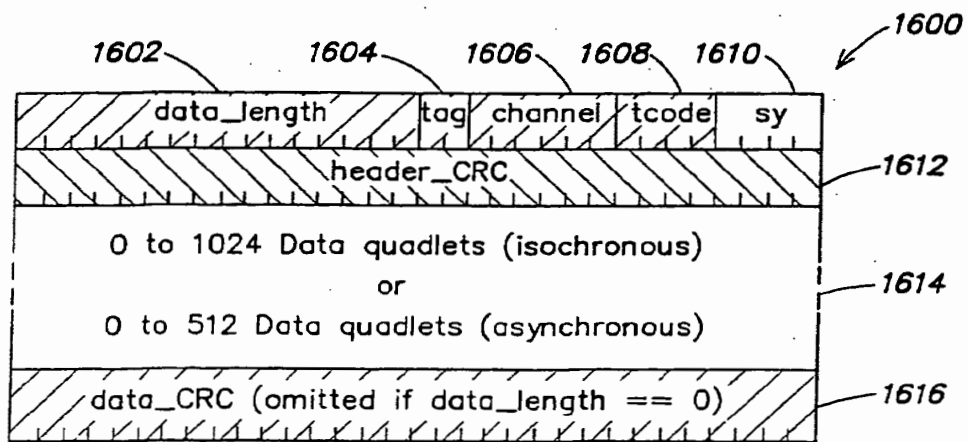
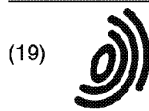


FIG. 16



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(71) Applicant: Phoenix Gold International, Inc.
9300 North Decatur Portland, OR 97203 (US)

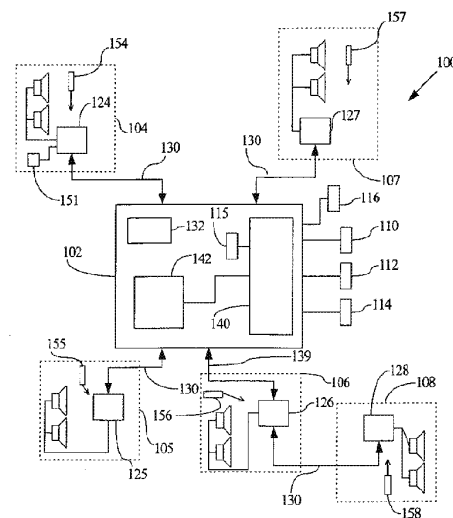
(72) Inventors:
• Devin, Craig, c/o Devin Design & Development
90277 Redondo Beach, CA (US)
• White, Dorian
90025 Los Angeles, CA (US)
• Williams, Steve Phoenix Gold International, Inc.
97203 Portland, OR (US)
• MacBeth, Roland c/o Phoenix Gold Intern., Inc.
97203 Portland, OR (US)

(74) Representative: Lang, Johannes, Dipl.-Ing.
Bardehle Pagenberg Dost Altenburg Geissler,
Postfach 86 06 20
81633 München (DE)

(54) Digital audio distribution system

(57) A digital audio distribution network includes a hub configured to receive analog or other audio signals from one or more audio sources. The hub includes analog-to-digital converters that produce digital audio signals and a remote control input that receives control data from a remote control device. An audio processor combines the digital audio signals and the control data for delivery to one or more audio zones as a digital audio distribution signal. The audio processor also receives a digital audio distribution signal and produces an analog audio signal for delivery to one or more speakers. The audio processor also decodes control data from the input digital audio distribution signal and directs the decoded control data to an infrared repeater. The hub also delivers electrical power to additional network hubs or other network elements using a cable that also conducts one or more digital audio distribution signals. A remote control processor is configured to learn additional remote control commands, and a serial interface is provided for network configuration and troubleshooting.

FIG. 1



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Description**TECHNICAL FIELD**

[0001] The disclosure pertains to digital audio distribution systems.

BACKGROUND

[0002] Residential audio systems typically provide audio content from a variety of sources. A typical system offers AM/FM radio, analog and digital audio tape, compact discs, and a variety of digital file formats. Thus, audio system users have available a broad spectrum of audio materials. Audio systems adapted to this variety of audio sources can require corresponding components such as AM/FM tuners, CD players, satellite television or set top box audio feeds, DVD audio, SuperCD format, computer streamed audio from the internet, etc., and can be large and expensive. In addition, system users frequently customize audio installations for a particular setting to, for example, accommodate room acoustics or permit convenient selection of favorite audio sources.

[0003] While sophisticated, versatile audio components are available, system users still encounter significant difficulties in obtaining the intended audio programming. Audio components are typically situated in and configured for a particular location, and require the user to remain in this location to enjoy the intended audio programming. In addition, sophisticated audio components can represent a considerable expense, and users are typically unwilling to configure additional audio components for other locations. Thus, while an audiophile may have a state-of-the-art audio system in a designated area in her residence, she typically must still resort to listening to a portable audio system or lower fidelity component such as a clock radio or intercom equipped with CD player or FM tuner boom box while in other areas. Therefore, methods and systems for audio distribution are needed.

SUMMARY

[0004] Digital audio networks comprise a hub configured to receive a plurality of stereo signals and combine the stereo signals with control data to produce a digital audio network signal. An output connector is configured to communicate the digital audio network signal from the hub to additional network elements downstream or upstream along an output cable. In representative examples, the output connector is also configured to deliver electrical power to the output cable in accordance with, for example, standard IEEE 802.3AF usage guidelines. In additional examples, the output connector is configured to deliver the digital audio network signal to a first pair of conductors in the output cable and electrical power to at least a second pair of conductors in the output

cable. In representative examples, the output connector is configured to deliver the digital audio network signal to a first pair of conductors in a CAT 5 cable, and to deliver electrical power to at least a second pair of conductors in the CAT 5 cable. In additional examples, the hub includes an input connector configured to receive a digital audio network signal. In additional examples, an infra-red control input is configured to receive the control data. In further examples, a digital signal processor is configured to process stereo audio data associated with at least one of the stereo signals.

[0005] Controllers for a digital audio network comprise a keypad configured for entry of at least one audio source identifier or configuration parameter for at least one audio playback output or a plurality of output locations grouped as a zone. An audio processor is configured to receive an input digital audio distribution signal that includes digital audio data and control data, and produce an output digital audio distribution signal based on the entry of the at least one audio source identifier or configuration parameter. In specific examples, a display is configured to display the at least one audio source identifier or configuration parameter. In other examples, an infrared sensor is configured to receive control data from a modulated infrared emitter and the audio processor is configured to include the control data from the modulated infrared emitter in the output digital audio distribution signal.

[0006] Audio network devices comprise an infrared sensor configured to receive control data from a modulated infrared emitter, and an audio processor configured to produce a digital audio distribution signal based on the received control data. In representative examples, an input is configured to receive at least one audio input signal, wherein the audio processor is configured to produce a digital audio signal based on the input audio signal, and the digital audio distribution signal is based on the digital audio signal. In additional examples, a user input is configured to select at least one audio signal to be decoded from the digital audio distribution signal. In further examples, a tone generator is configured to produce an audio tone, wherein the audio processor is configured to include the audio tone in the digital audio distribution signal. In additional examples, an input audio transducer is in communication with the audio processor and configured to generate an audio signal. The audio processor is configured to include the audio signal generated by the input audio transducer in the digital audio distribution signal.

[0007] Audio distribution methods comprise receiving at least one audio signal from an audio signal source and digitizing the at least one audio signal to produce a digital audio signal. The digital audio signal is combined with control data associated with the audio signal source to produce a digital audio distribution signal. In additional examples, the control data is received from a remote control device. In other examples, the digital audio distribution signal is transmitted to an audio zone. An ana-

log audio signal is extracted from the digital audio distribution signal, and the extracted analog signal is delivered to an audio output device. In other examples, the extracted analog signal is configured based on the control data.

[0008] Audio distribution methods comprise receiving a plurality of digital audio signals and receiving control data. A digital audio distribution signal is produced based on a combination of selected digital audio signals and the control data. In other examples, the digital audio distribution signal is transmitted and the control data is extracted from the digital audio distribution signal and delivered to at least one audio source. In additional examples, the control data is associated with at least one of source volume, loudness, or muting. This control data is separated and routed to an on-board microprocessor for interpretation, routing and digital signal processing control functions. In other examples, the control data is associated with identification of a particular digital audio signal for delivery to a selected zone. In other representative examples, the method includes delivering electrical power with the transmitted digital audio distribution signal.

[0009] In additional examples, systems are configured to recognize previously undefined control commands received from, for example, an IR remote control, notify a system user of such receipt, and initiate a programming or "learning" sequence to associate a selected function with the undefined command, thereby defining the command. A remote control processor can be included that is in communication with a library of commands, or that can process new commands that are not in the command library. In other examples, network elements are configured to include program material in a digital audio distribution signal, and to recognize control commands to be executed remotely for inclusion in the digital audio distribution signal. In other examples, a communication interface is provided for communication with an external device such as a PDA or a personal computer or a touch panel controller for network installation, maintenance, and updates.

[0010] These and other features and advantages are described below with reference to the accompanying drawings.

BRIEF DESCRIPTION OF THE DRAWINGS

[0011]

FIG. 1 is schematic diagram of a digital audio distribution network.

FIG. 2A is schematic diagram of connection panel for a digital audio distribution network hub.

FIG. 2B is a schematic block diagram of an eight stereo channel digital audio distribution network hub.

FIG. 3 is schematic diagram of a portion of a network hub illustrating analog-to-digital conversion of

two stereo audio signals.

FIGS. 4A(1)-4A(3) contain a schematic diagram of a portion of a network hub configured to receive digitized stereo audio signals and produce a digital audio network distribution signal that includes digital audio data for a plurality of stereo sources and control data associated with control of the stereo sources.

FIG. 4B is a schematic diagram of a portion of a network hub that includes status inputs for a plurality of audio sources and output connections for IR emitters associated with control of audio sources and network elements.

FIG. 5 is a schematic diagram of a portion of a network hub configured to produce analog audio signals based on digital audio signals received from a digital signal processor.

FIGS. 6A-6C contain a schematic diagram of a keyboard module for a digital audio distribution network.

FIGS. 7A-7B are schematic diagrams of front panels of representative user control modules.

FIG. 8 is a schematic diagram of a front panel of a remote power module.

FIG. 9 is a schematic diagram of a front panel of an input/output module.

FIGS. 10A(1)-10A(3) is a schematic diagram of user control module that includes a processor configured to learn remote control commands and a serial interface.

FIG. 10B is a block diagram illustrating processing of command received from a remote control.

FIG. 11 is a schematic diagram of a digital audio distribution network arranged in a point-to-point configuration.

FIG. 12 is a schematic diagram of a digital audio distribution network arranged in a star configuration.

FIG. 13 is a schematic diagram of another representative digital audio distribution network.

FIGS. 14A-14C contain a schematic diagram of a hub that includes a serial interface and a processor configured to learn control commands.

FIGS. 15A-15B contain a schematic diagram of a keypad module configured to communicate with a digital signal processor.

FIG. 16 is a schematic diagram of an IR output module configured to control IR emitters based on a processed IR signal.

DETAILED DESCRIPTION

[0012] With reference to FIG. 1, a representative audio distribution network 100 includes a hub 102 configured to communicate with representative zones 104-108. The hub 102 is configured to receive audio signals and other signals from external sources 110, 112, 114 such as, for example, CD players, DVD players, tun-

ers, satellite decoders, intercom systems, alarm systems, intercoms microphones, etc. The hub 102 can be configured to accept, for example, sixteen audio data channels (eight stereo sources) as inputs, and can transmit eight audio data channels (four stereo sources) to the zones 104-108, but fewer or more channels can be provided. The zones 104-108 are typically associated with areas in which audio signals are to be received. In a residential installation, the zones can correspond to various rooms in a residence (media room, dining room, family room) or locations (deck, pool, workshop, back yard). One or more audio sources such as a representative source 115 can be included in the hub 102, and digital audio sources such as a representative digital source 116 can be used. In addition, sources can be situated in one or more zones such as a source 151 situated in the zone 104.

[0013] As shown in FIG. 1, the zones 104-108 are provided with digital audio distribution (DAD) processors 124-128, respectively, that are interconnected with cables 130, typically so-called Category 5 ("Cat 5") cable. Cat 5 cables include eight conductors arranged in four pairs of conductors. As described below, the DAD processors of the zones can be different, and DAD processors can be connected via another DAD processor to the hub 102 or to other hubs. The DAD processors are typically configured to receive a digital audio distribution signal and decode/encode digital audio signals and control data as described below.

[0014] The hub 102 also receives and routes infrared remote (IR) or other control information from remote controllers 154-158 in the zones 104-108, respectively, to and from system components such as the DAD processors 124-128 and the sources 110, 112, 114, 116, 151. The IR information is typically transmitted as control data packets that are combined with digital audio signals for full duplex or bi-directional delivery over the cables 130. For example, a DAD processor in a selected zone can be configured to receive a command from an infrared or other wireless remote controller, and deliver the command to the hub 102. In this manner, users can control remote sources as well as control other DAD network elements.

[0015] The hub 102 also includes a power supply 132 configured to provide power to the digital audio distribution system via the cables 130. In a representative example, the power supply is configured to operate with input voltages of between about 90 VAC to 260 VAC and produce an output of about 24 VDC at 2.5 Amps.

[0016] A digital audio processor 140 is configured to receive analog audio signals from the sources 110, 112, 114 and produce corresponding digital signals. In a representative example, the processor 140 is based on a so-called ADAT chipset available from ALESIS SEMI-CONDUCTOR and is configured to convert the eight channels of analog input (four stereo pairs) to a single digital signal, typically a 24-bit, 48 kHz digital signal. Operation of this chipset is described in U.S. Patent

5,297,181. The processor 140 also receives IR information and other control information, and combines the control information with the digital audio signal. The combined audio/control signal is transmitted over a pair of conductors, and return (or duplex) information is received on a second pair of conductors. Power (e.g., 24 VDC) and ground are carried on the third and fourth pair of conductors, respectively. The hub 102 is also configured to sense input signal voltages that can be used to verify the status of a source component as ON or OFF based on, for example, detecting that a signal is present for greater than 2 seconds.

[0017] The digital audio processor 140 is configured to implement digital signal processing to provide user-selectable and customizable audio processing. The hub 102 and the digital audio processor 140 can be configured to provide user-selectable audio or control functions such as Auto Level in which audio signal amplitudes of signals from different sources are adjusted via intelligent compression. Auto Loudness can also be selected in which magnitudes of audio signal portions in selected frequency bands, typically low and high frequency bands, are adjusted based on an audio volume level. Typically, low and high frequency signal components are increased at low audio volume levels. Multi-band parametric equalization can also be provided in which, for example, parameters for five frequency bands can be selected for parametric equalization for a selected zone or zones. The frequency bands can be user-selectable, and adjustable boost/cut and "Q" can be provided at selected frequencies or frequency bands. Muting during power up/power down, and temporary muting for one or more selected zones can also be provided.

[0018] The hub 102 can be configured to have a default configuration such as, for example, a power-on configuration. In addition, the hub 102 can include a display 142 and an input interface for controlling and configuring audio delivery for one or more of the zones. For example, upon initialization, audio inputs 1-4 can be identified in the display 142 as INPUT1-INPUT4, respectively. During a configuration process, the inputs can be assigned other names such as "CD Player," "Tuner," "Tape," or other names. These assignments can be stored for display in subsequent system initializations. These assignments can also be displayed by DAD network elements situated in other zones by combining name assignment data with audio data in the digital audio distribution signal.

[0019] The hub 102 can also include a tone generator and can be configured to receive an alarm signal or to be in communication with a doorbell switch. In response to the alarm signal or the doorbell switch, a tone generated by the tone generator can be delivered to one or more zones and to override previously selected programming. In addition, intercom components can be included in the network 100 for operation under control of the hub 102.

[0020] With reference to FIG. 2A, connections to a

hub such as the hub 102 can be provided with a connection panel 200 that includes source modules 201, 207, 213, 219 that are configured to receive and output stereo audio signals as left/right (L/R) pairs. The source module 201 includes a left input connector 202, a right input connector 203, and associated L/R output connectors 204, 205, respectively. Various connector types can be used, but in one example, so-called RCA connectors are used. The source modules 207, 213, 219 can be similarly configured. The source modules 201, 207, 213, 219 are associated with respective trigger inputs 206, 212, 218, 224 that are configured to receive "trigger signals" that indicate that the associated source is turned on. In addition, the source module inputs are sampled by a digital audio processor to determine if the associated audio input pair is active (i.e., is receiving audio), and a STATUS LED 272 is used to indicate if any source module inputs are active. A determination of activity can be based on, for example, presence of an audio signal for a predetermined interval, such as 2 seconds. The panel 200 includes a coaxial connector 232 or other connector that can be configured to receive electrical power from an external power supply.

[0021] A remote source module 240 includes left and right output connectors 242, 244 and can be associated with a selected source module to provide, for example, subwoofer output or speaker output, or to receive a mono output associated with a stereo input. An RJ45 connector 250 is configured to communicate audio data and control data in a digital audio distribution signal and electrical power to and from additional network components located in one or more zones. An additional RJ45 connector 252 is provided for expansion or connection to a digital bus. A universal serial bus (USB) port 256 can also be provided for software revisions or for network control. A 4-conductor connector 258 is provided for a remote IR receiver for zones for which a system controller is unavailable. A master/slave switch 262 can be configured to select a selected hub as an originating source for a system clock. A green LED 268 is configured to indicate that power is available to the hub, a red LED 270 is configured to indicate data transmission on all RJ45 connections, and the blue LED 272 is configured to indicate that a voltage is present at a Status Input, typically 5-15 VDC. Connectors 281-284 are configured to receive infrared repeaters for transmission of IR-controller based commands. The connectors are typically 3.5 mm phone jacks, but other types of connector can be used. These IR repeaters are generally wired in parallel.

[0022] With reference to FIG. 2B, a hub configured for eight stereo sources (sixteen mono channels) typically includes A-D converters 285-288 that are configured to receive two analog stereo inputs and produce corresponding stereo digital signals that are delivered to a controller 289. The stereo digital signals typically include both left and right channel audio in a single digital signal. The controller 289 is configured to receive net-

work control commands or commands associated with audio or other sources via an infrared sensor unit 290 that can be configured to receive commands from a conventional IR remote controller. The controller 289 combines the stereo digital signals with control data and delivers to produce a digital audio network signal that is delivered to an output 291. The controller 289 is also configured to receive a digital audio network signal from an input 292 and to produce processed audio signals and a word clock signal that are delivered to a processed audio output 293. Operation of the hub is indicated with an LED display panel 294 that indicates audio channel status and the presence of audio data, as well as producing an IR signal based on a processed IR signal produced by the controller 289. Analog stereo output signals are produced by D-A converters 294-297 and can be based on audio signals provided to the A-D converters 285-289 or on audio signals associated with a digital audio network signal received at the input 292, or both.

[0023] Referring to FIG. 3, a portion of a network hub is configured to receive audio signals from two stereo sources at a left source 1 (L1) input 302, a right source 1 (R1) input 304, a left source 2 (L2) input 306, and a right source 2 (R2) input 308. Alternatively, monophonic audio signals can be provided to one or more of these source inputs. The source inputs are connected to signal conditioners 303, 305, 307, 309 that process the audio input signals for delivery to respective direct outputs 310-313 as processed analog signals. In addition, processed audio signals from the L1 input 302 and the R1 input 304 are delivered as differential signals to an analog-to-digital converter 320, and processed audio signals from the L2 input 306 and the R2 input 308 are delivered as differential signals to an analog-to-digital converter 322. Corresponding digital signals are delivered to A-D outputs 326, 328, respectively. A word clock (WDCLK) input 330 is configured to receive a clock signal for synchronization of the digital audio output signals produced by the A-D converters 320, 322.

[0024] With reference to FIGS. 4A(1)-4A(3), a representative network hub includes input/output connectors 402, 404 configured to receive and transmit digital audio distribution network signals and deliver electrical power to other remote or local network elements. The I/O connectors 402, 404 are typically RJ45 connectors that are configured so that pin pairs 1, 2 are configured to output digital audio distribution network signals that include audio data for up to four stereo pairs and control data, pins 3, 6 are configured to receive a digital audio distribution network signal that includes digital audio data for up to four stereo pairs and control data, and pin pairs 4, 5 and 7, 8 are configured for DC power and ground, respectively. Digital audio distribution network signals are transmitted to and from the I/O connectors 402, 404 via respective signal conditioners 406, 408 and are delivered to respective digital audio decoders 410, 412 that are configured to produce up to four digital outputs corresponding to each of the stereo pairs presented to the

I/O connectors 402, 404. The decoded digital audio signals are delivered to a complex programmable logic device (CPLD) 414 such as, for example, a XILINX XC9572 or other logic device. The digital audio decoder 410 is also configured to produce a clock signal (WD-CLK) that is delivered to the digital audio decoder 412 and to a WDCLK output 411, but in other examples, the decoder 412 can be configured to produce a clock signal or a clock signal can be provided from another network component.

[0025] Digital stereo inputs 420 are configured to receive digital stereo signals produced by, for example, A/D converters as shown in FIG. 3. Such digital stereo signals are delivered to the CPLD 414 along with status signals for up to eight stereo channels received from one or more status signal inputs 422. Rotary encoders 424, 426 are configured to provide control signals to encoder inputs of the CPLD 414. The CPLD 414 is configured to deliver selected digital stereo signals to a digital signal processor (DSP) 428 that delivers processed digital audio signals to processed digital outputs (PDOUT) 430, 432. Referring to FIG. 5, the processed digital outputs are in communication with respective D-A converters 502, 504 that produce associated left channel and right channel analog audio signals. The left channel audio signals are delivered to signal conditioners 510, 511 and output connectors 520, 521, and the right channel audio signals are delivered to signal conditioners 512, 513 and output connectors 522, 523.

[0026] Referring again to FIG. 4, a microprocessor 450 is configured to receive control data extracted from an input digital audio distribution network signal by the audio decoders 410, 412. This control data is generally associated with IR remote control signals and is processed for delivery to a processed IR output 452. The audio decoders 410, 412 also communicate with a Data Present output 453 to indicate that a digital audio distribution network signal is available for decoding. The microcontroller 450 is configured to selectively provide a mute signal to the decoders 410, 412 based on control data extracted from the digital audio distribution network signal or otherwise provided. The microprocessor is also configured to communicate with and control signal processing by the DSP 428. An EEPROM 454 is in communication with the microprocessor 450, and a power conditioner 456 is also provided. An encoder 417 is configured to combine digital audio signals received from the CPLD 414 with control data for delivery as a digital audio distribution signal to pins 1, 2 of connector 404.

[0027] With reference to FIG. 4B, set 470 of input connectors is configured to receive signals associated with a power on/power off condition of up to eight audio sources. A status output (STATUS[1..8]) 471 is configured to deliver associated status signals to, for example, the CPLD 414. Based on the power on/off signals and a data present signal delivered to a DATA PRESENT input 472, LEDs 474-476 are turned on. A processed IR input receives a processed IR signal for delivery to a

set 480 of connectors for delivery to IR repeaters that can be situated to deliver IR signals to, for example, IR remote control inputs of one or more audio sources.

[0028] Referring to FIGS. 6A-6C, a network keypad 600 includes audio signal conditioners 602, 604 and respective audio decoders 610, 612 that are in communication with respective input/output connectors 601, 603. Decoded digital audio (typically as a combined left/right digital signal) is delivered to a CPLD 614 and control data is delivered to a microprocessor 616. An audio encoder 618 is configured to deliver a digital audio distribution signal to the signal conditioners 602, 604 and to the connectors 601, 603. Audio data can be selected for inclusion in the output signal using encoders 620, 622. Keypad switches 631, 632, 633, 634 are configured for control of audio volume, selection of a particular zone, muting, and source selection, respectively. An encoder 636 is configured for selection of zones or audio sources, or other audio network features. The microprocessor 616 is configured to deliver a processed IR control signal to an LED array 640 that is used to, for example, communicate infrared remote control signals to one or more audio sources. These control signals are typically based on conventional infrared remote control signal protocols. The microprocessor is also in communication with an IR sensor 642 that is configured to receive IR control signals from an IR remote controller and transmit associated electrical control signals. A connector 644 can be provided for communication with a DSP module.

[0029] With reference to FIG. 7, a keypad-based controller 700 includes an LCD display, typically a 128 by 64 back or side illuminated LCD. The controller 700 typically includes two IDC, RJ45, or other connectors for Cat5/Cat5e cable for connection to a DAD network, and is configured to receive electrical power from, for example, four of the eight conductors available in a Cat5 cable. The controller 700 is configured to fit into a standard 2-gang electrical workbox and includes a cover 702 that can be attached to the workbox with clips, screws, or other mounting mechanisms. The controller 700 is also configured to receive an amplifier module to power speakers as well.

[0030] The controller 700 includes user perimeter input buttons 704-709 and central input buttons 722-726. The input buttons can be configured to select listening sources, name the sources, define the routing of sources, set parametric equalization and auto-leveling compression and limiting, as well as adjust volume and balance in a local listening area (a local zone) or in remote zones. A display unit 730 includes a display area configured to display source names or other data, and regions 731-733 configured to display audio power, balance, tuner signal strength and other audio network and audio source information. The controller 700 is generally configured so that switching between audio sources produces a default audio volume level, typically -30 dB, unless an auto level function is activated so that audio output level is unchanged. If the input buttons are un-

used for a predetermined time such as, for example, 10 seconds, a currently selected audio source name is displayed such as CD, DVD, tuner etc. on the display 730. The controller 700 also includes an IR input 712.

[0031] With reference to FIG. 7B, an alternative representative controller 750 includes general purpose input buttons 751-755 as well as illuminated buttons 762, 764 associated with selection of auto audio leveling and loudness, respectively. Pressing such buttons toggles the corresponding function from on to off or off to on. If the function is active, the associated button can remain illuminated. Source selection buttons 766-769 permit selection of a source for audio delivery or assignment of configuration parameters. These buttons function similarly to the input buttons 751-755.

[0032] Controllers such as the representative controllers 700, 750 can be used as follows. For example, in response to pressing and holding an "ENTER" button for 1 sec., messages such as "Press UP and DOWN buttons to adjust Volume" or "Press LEFT and RIGHT buttons to adjust balance" are presented on a display. Pressing a "SOURCE" button allows an installer or user to select a source and assign the source to multiple destinations and create a zone from these destinations. The display can present messages such as, for example, "Use the UP / DOWN buttons to select a source," "Use the RIGHT / LEFT buttons to select destinations" and "Press ENTER to link source/destination." A selected input button can be designated as, for example, an "Auto Volume" button to adjust all audio sources to produce a similar volume in a selected zone or zones, or a "Remote Off" button to turn off a remote input from an I/O module in a selected zone. Transmission from selected I/O Modules in other zones can also be inhibited. One or more of the input buttons can serve as a menu button for user selection of functions such as equalization adjustments and assignment of names to rooms and zones as well as configuring zones. An IR sensor in a controller is configured to receive IR commands from a remote control for transmission over the network back to one or more source components, as well as commands associated with volume control or balance in one or more zones, zone muting, and switching a zone or the system on or off.

[0033] With reference to FIG. 8, a power module 800 is configured for connection to a digital audio distribution network using IDC, RJ45, or other connectors and CAT-5 or CAT-5e cable, or other cables. The power module 800 also includes a coaxial or other input connector to receive supplemental power from an external power source (not shown in FIG. 8). The power module 800 includes a green LED indicator 802 configured to indicate that an additional power supply connected to the power module 800 is providing power. Supplemental power is provided to the network using selected conductors in the CAT-5 or other cable. The power module 800 is typically included in a network installation to compensate power losses produced by, for example, power dis-

tribution over system cables. The power module 800 also provides, for example, 24-bit, 48 kHz digital audio transmission. In a particular example, the power module 800 is configured to receive 24 V DC from the supplemental power source. The power module 800 is conveniently configured for mounting in a conventional electrical box. In other examples, a power module includes an internal power supply. The power module can be configured to act as a hub as well, and can include IR sensors, D-A or A-D converters, DSP, and other functions.

[0034] With reference to FIG. 9, an input/output (I/O) module 900 for a DAD network includes a connector configured to receive a CAT-5 or CAT-5e cable. The I/O module 900 also includes left and right stereo inputs 902, 903 configured to receive an analog signal from a remote source, typically a source located in a zone associated with the I/O module 900. The I/O module 900 includes a digital audio processor that is configured to convert the analog signal to a corresponding digital signal that can be included in a digital audio network distribution signal that is transmitted to other zones. The I/O module 900 also includes left and right channel stereo outputs 904, 905 configured for delivery of an audio signal associated with a selected source to a local amplifier (i.e., an amplifier associated with a selected zone) or directly to speakers or headphones. The inputs and outputs can be configured to use so-called RCA connectors or other types of connectors. A power supply input connector 908 is also provided if additional electrical power is needed due to, for example, a large number of DAD network components and an associated large power demand, or power losses associated with deliver of power to the I/O module 900 via long cable lengths. The I/O module 900 also includes a connector 910 (e.g., a 3.5 mm phone jack) to deliver a control signal to an IR emitter for control of a source in the local zone from a remote zone. A similar connector 912 is configured as a 2 V trigger input to receive a signal from a local signal source that indicates that the local signal source is operational. Data associated with the trigger input is included in the digital audio network distribution signal that is transmitted by the I/O module 900. For convenience, the I/O module 900 can be configured to fit in a standard electrical box. The I/O module 900 can also include a DSP or other processor, one or more internal audio sources, or be configured to receive digital audio directly from one or more audio sources.

[0035] Another example of a user control module is illustrated in FIGS. 10A(1)-A(3). The user control module is similar to that shown in FIGS. 6A-6C, but includes a processor 1002 configured to communicate with an IR emitter and an IR detector via a jack 1004 and a connector 1006, respectively. The processor can be, for example, an SP8001 Serial Port Remote Control Integrated Circuit with "Learning" produced by Innotech Systems Inc. Such a processor includes a library of remote control codes for remote controls made by many manufacturers and can be operated to acquire or "learn" ad-

ditional codes that can be associated with activation of various keys/buttons on a remote control. In addition, such a processor can be configured to automatically search for a correct code. In addition, such a processor can store three digit ("ISI") codes for several devices. The processor 1002 also includes outputs associated with LEDs 1008, 1009 for evaluating progress in learning codes. In a particular example, a code is learned by providing a button code for a virtual button to be associated with a code and the code to be learned is directed from a remote control to an IR detector and to the processor 1002. Confirmation that the code is learned is obtained by monitoring the LEDs 1008, 1009. With such a processor, commands associated with arbitrary remote control codes can be implemented without hardware modification. The user control module (or a network hub or other network element) can recognize control codes received via IR, wireless, or otherwise and convert or translate control codes for control of a selected device such as a program source or a network element. Such a processor can be associated with other network elements such as network hubs or input/output modules or other network locations. Some commands can be executed directly at the user control module (volume changes, muting) while other commands are directed to other network locations. For example, a command associated with a pause function is directed to the program source to be paused. In another example, a command is associated with selection of a different program source, and execution of such a command can be based on selection of particular audio data from a digital audio data distribution signal, without communication with the program source. In some cases, the digital audio data signal does not include the selected program material, and the command is directed appropriately in the network.

[0036] FIG. 10B illustrates processing of a command received by a control module from a remote control such as an IR remote. In a step 1050, a command is received and in a step 1052 the command is checked to determine if the command is a known command. If the command is not recognized as a known command, an error can be posted or control passed to a command learning procedure 1054. If the command is recognized, the command can be evaluated to determine if the command can be executed by the control module in a step 1056. If a "local" command is received, the command is executed in a step 1060. Otherwise, the command is distributed by the network in a step 1062 to other network locations in the digital audio data for execution at an appropriate location.

[0037] The user control module of FIG. 10A also includes a serial interface 1012 that can be configured to receive or transmit network configuration parameters such as, for example, program source information, equalization settings, zone definitions, or other settings. The serial interface 1012 can be implemented based on, for example, RS-232, universal serial bus, FIREWIRE (IEEE 1394), or other communication protocol. As

shown in FIG. 10A, serial output is delivered to and receive from a DB9 connector 1014. Using the serial interface 1012, network settings can be stored, evaluated, or copied to duplicate a particular network configuration. In addition, the serial interface 1012 can be used for communication with an additional touch panel controller, a personal computer, a personal digital assistant, or other control system. Such a control system can include a display that permits visual presentation of the network configuration and network parameters. The serial interface 1012 can also be configured to communicate with a touch panel controller using a wireless interface. In addition, the serial interface 1012 can be used to reprogram or update an installed network to, for example, learn additional commands or to add additional features such as room monitoring, door chime functionality, or a security system interface if not provided at installation. The serial interface 1012 can also be used to establish a remote connection to the network using, for example, a phone line and a modem for remote control, repair, and evaluation. Alternatively, the serial interface 1012 can be configured to communicate over the Internet.

[0038] In other examples, audio distribution systems can be configured to distribute audio program material or other program material to one or more zones that can be configured as one or more rooms or other regions in a user's residence or business, or other location. As noted above, program material can be situated in one or more zones and made available to all zones or to selected zones, and while a central location for generation of source material can be used, such a location is not necessary. A system can be arranged in a point-to-point configuration (a "daisy chain") or in a star configuration. A typical point-to-point configuration is illustrated in FIG. 11. An equipment rack 1100 includes various program sources such as a first tuner 1101, a CD player 1102, a cassette player/recorder 1103, a second tuner 1104, and a slot 1105 for an additional source. Analog audio signals from these program sources are delivered to a hub 1106 that is in communication with a user interface module 1108 situated in, for example, a kitchen. The user interface module 1108 is typically configured to control program material delivered to speakers 1110, 1111 and to receive and transmit digital data to/from the hub 1106 and a user interface 1114 situated in a study or other room or zone. Digital data is communicated between the user interface 1114 and a user interface 1116 situated in master bedroom or other zone. The user interface 1116 is in communication with an input/output unit 1118 that is configured to communicate analog program material to and from a television 1120. The input/output unit is in communication with a user interface 1122 situated at a patio or other zone or location. As shown in FIG. 11, each zone/room includes speakers that are configured to receive analog program material from the associated user interface. Category 5e cable and RJ45 connectors are typically used to interconnect the user interface modules and input/output modules,

but other types of cables can be used, or one or more of the system modules can include a wireless interface so that cables are not needed. In addition to program material, command, control, and program information can be communicated from zone to zone or room to room in the digital audio data signal. Typically each of the digital network components (such as, for example, input/output modules, user interfaces, or hubs) can be associated with a device identifier that can be set with, for example, one or more encoder wheels.

[0039] With reference to FIG. 12, a digital audio network 1200 arranged in a star configuration includes user interface modules 1202, 1204, 1206, 1208 that are in communication with an expansion module 1210. The expansion module 1210 is in communication with a hub 1214 and a set of program material sources 1212. As shown in FIG. 12, each user interface module is associated with a single user location (patio, bedroom, study etc.), but in other examples, one or more user interface modules can be associated with a collection of rooms or other locations that form a zone. A master bedroom zone 1216 include an input/output module 1218 to, for example, communicate an audio portion of television programming to the network 1200 or to supply audio programming from, for example, any of the set of sources 1212 to a television 1220.

[0040] FIG. 13 illustrates another example network that includes an equipment rack 1302 configured to retain, for example, AM/FM tuners 1304, 1305, a satellite tuner 1306, CD and DVD players/recorders 1307, 1308, a videocassette recorder 1309, a cable television box 1310, and a cassette player/recorder 1311. Hubs 1312, 1314 are configured for distribution to respective point-to-point networks. Zone 1320 includes a power amplifier 1322 that processes analog audio signals for delivery to speakers 1324, 1325.

[0041] Other configurations of network components include a hub that includes a processor 1402 configured to learn remote control commands and a serial interface 1404 as shown in FIGS. 14A-14C. FIGS. 15A-15B illustrates an additional keypad controller configuration. FIG. 16 illustrates an IR output module that includes jacks 1601-1608 configured to communicate IR emitter drive signals based on processed IR received at an input 1610. In the above examples, any zone can typically be configured to receive and transmit control data to, for example, control remote program sources (e.g., turn on or off a CD player in another room, or select an FM channel at a remote tuner) or to control the network (e.g., select parameters for audio processing, or select audio sources for delivery to a selected zone). Control data is conveniently provided using infra-red emitting diodes and detectors, and an IR remote in one zone can select and control program material delivery in other zones.

[0042] Typically, network components and modules are interconnected using so-called Cat 5 or similar cables using RJ45 connectors. These components and modules are configured so that crossover cables are un-

necessary, and all network cables can be configured as "straight-through" cables.

[0043] User interface modules can be conveniently arranged as wall mounted key pads with an LCD or other display that is configured to show, for example, the source currently being played, other available sources, equalization and loudness settings, volume, tone and other settings. In addition, a user interface can be configured to show a source material identifier such as a performer or song name and this information included in a digital audio data distribution signal. The user interface module can also be configured to indicate user preferences for tone, equalization, loudness etc. as a function of volume or other parameter. Typically a user interface module is configured to receive an amplifier module so that the user interface can directly drive one or more speakers. Amplifier power is typically received at the user interface module via the associated Cat 5 cable. In addition, user interface modules are generally configured to include a "last source recall" so that upon activation of a single button, a recently played source is returned to a previous configuration. For example, a particular FM station can be selected via "one touch" and played with previous volume, tone, equalization, or other parameters. In addition, a user interface module can be configured so that audio levels associated with all or a selected number of program sources are played at a common audio level.

[0044] Other system modules include power input modules to provide additional power, system expanders, and infrared (IR) repeaters configured to reproduce IR commands sent from remote zones to a program material source. An IR repeater is typically configured to be situated at an IR receiver at the program source.

[0045] For some program sources, program material details such as composer, performer, selection length, playlists, or other details are available, and such information can be provided in the digital audio distribution signal. For example, program information concerning channel selection and current program material obtained from a satellite tuner can be distributed. In addition, text messages can be distributed.

[0046] Only selected specific examples of audio networks and the use of these networks is described herein, but it will be apparent that such networks permit users to conveniently select, process, control, and deliver audio signals from one or more sources to one or more destinations or zone. For example, a user located in a first zone can direct a command to an IR remote to select a remote audio source for audio delivery to the first zone (or other zones), and control the audio source by selecting, for example, CD track, FM tuner frequency, or other audio source configuration. A network hub in the first zone is responsive to the IR remote, and processes a remote control command for the network. In some cases, the remote control command is relayed to the audio source or other destination. For example, selection of an audio source is based on network hub configuration,

while audio source control (e.g., initiating playing of a CD or selecting a CD track) is based on delivery of a command to an audio source.

[0047] In the examples above, examples of particular network elements have been described with respect to associated network functions. However, any network element can be configured as a network controller or a user input device, and can serve as a hub. Such assignments of functionality to particular network elements in the examples is for convenience in explanation only. In addition, in the examples, digital audio distribution signals are transmitted and received on respective assigned pairs of an 8-conductor cable while the remaining four conductors are used to provide electrical power. In other examples, signals and power can be arranged on additional or fewer conductors, and the used of 8-conductor CAT-5 cable is for convenience only. In addition, analog or other signals associated with audio, video or other media can be transmitted or received using some cable conductors. In some examples, system functions are implemented in hardware or software, but generally functions can be implemented in any combination of hardware and software, or using only hardware or software. In other examples, program parameters can be stored at some or all distribution points. For example, parameters for functions such as automatic loudness control can be stored for frequently used or favorite sources and distribution zones. Thus, it will be apparent that the examples described above can be modified in arrangement and detail without departing from the scope of the disclosure. We claim all that is encompassed by the appended claims.

Claims

1. A digital audio network, comprising:

a hub configured to receive a plurality of stereo signals and combine the stereo audio signals with control data to produce a digital audio network signal; and
an output connector configured to communicate the digital audio distribution signal; and
an interface configured to communicate at least one network parameter.

2. The digital audio network of claim 1, wherein the hub is configured to include program material information in the digital audio distribution signal.

3. The digital audio network of claim 1, wherein the output connector is configured to deliver the digital audio distribution signal to a pair of conductors.

4. The digital audio network of claim 1, further comprising an interface configured to communicate at least one network parameter.

5. The digital audio network of claim 1, wherein the output connector is configured to deliver the digital audio distribution signal to a first pair of conductors in a CAT-5 cable, and to deliver electrical power to at least a second pair of conductors in the CAT-5 cable.

6. The digital audio network of claim 1, further comprising an input connector configured to receive a digital audio distribution signal.

7. The digital audio network of claim 1, further comprising a remote control processor configured to process remote control commands.

8. The digital audio network of claim 1, further comprising a display configured to present program material information.

9. A controller for a digital audio network, comprising:

a keypad configured for entry of at least one audio source identifier or a configuration parameter for at least one audio zone;
an audio processor configured to receive an input digital audio network distribution signal that includes digital audio data and control data, and produce an output digital audio network distribution signal based on the entry of the at least one audio source identifier or configuration parameter; and
a serial interface configured to communicate at least one of the at least one audio source identifier or the configuration parameter for the at least one zone.

10. The controller of claim 9, further comprising a display configured to display the at least one audio source identifier or configuration parameter.

11. The controller of claim 9, further comprising a remote control processor configured to identify an unknown remote control command and to indicate entry into a command acquisition mode.

12. The controller of claim 11, wherein the audio processor is configured to include the control data associated with the unknown remote control command into the output digital audio network distribution signal after command acquisition.

13. An audio network device, comprising:

an infrared sensor configured to receive a remote control command;
a control data processor that includes an indicator configured to indicate if the received re-

remote control command is undefined and acquire an associated command definition; and an audio processor configured to produce a digital audio distribution signal based on the command definition.

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14. The audio network device of claim 13, further comprising:

an input configured to receive at least one audio input signal, wherein the processor is configured to produce a digital audio signal based on the input audio signal and the digital audio distribution signal is based on the digital audio signal.

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15. The audio network device of claim 14, further comprising a user input configured to select at least one audio signal to be decoded from the digital audio distribution signal.

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16. The audio network device of claim 14, further comprising an interface configured to communicate at least one network parameter.

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17. The audio network device of claim 16, wherein the interface is a serial interface.

18. An audio distribution method, comprising:

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receiving at least one audio signal from an audio signal source;
digitizing the at least one audio signal to produce a digital audio signal; and
combining the digital audio signal with control data received from a remote control to produce a digital audio distribution signal.

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19. The audio distribution method of claim 18, further comprising identifying a command received from a remote control as a new command.

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20. The audio distribution method of claim 19, further comprising associating a network function with the new command.

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21. An audio distribution method, comprising:

receiving a plurality of digital audio signals;
receiving control data;
determining a location for execution of a command associated with at least a portion of the control data; and
producing a digital audio distribution signal based on a combination of selected digital audio signals and the command location.

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22. The method of claim 21, further comprising ex-

ecuting a command associated with the control data without inclusion of the command in the digital audio distribution signal.

23. The method of claim 21, further comprising including the command in the digital audio distribution signal.

25. The method of claim 22, wherein the control data is associated with at least one of volume, loudness, equalization, balance, or muting.

FIG. 1

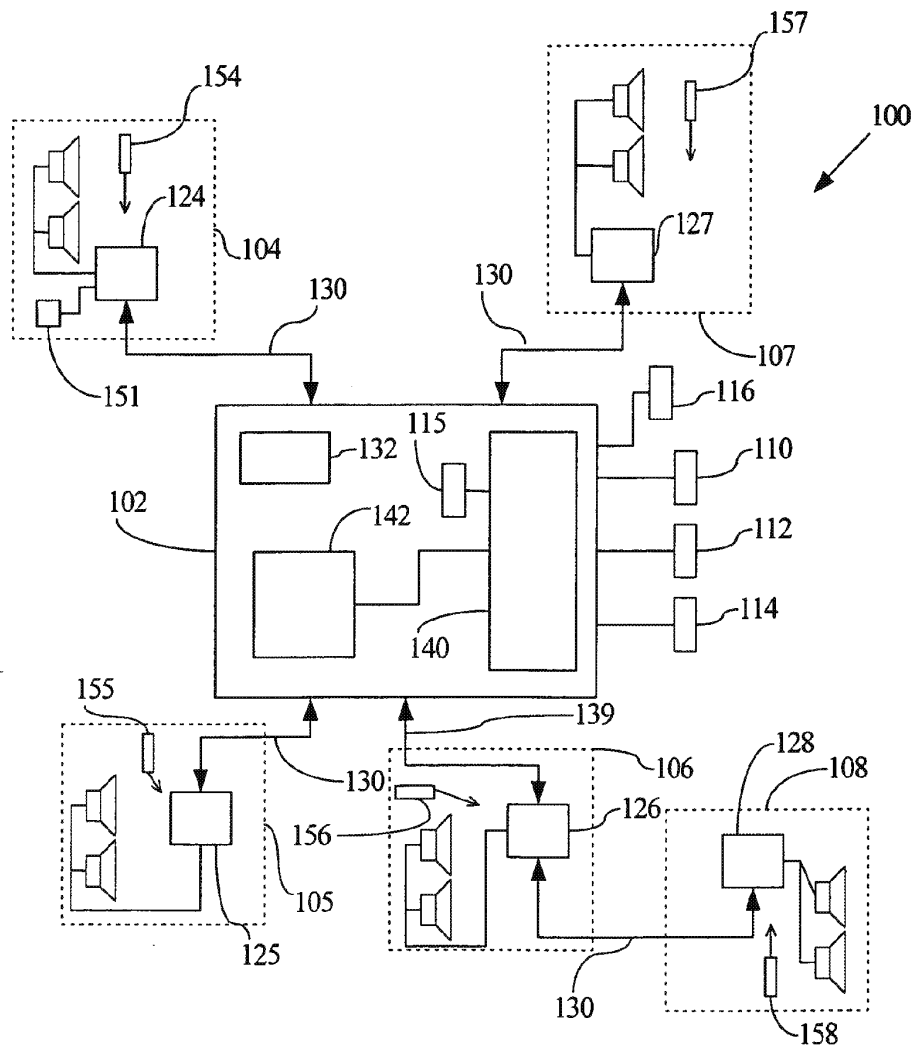


FIG. 2A

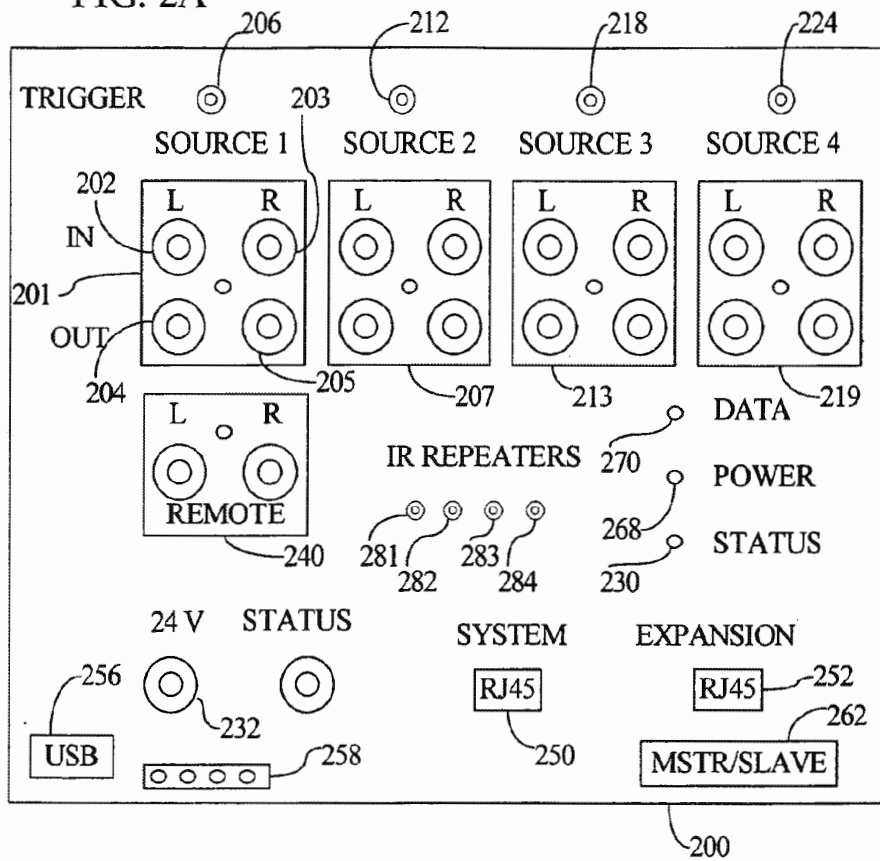
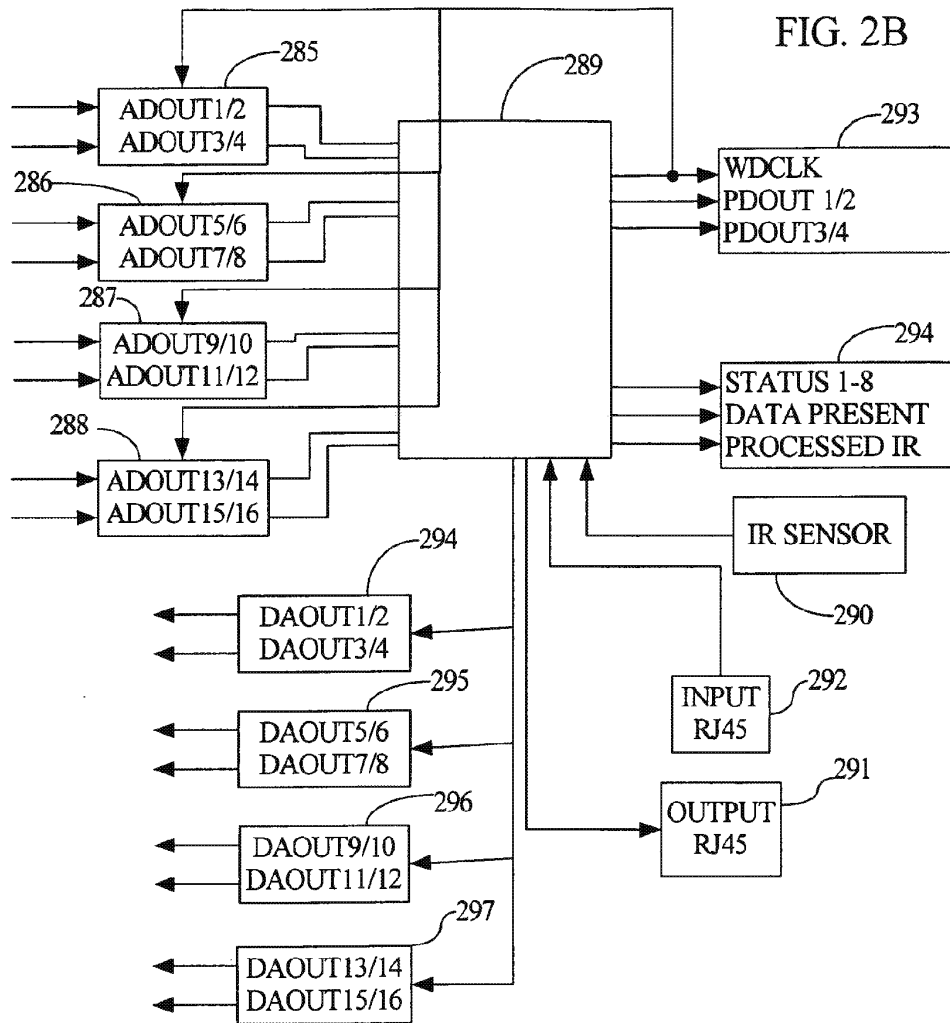
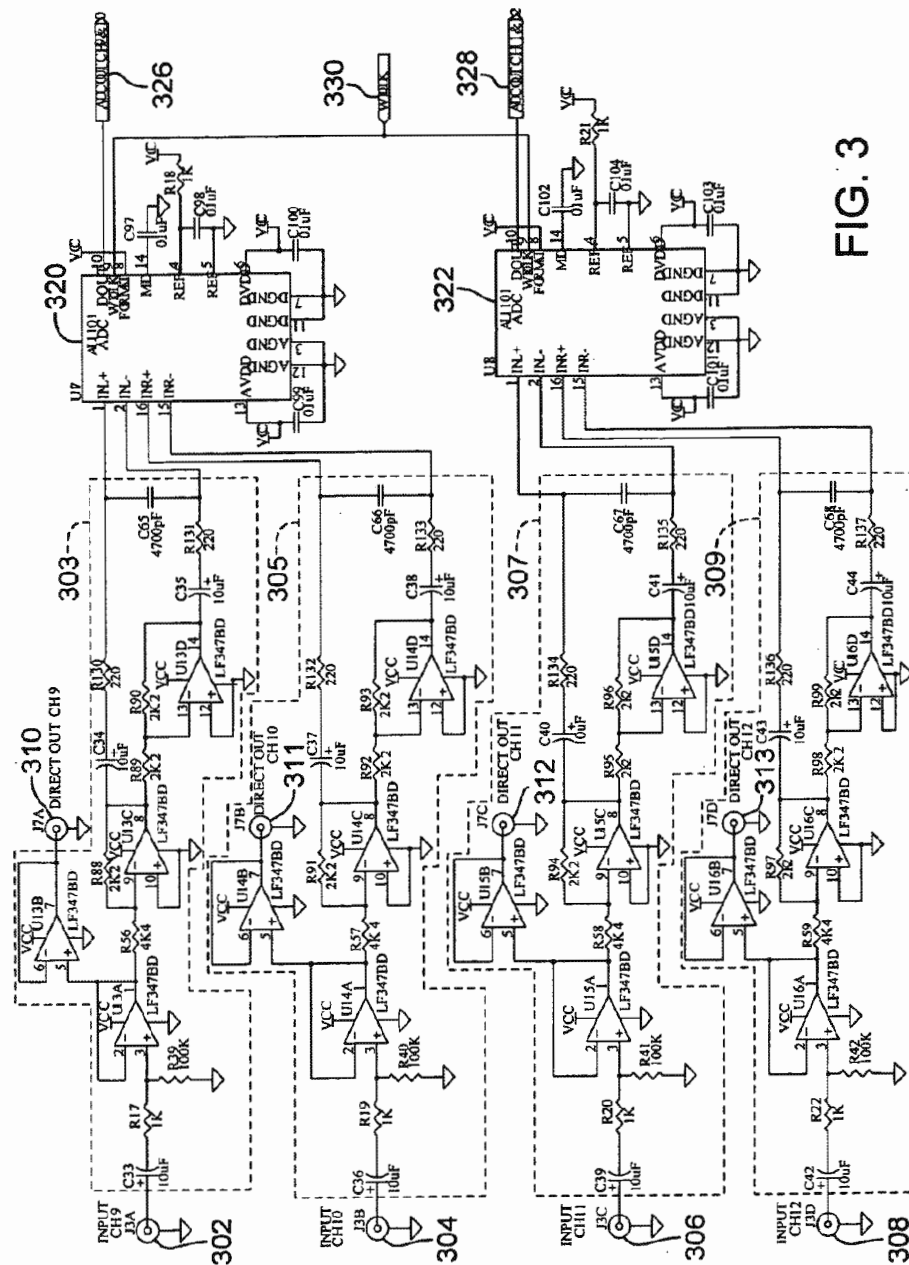


FIG. 2B





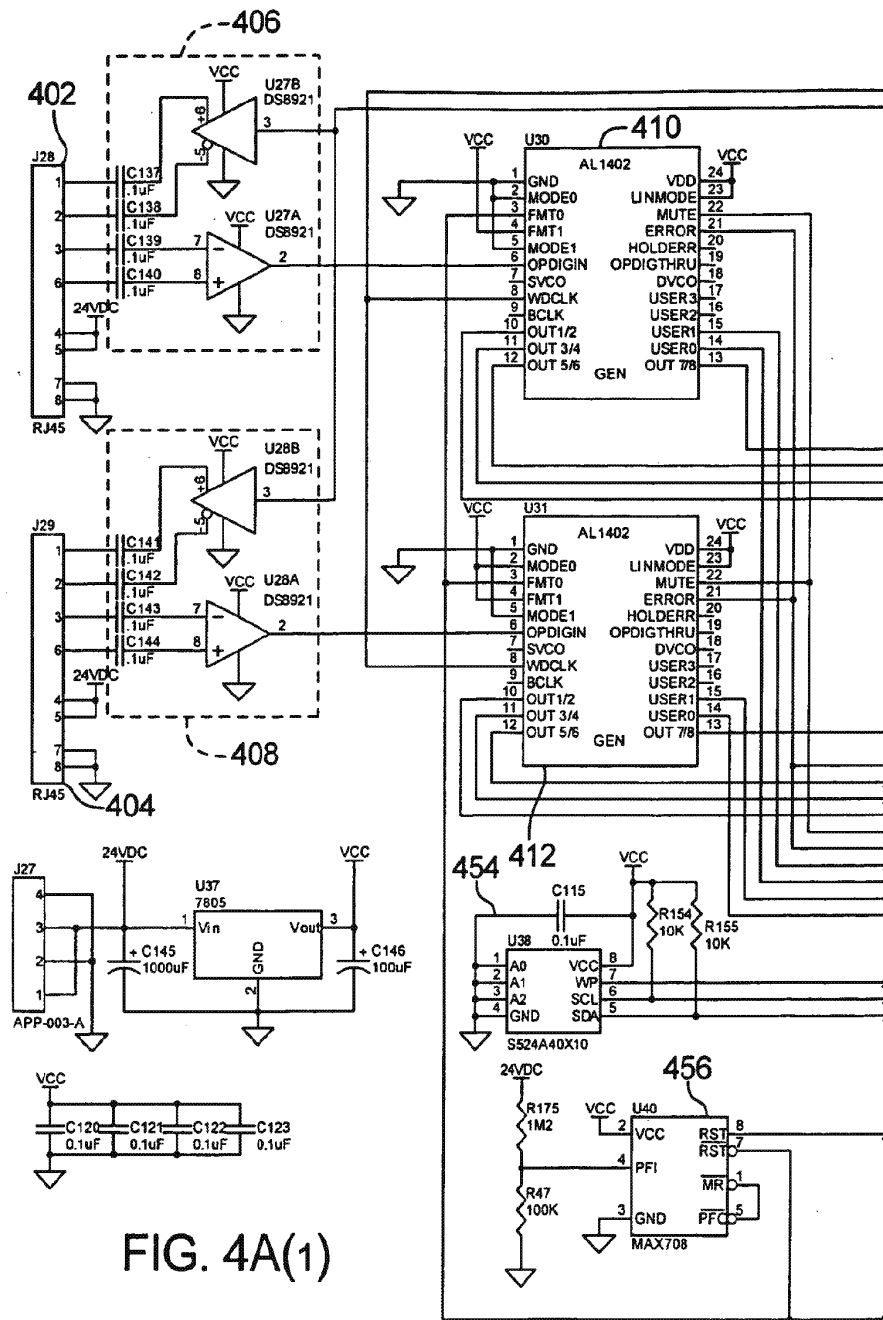


FIG. 4A(1)

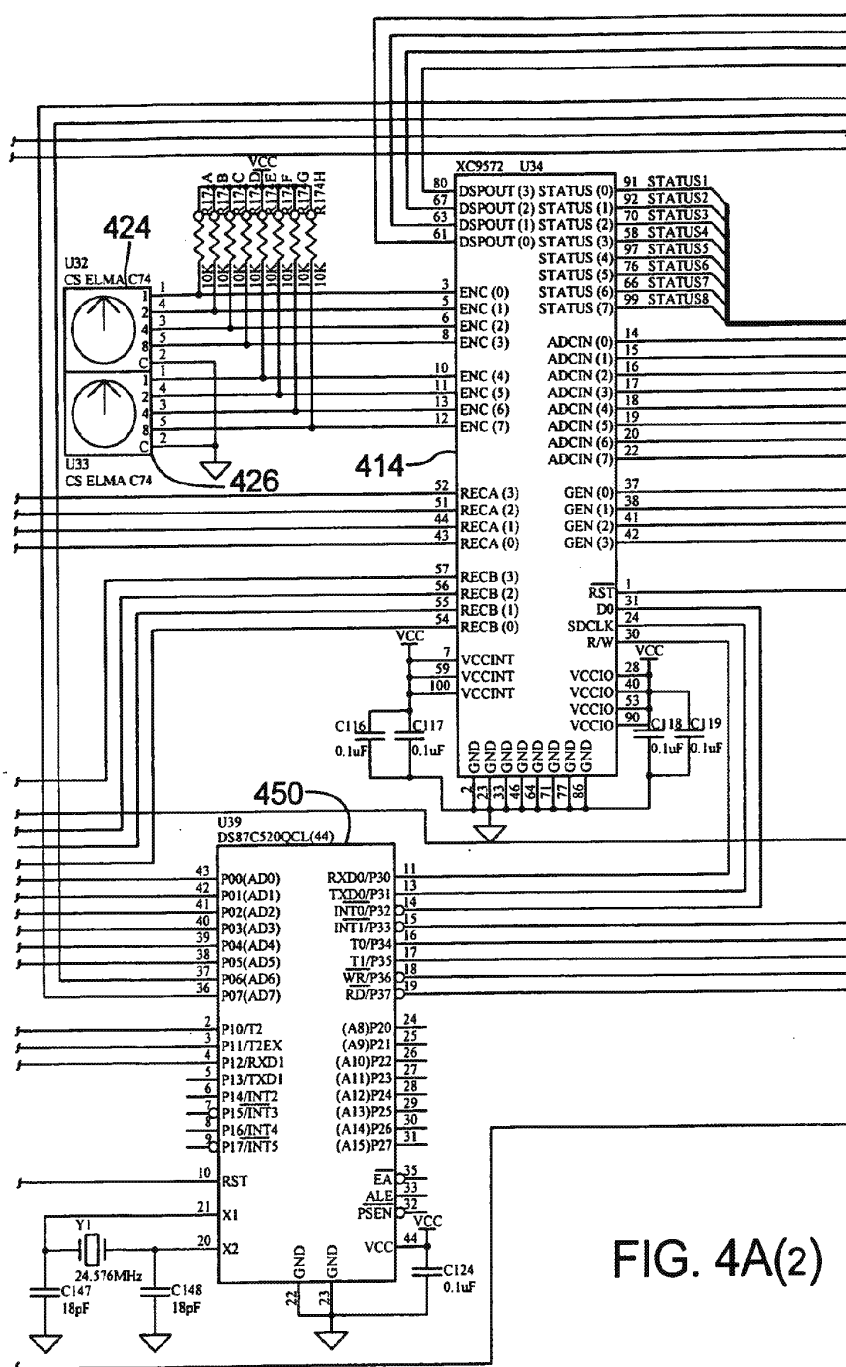


FIG. 4A(2)

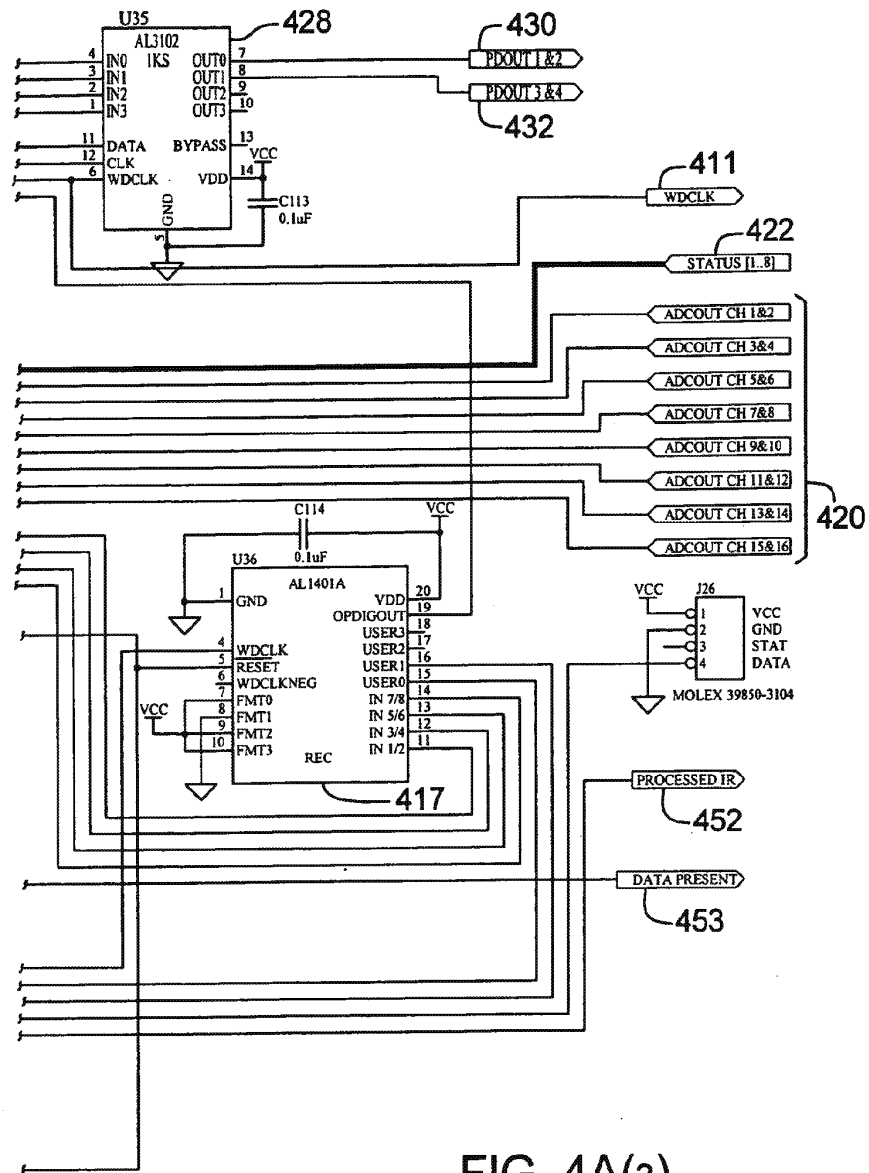


FIG. 4A(3)

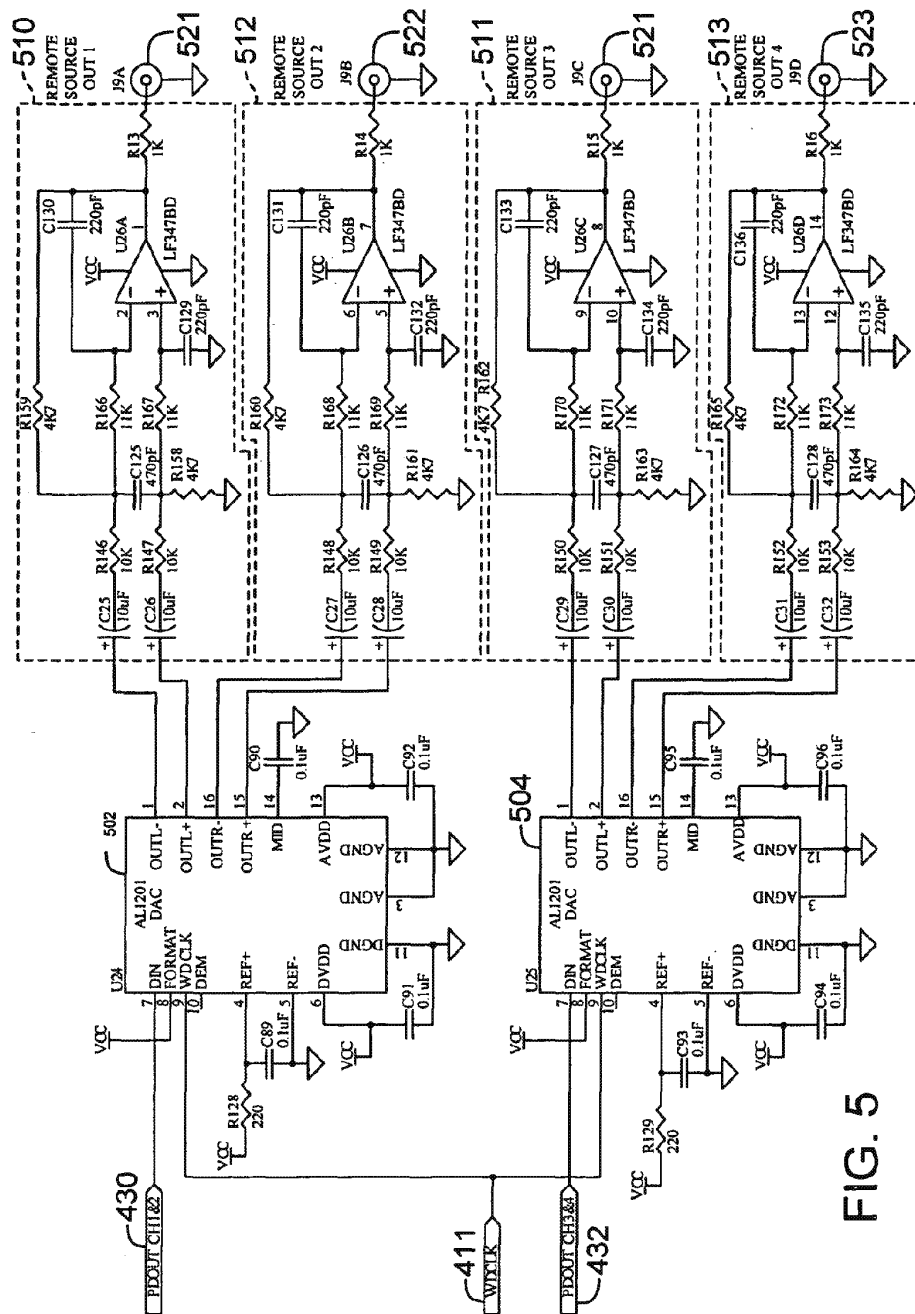


FIG. 5

FIG. 6A

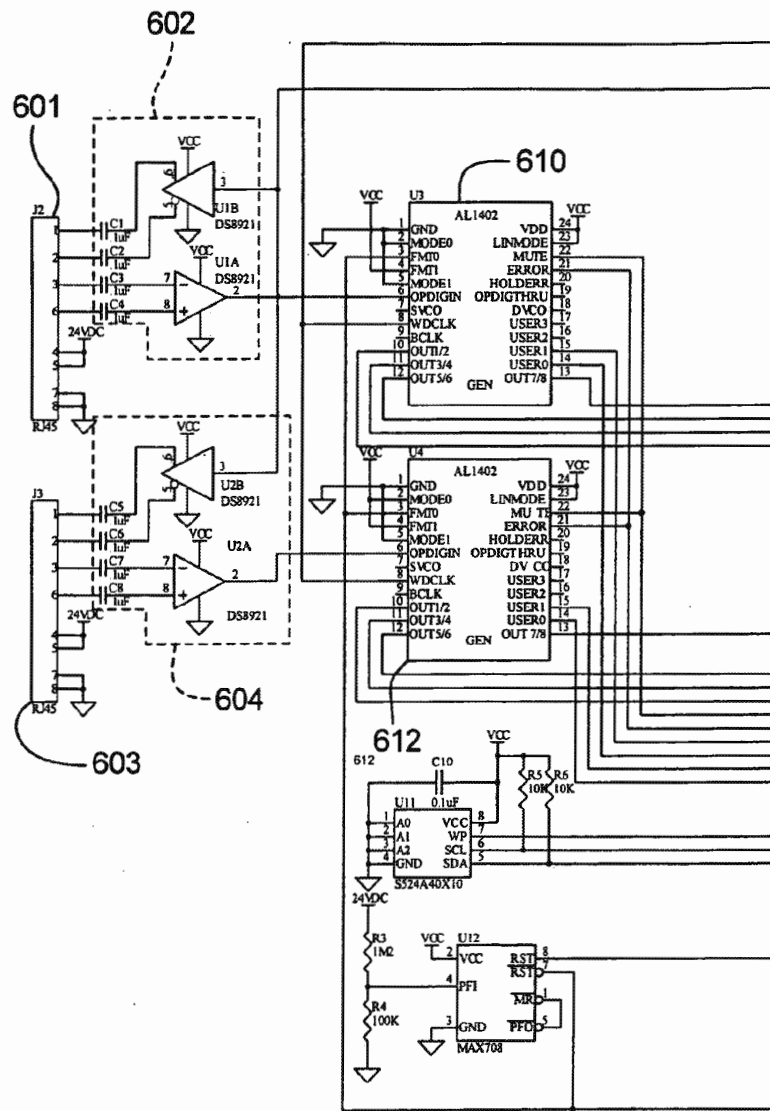


FIG. 6B

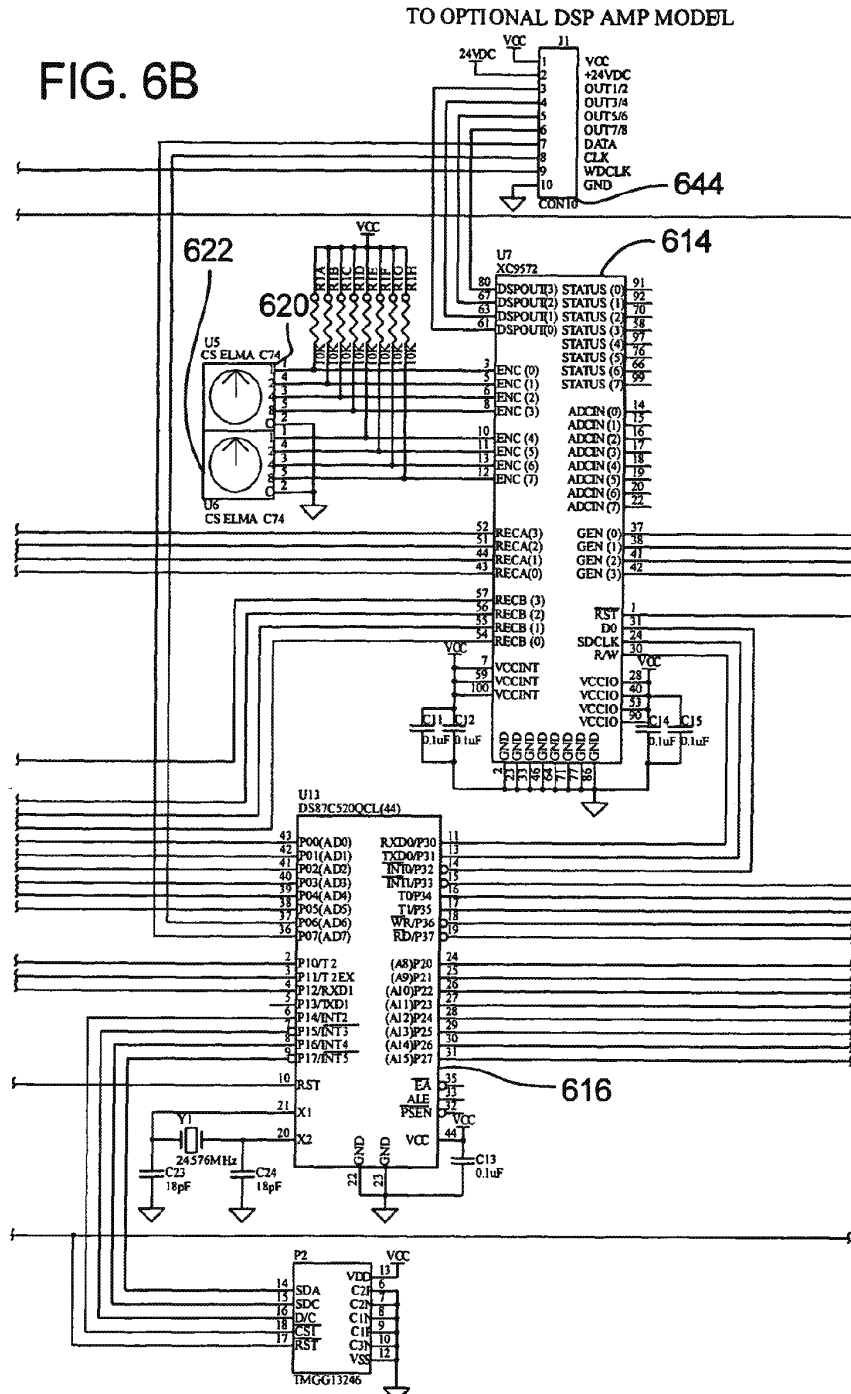
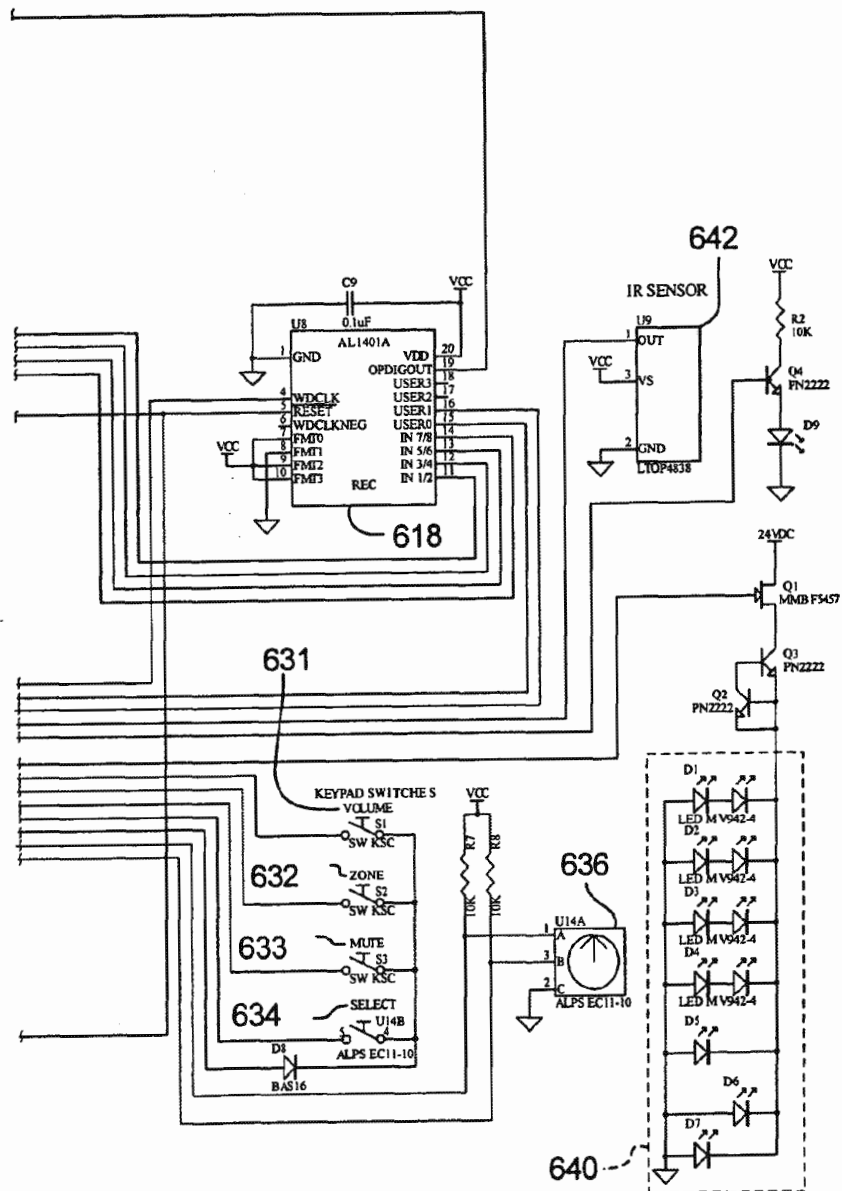
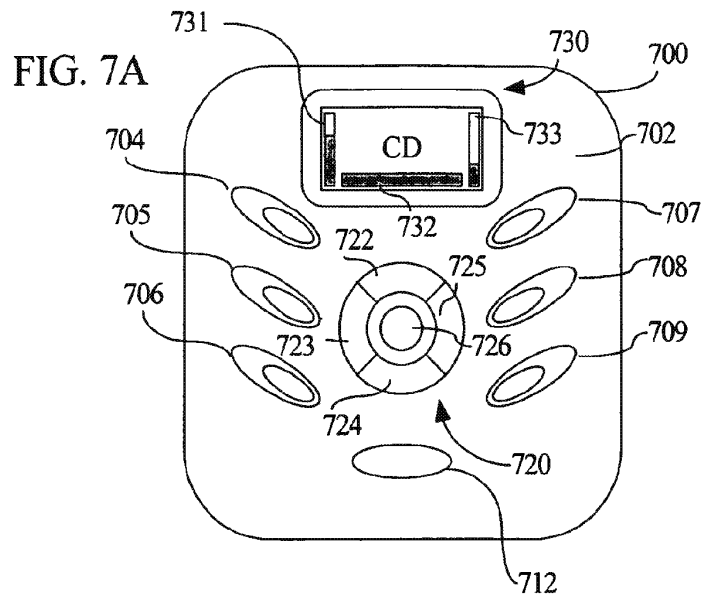
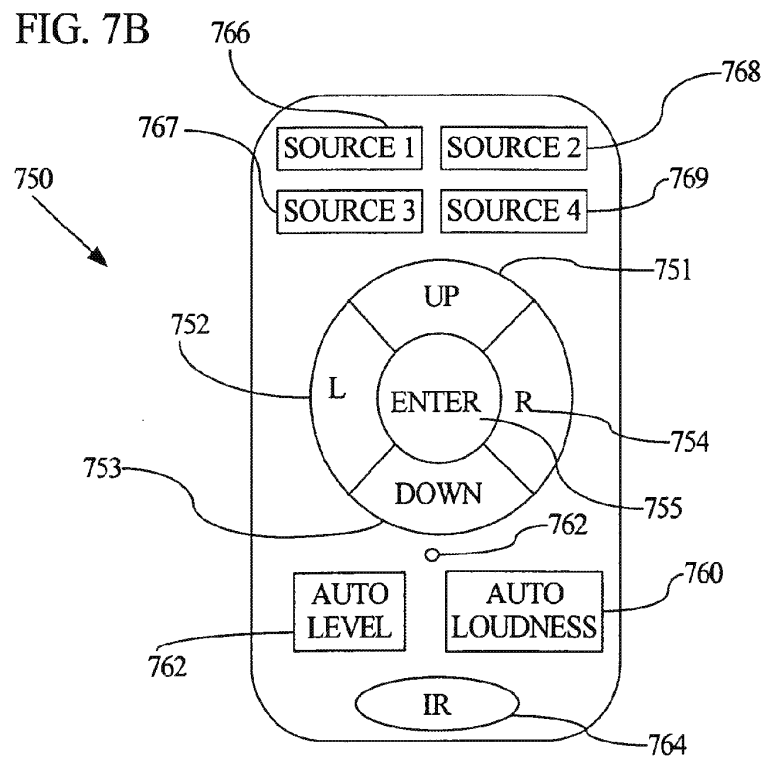
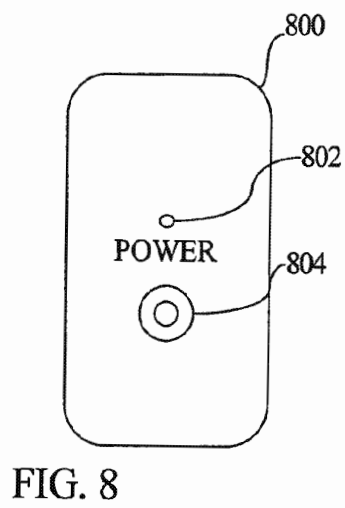
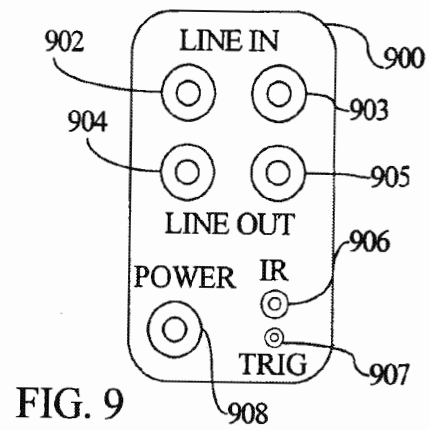


FIG. 6C







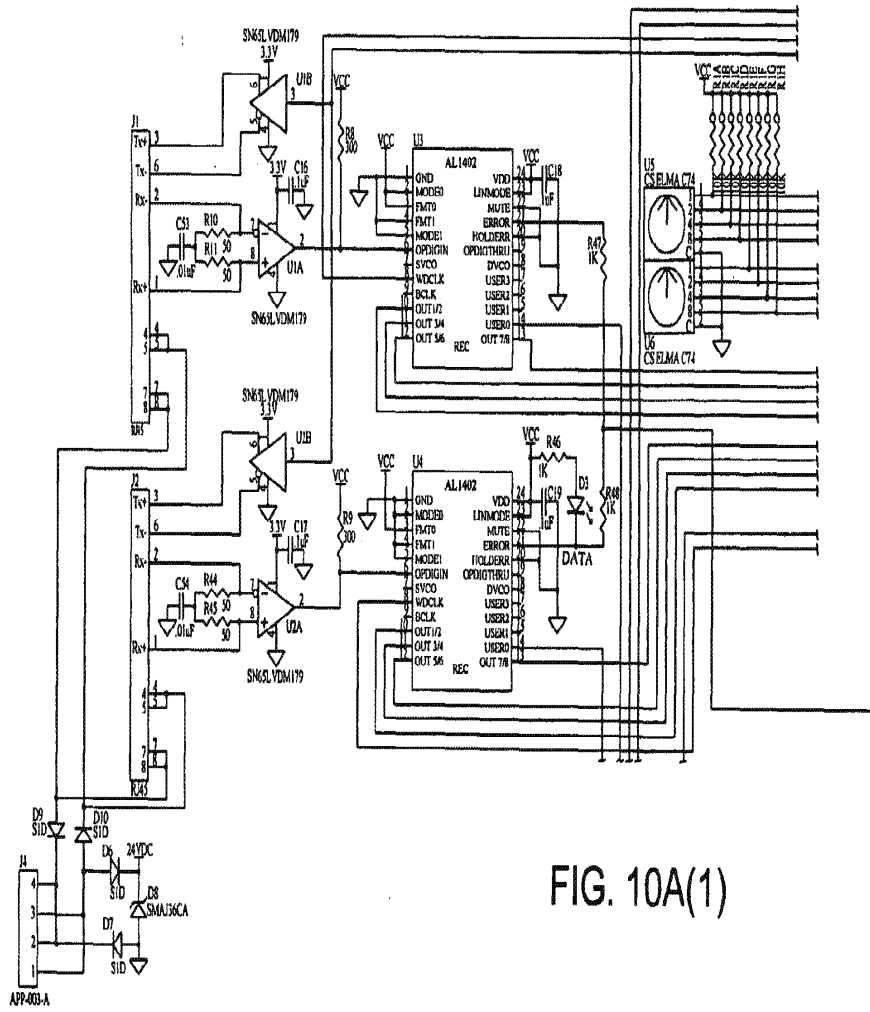
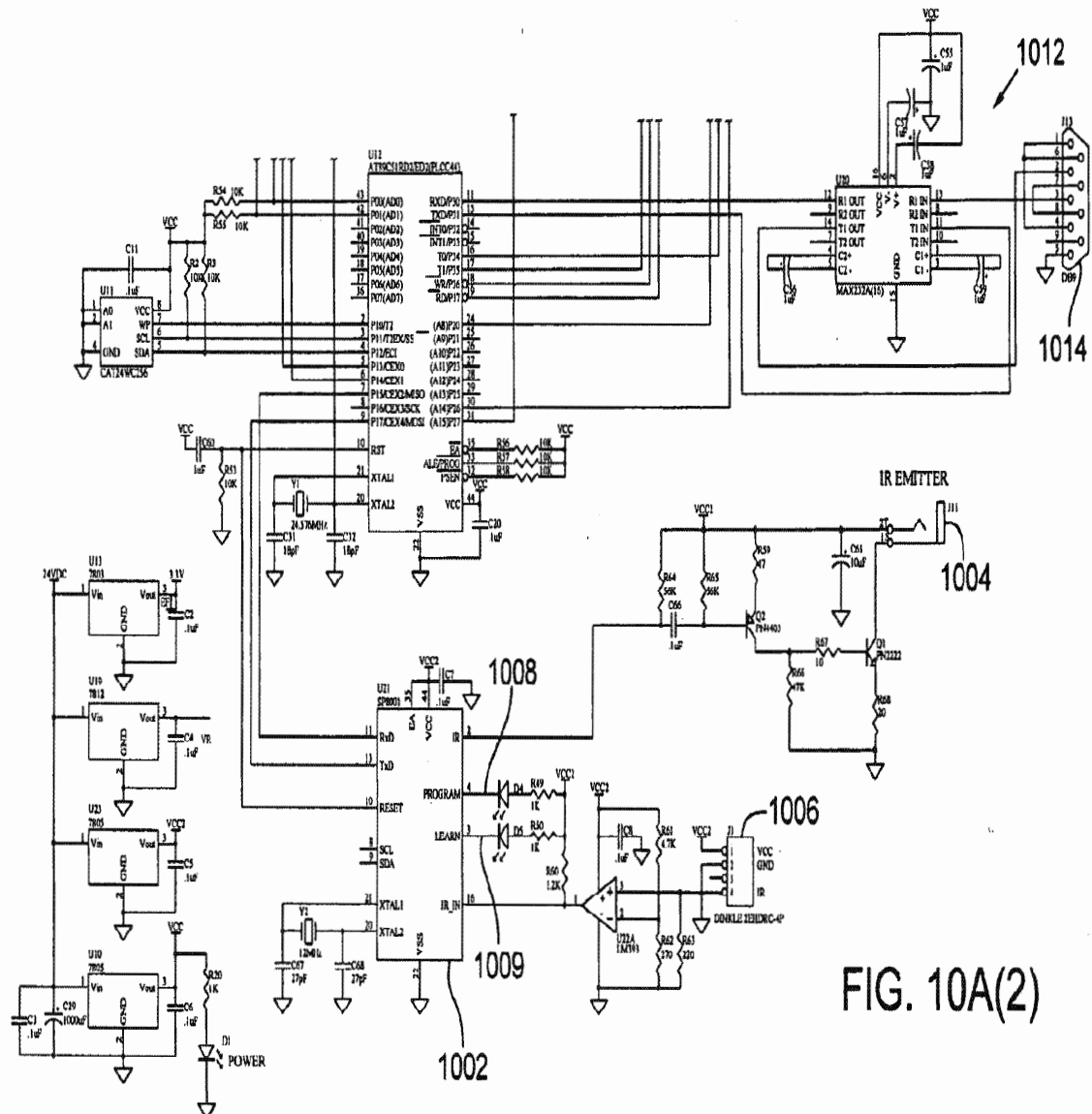


FIG. 10A(1)



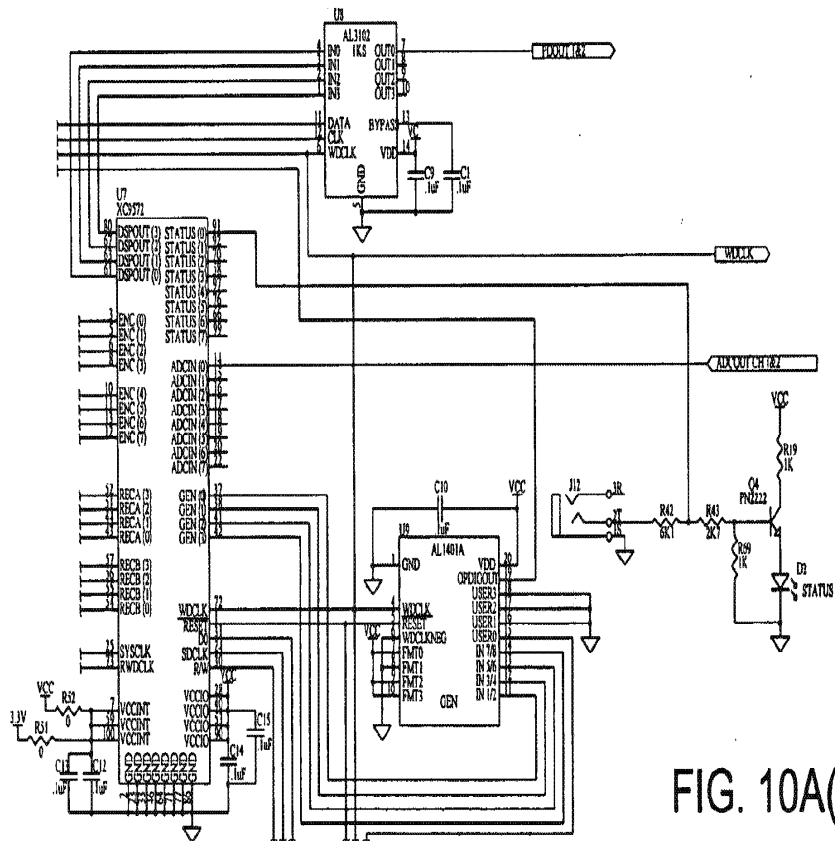
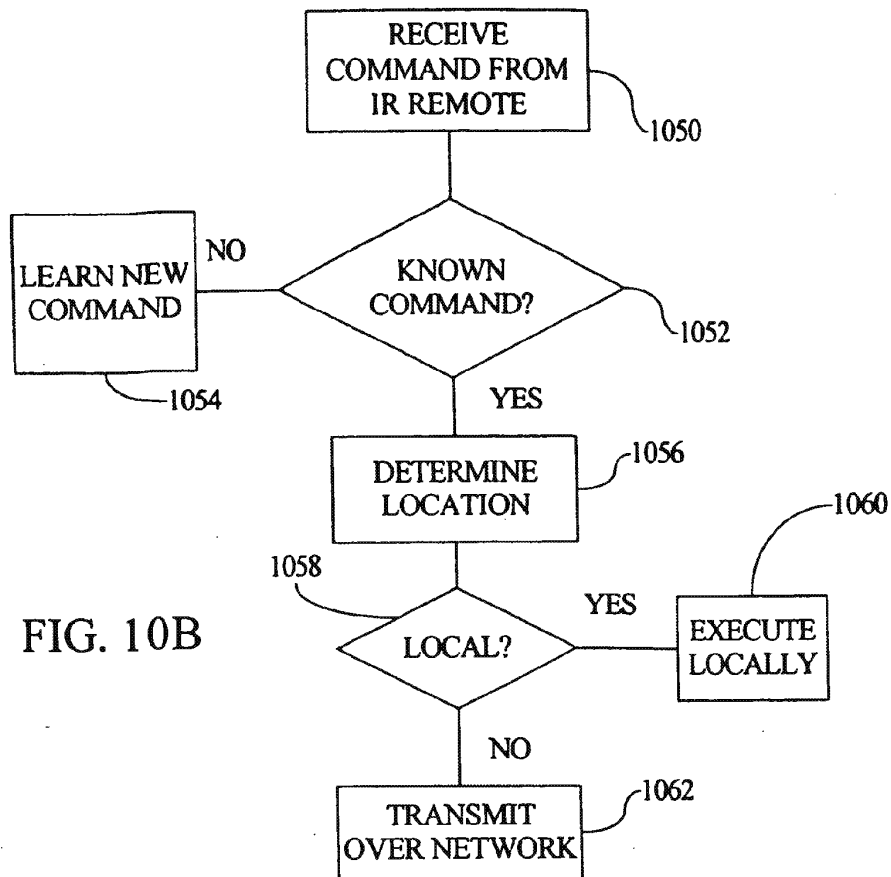
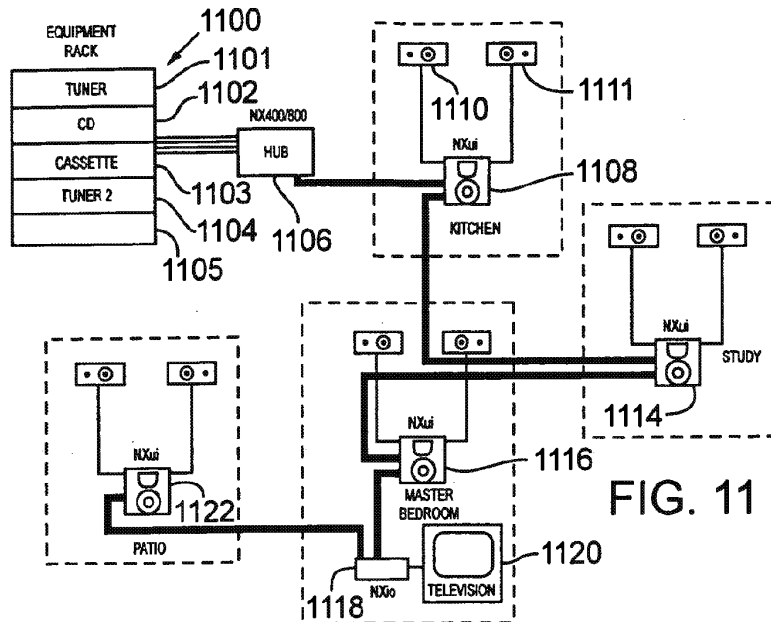
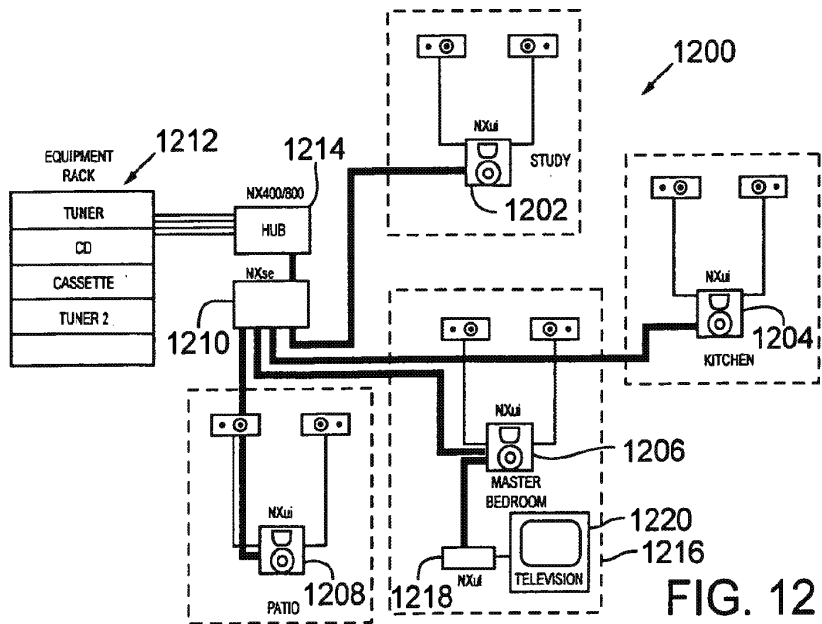
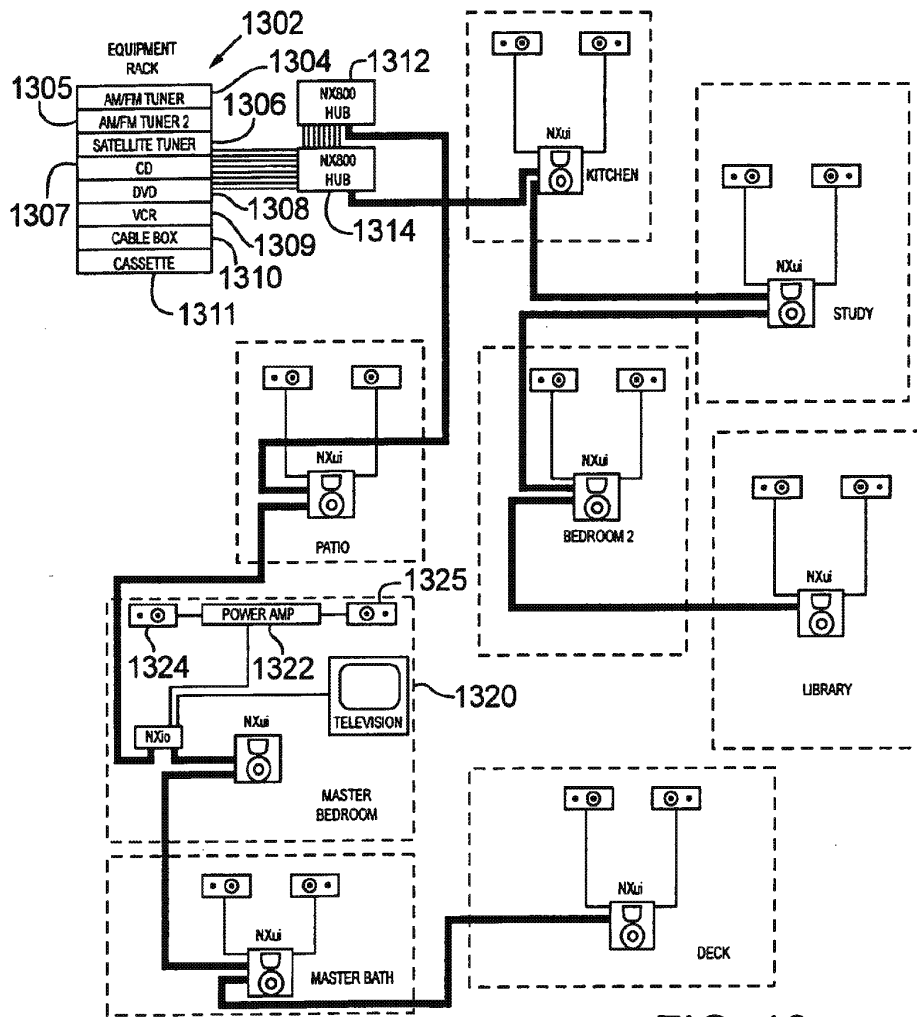


FIG. 10A(3)







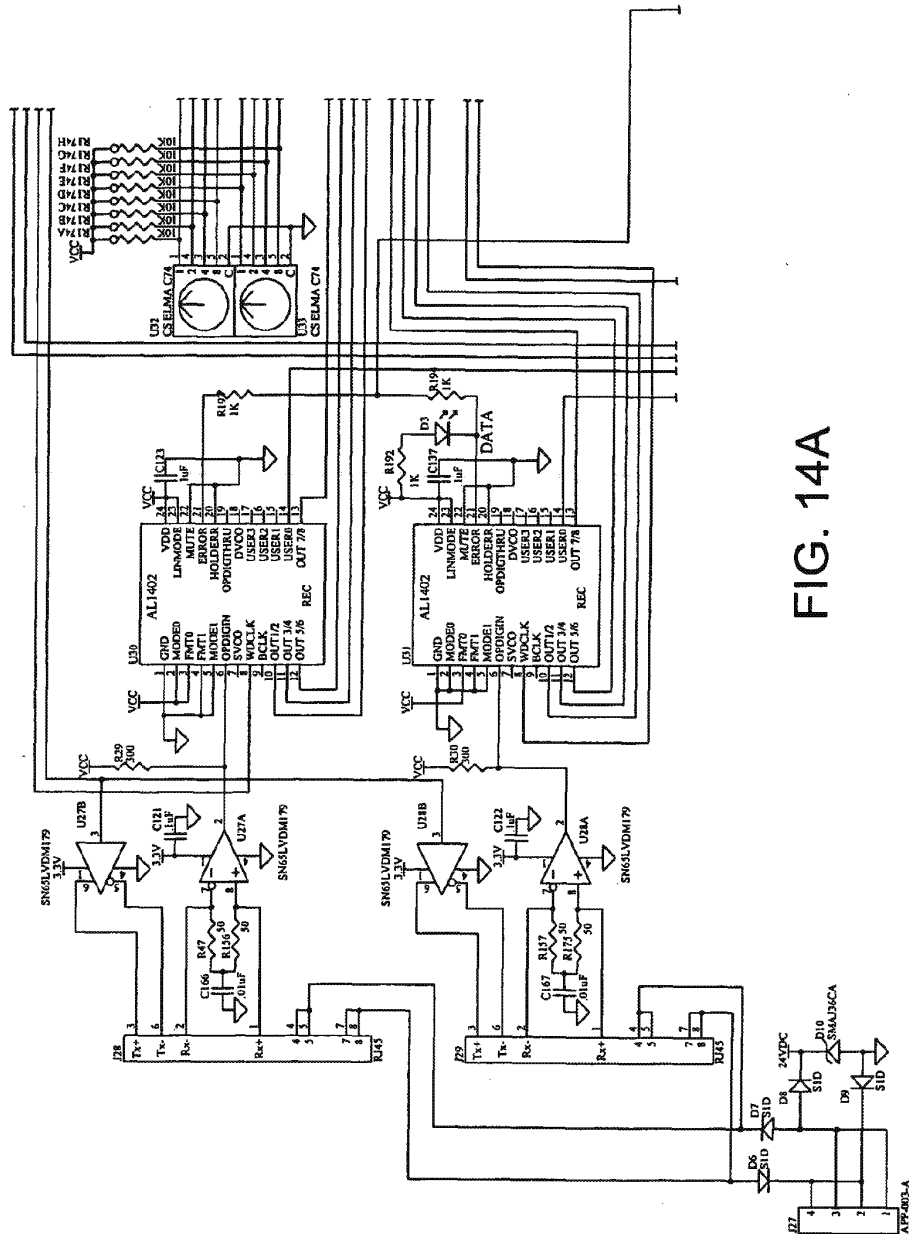


FIG. 14A

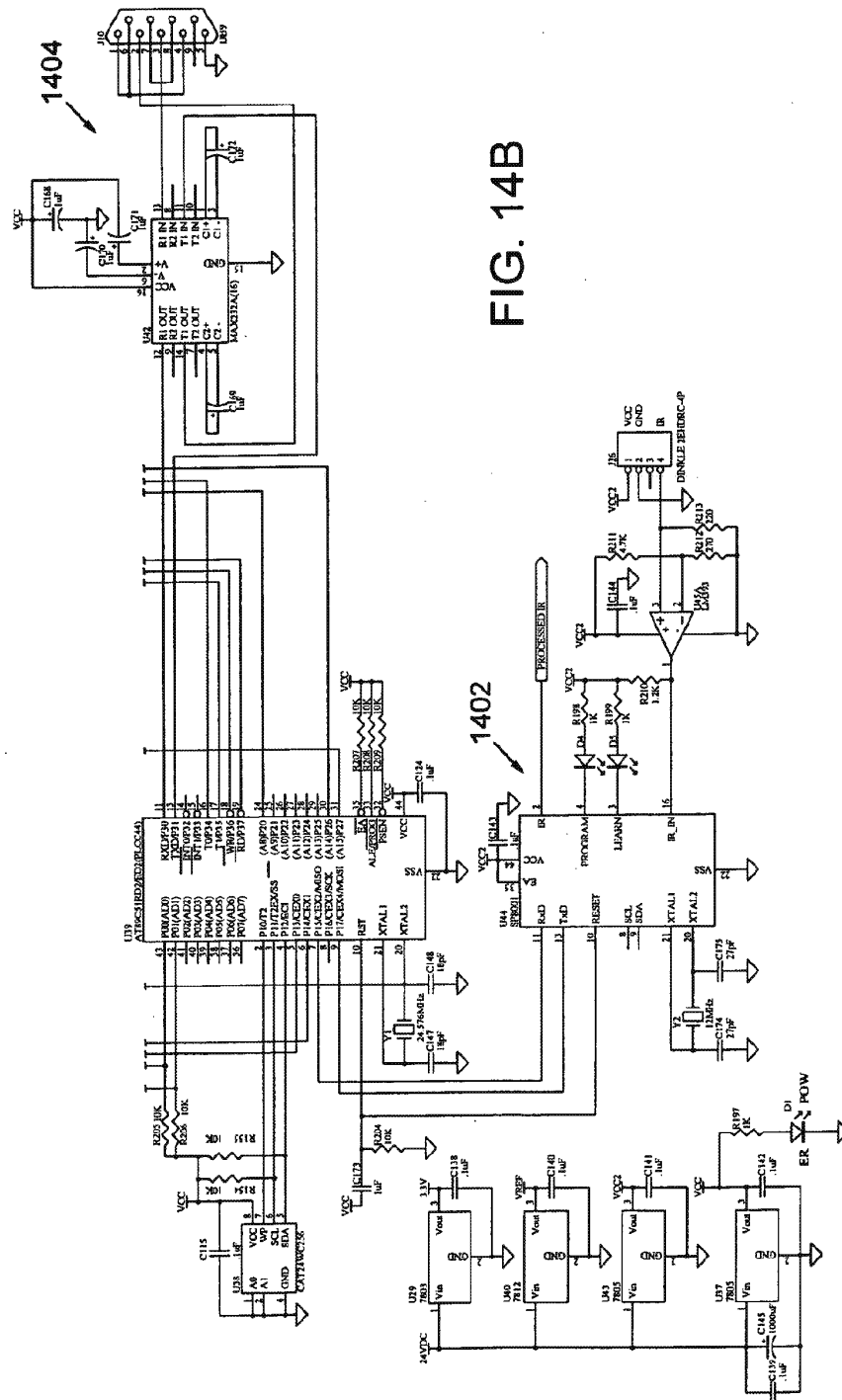


FIG. 14B

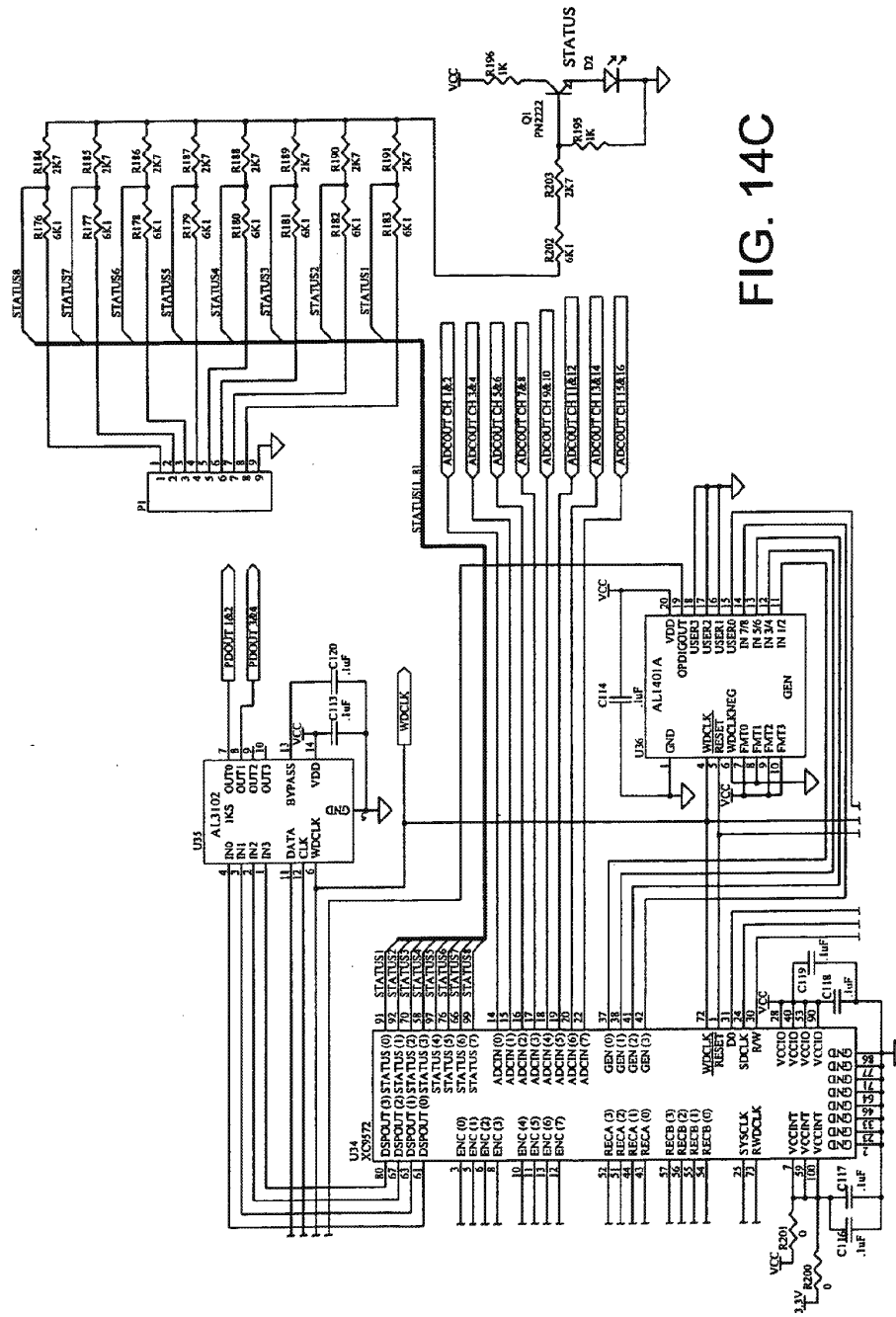


FIG. 14C

TO OPTIONAL DSP AMP MODULE

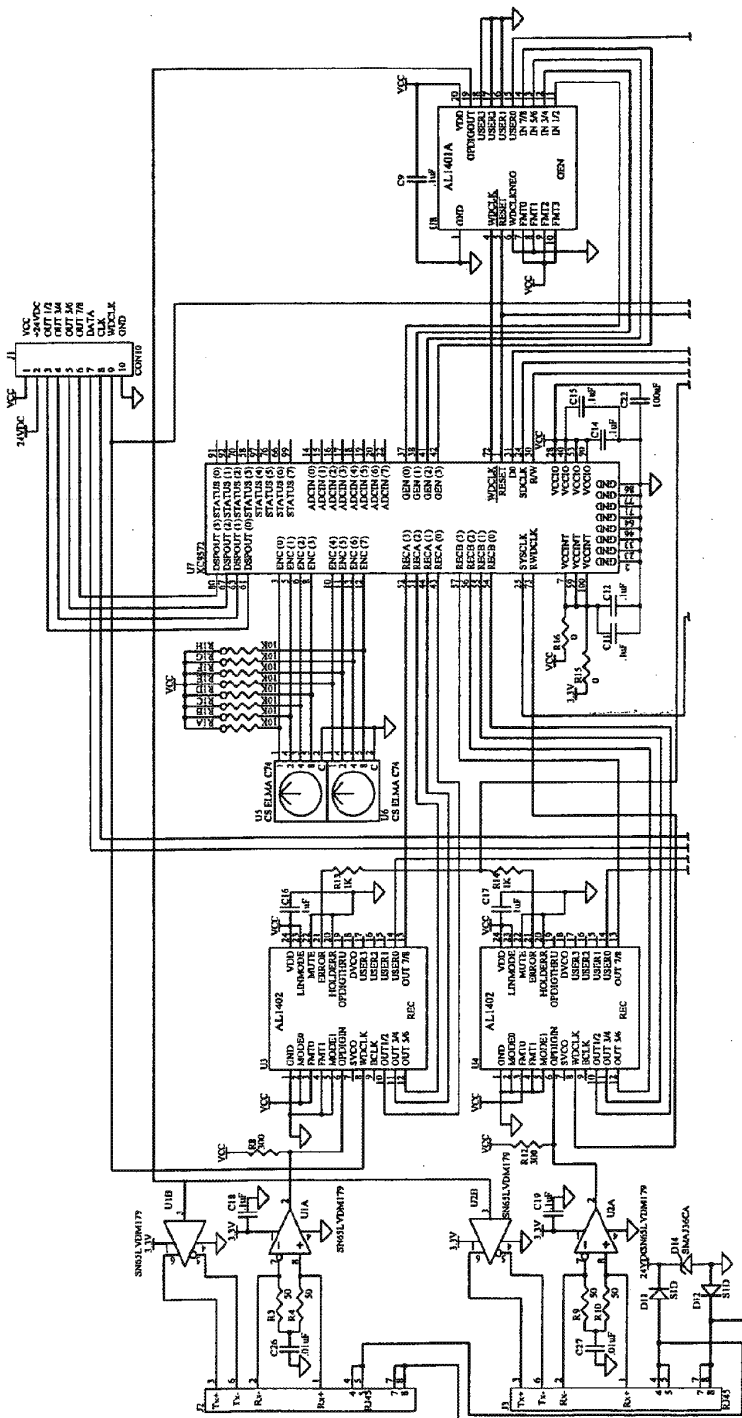


FIG. 15A

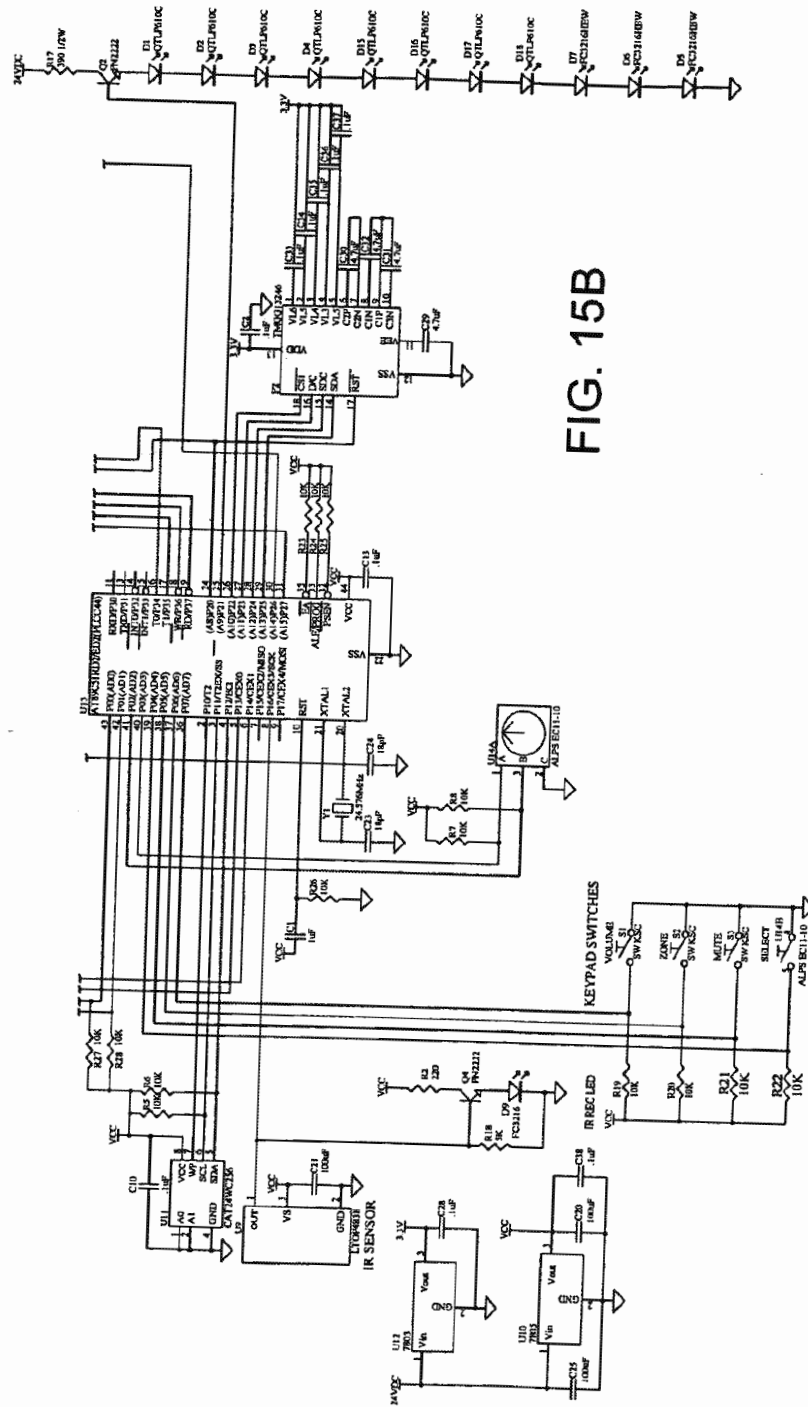


FIG. 15B

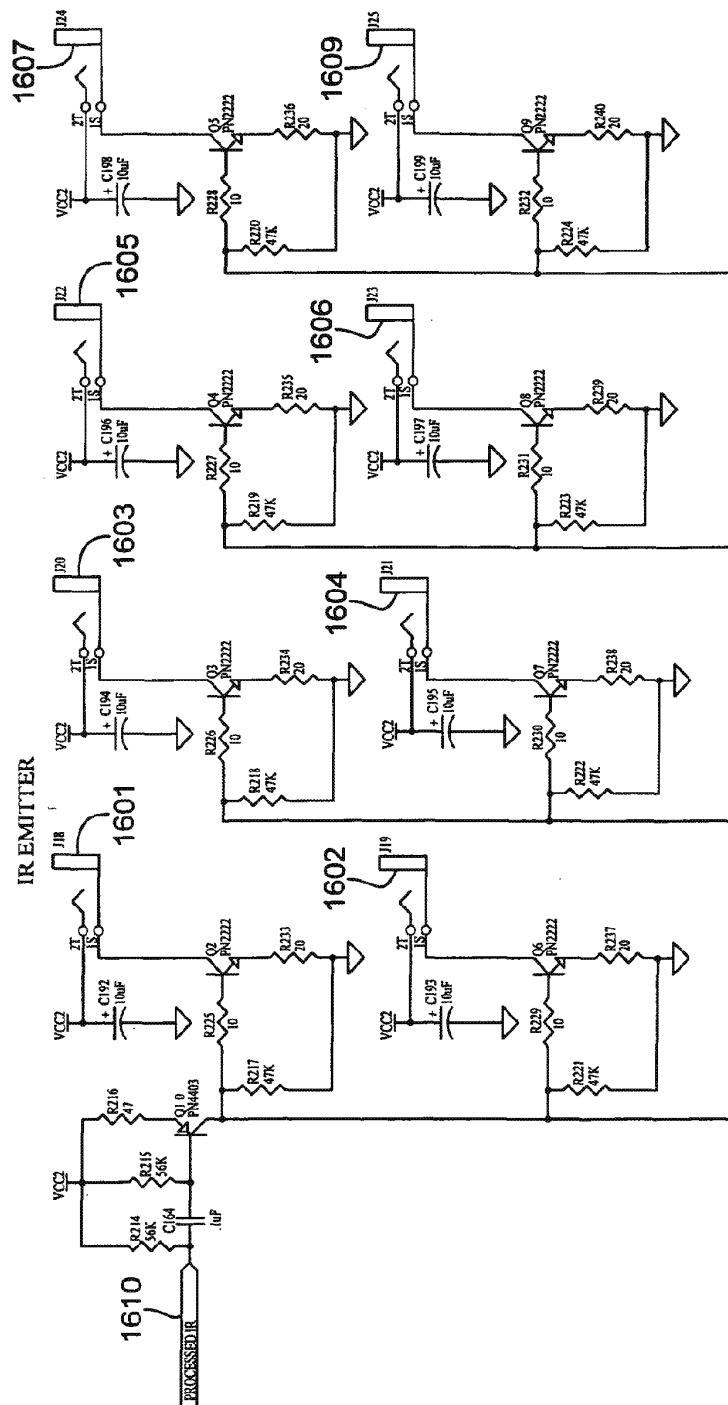


FIG. 16

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(71) Applicant(s)

Request Multimedia
(Incorporated in USA - New York)
435 2nd Avenue, Troy, New York 12182,
United States of America

(72) Inventor(s)

Harry Nick Carter
Ronald Cococcia
Zachary Piech
John Reine
Silvan Sauter
Steven Vasquez
Craig Willis
Hyung-Jun Brutus Youn

(51) INT CL⁷

G06F 17/30

(52) UK CL (Edition V)

G4A AMX

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[www.novell.com/news/press/archive/2001/03/
pr01021.html](http://www.novell.com/news/press/archive/2001/03/pr01021.html)

(58) Field of Search

UK CL (Edition T) G4A

INT CL⁷ G06F

Other: Online: JAPIO, EPODOC, WPI, Internet

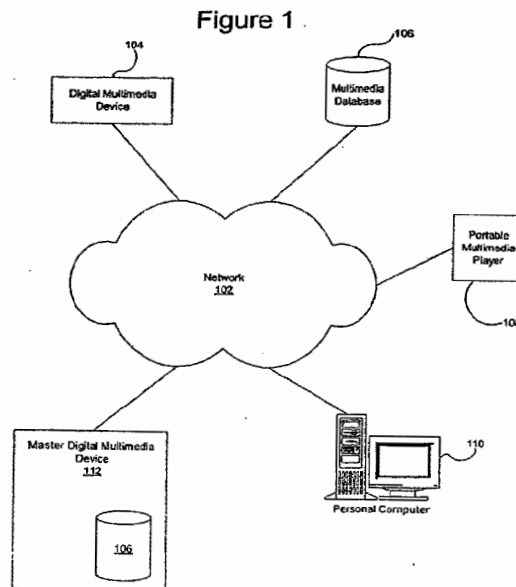
(74) Agent and/or Address for Service

Forrester Ketley & Co
Forrester House, 52 Bounds Green Road,
LONDON, N11 2EY, United Kingdom

(54) Abstract Title

Multimedia synchronisation method and system

(57) A system for synchronising networked devices in a multimedia environment has at least one central storage and interface device 106, and a plurality of zones each having a zone-specific storage and interface device 104, 108, 110, 112. Audio, video or photographic information, relating to at least one user and contained within the central device and each of the zone-specific devices, is updated and synchronised among the devices. This enables the user to be situated in any one of the zones and to access the same information. The synchronisation method includes determining whether a synchronisation point already exists and, if it does, then receiving information from a server device; or, if it doesn't, then sending information to at least one client device, the user determining what information is needed by the client device. The resultant state is established as a synchronisation point.



At least one drawing originally filed was informal and the print reproduced here is taken from a later filed formal copy.

This print takes account of replacement documents submitted after the date of filing to enable the application to comply with the formal requirements of the Patents Rules 1995

GB 2 379 533 A

Figure 1

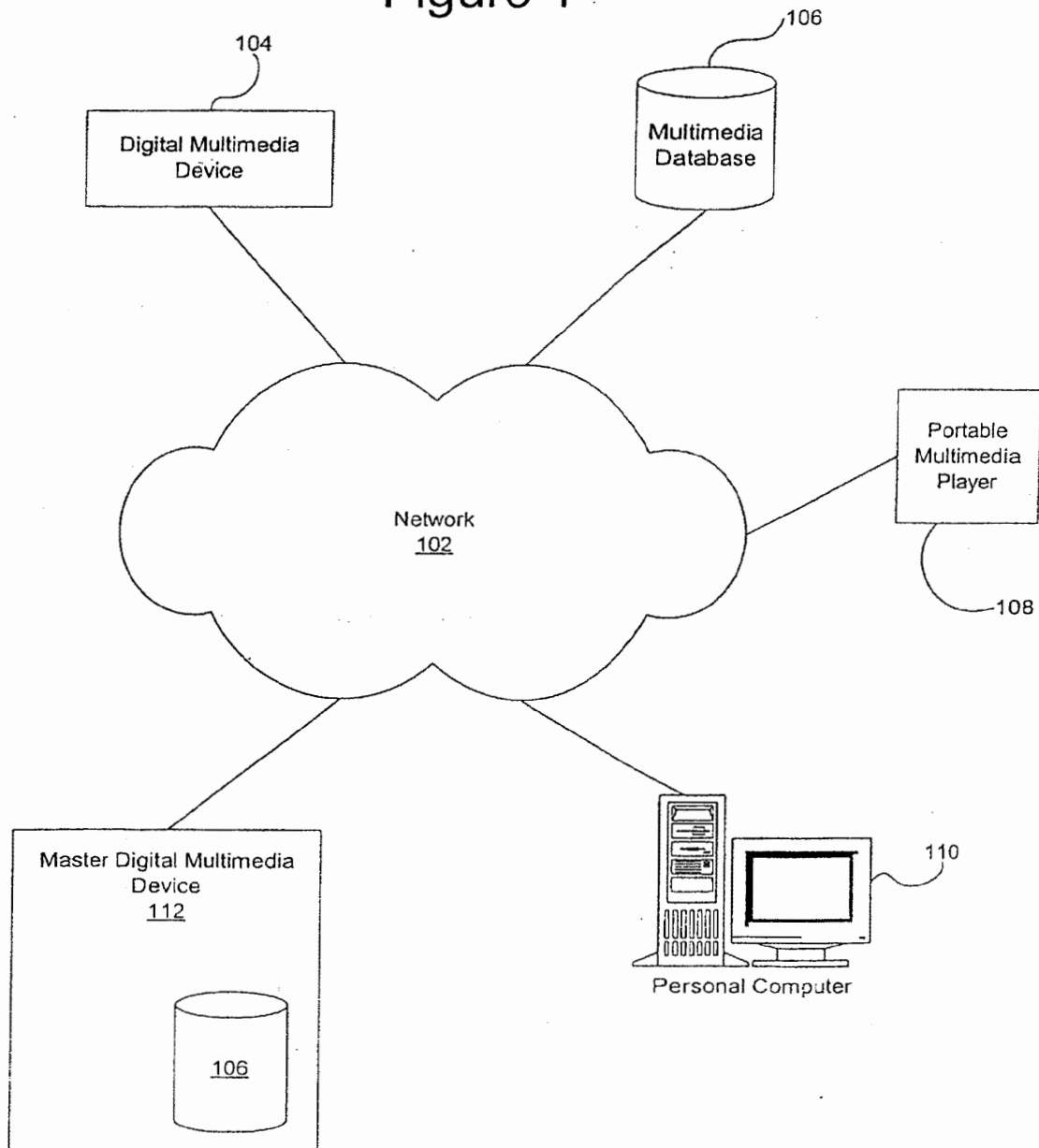


Figure 2

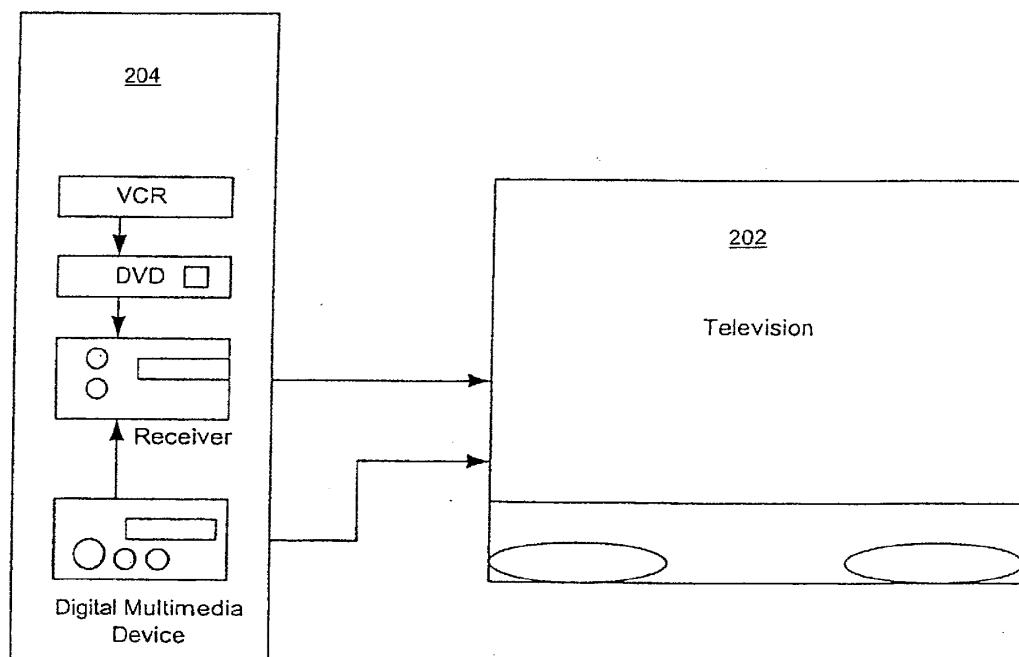


Figure 3

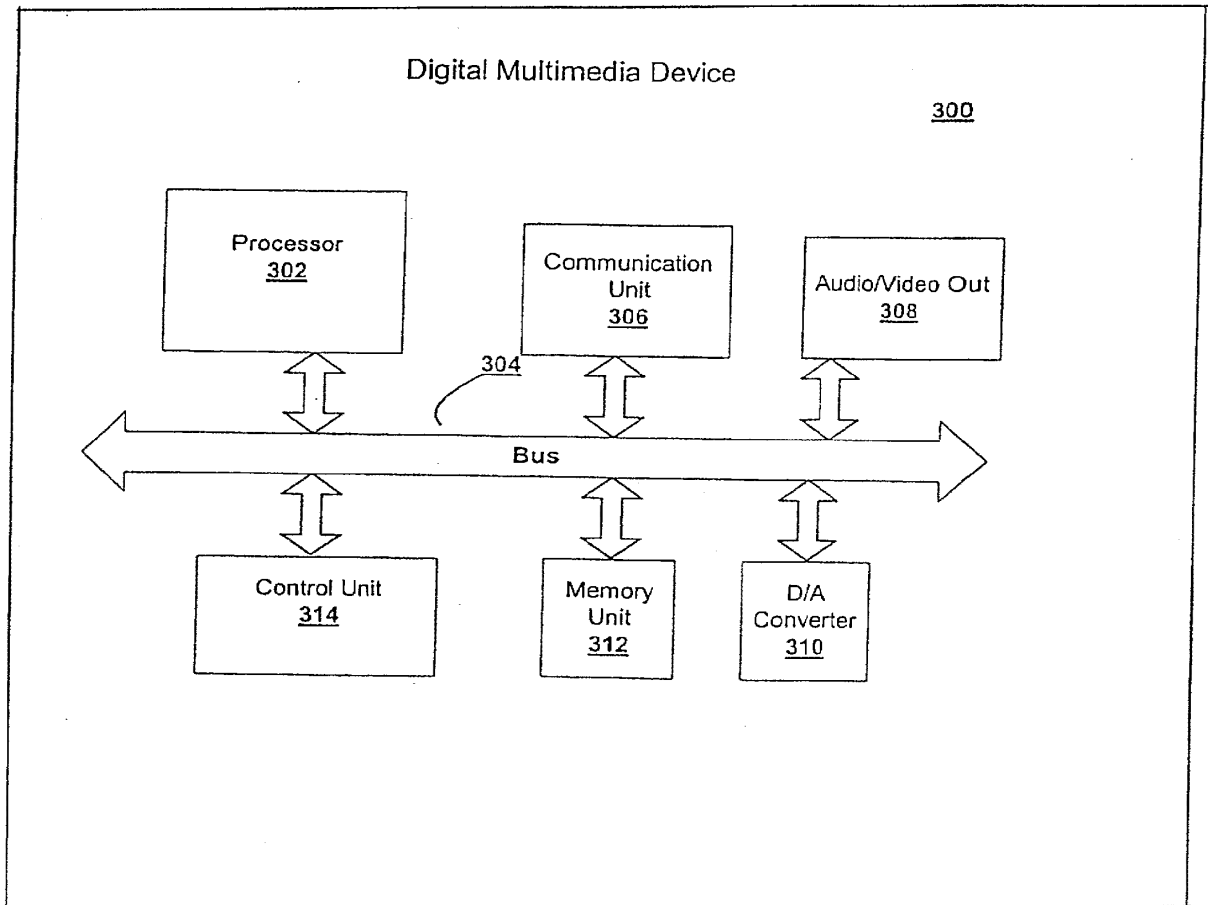


Figure 4

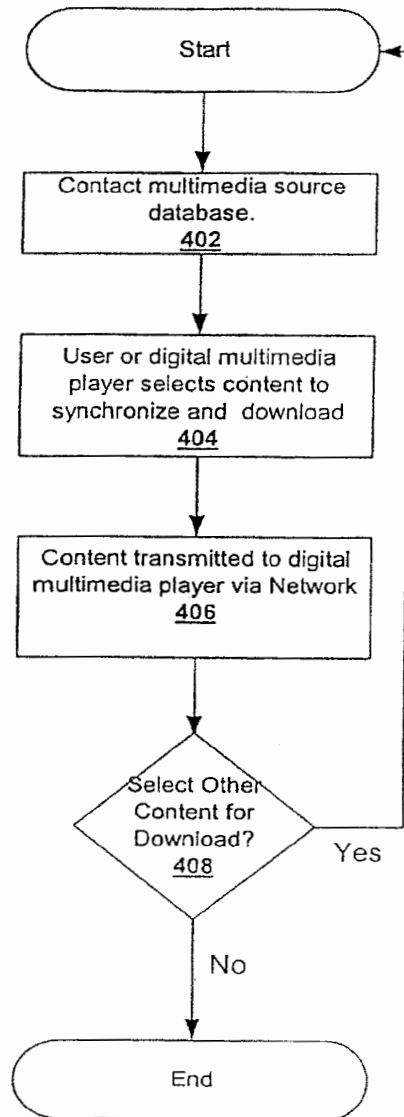
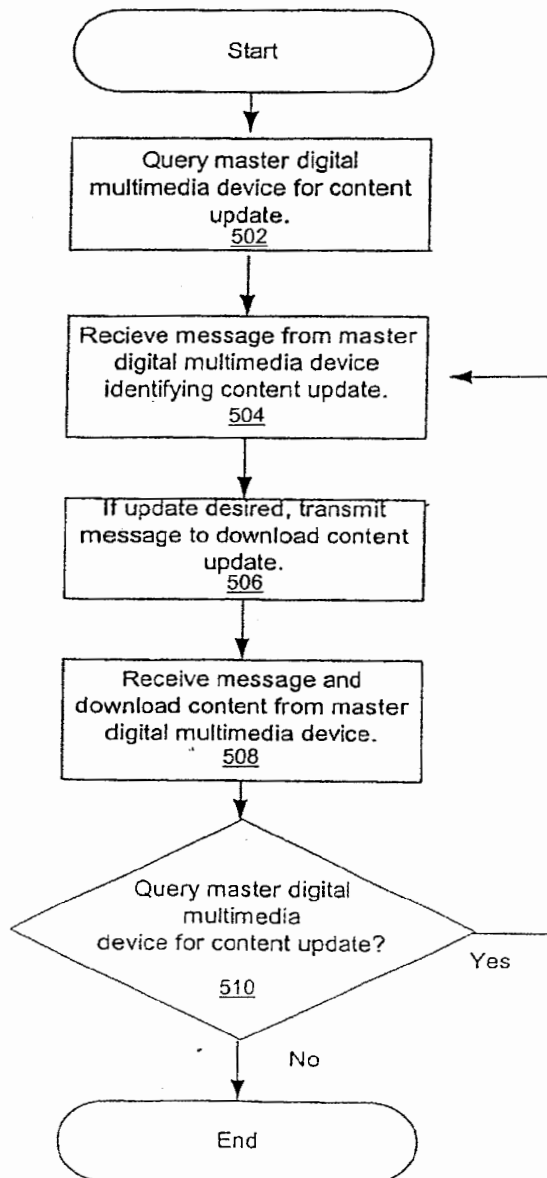


Figure 5



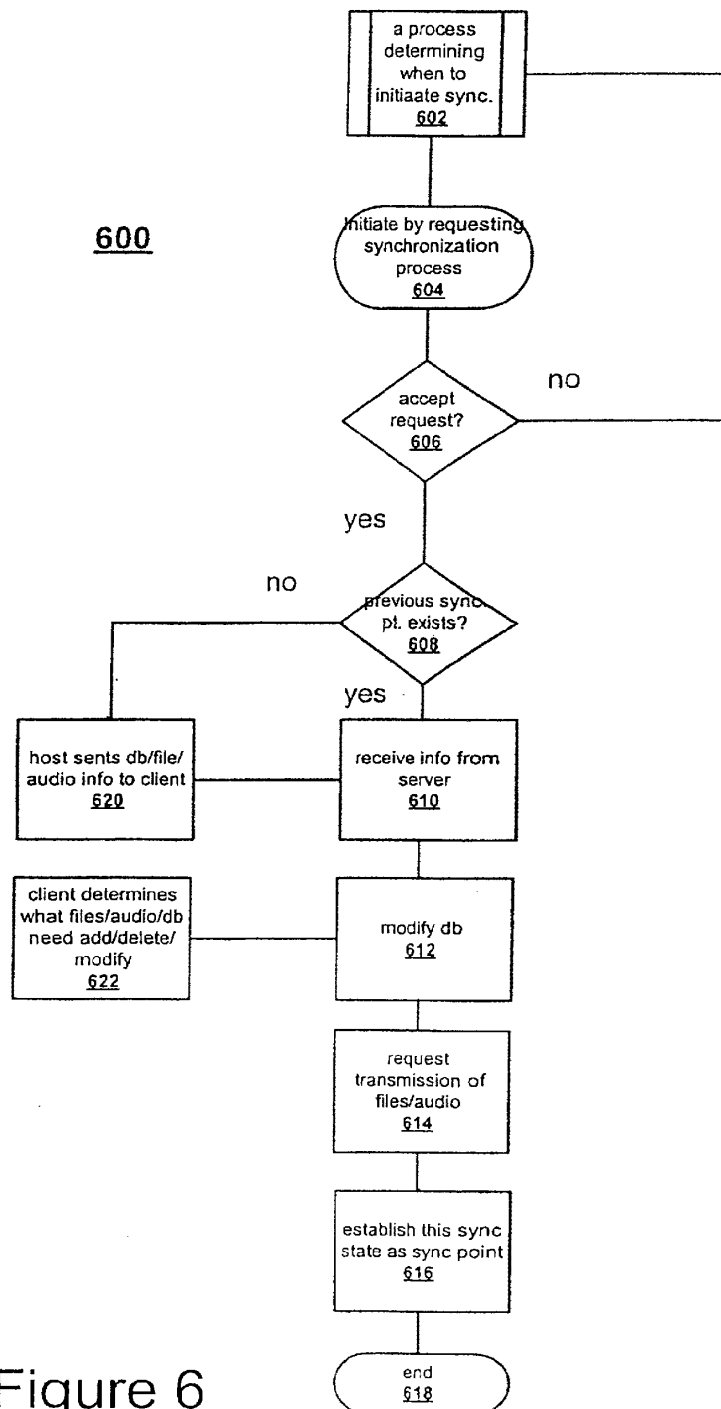


Figure 6

717

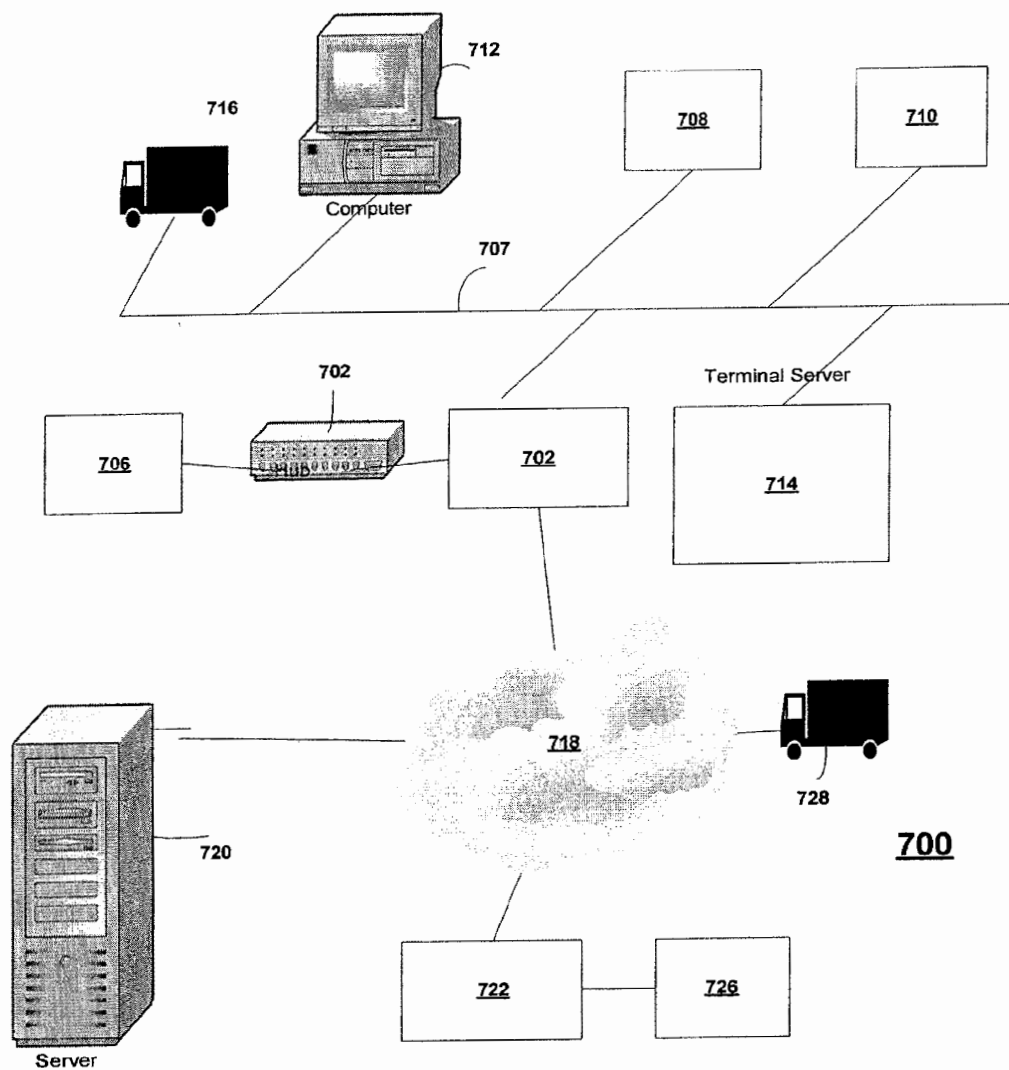


Figure 7

DESCRIPTION OF INVENTION

**“IMPROVEMENTS IN OR RELATING TO A MULTI-MEDIA
SYNCHRONISATION METHOD AND DEVICE”**

1 **BACKGROUND OF THE INVENTION**

2 This invention relates generally to the field of multimedia record and playback
3 devices, and more particularly to such devices which incorporate digitally stored multimedia
4 data on a stationary or removable memory and playback device which can convert the stored
5 data into audible sound and visible video images.

6 Historically, radio and television have been the major audio and video content
7 providers to the general population. Both radio and television provide the listener/viewer
8 with a selection of programming directed at different preferences and tastes. Unfortunately,
9 listener/viewers have very little, if any, input into the broadcast formats selected by the radio
10 stations and television networks. As such, the listener/viewer's optional programming choices
11 are limited to the number and different formats played by local radio stations and national
12 television networks.

13 Because of these and other limitations of music and video broadcasting, cassette
14 players, compact disc (CD) players, video cassette recorders (VCR) and digital video disks
15 (DVD) have become very popular. These devices allow the listener/viewer to select and
16 control the type and frequency of music and video content they desire to listen to or watch at
17 any given moment. However, several drawbacks to these devices include that the
18 listener/viewer must individually purchase the cassette tapes, CDs, videotapes or DVDs
19 which also have limited storage capacity. Likewise, the listener/viewer must transport a large
20 number of cassettes, CDs, videotapes or DVDs to provide a range of musical and video
21 selections.

22 U.S. Patent Number 6,161,132, entitled System For Synchronizing Playback Of
23 Recordings And Display By Networked Computer Systems, discloses entertainment content

1 complementary to a musical recording being delivered to a user's computer by means of a
2 computer network link. The user employs a browser to access the computer network. A plug-
3 in for the browser is able to control an audio CD or other device for playing the musical
4 recording. A script stored on the remote computer accessed over the network is downloaded.
5 The script synchronizes the delivery of the complementary entertainment content with the
6 play of the musical recording. However, this patent does not teach transferring of
7 information related to a particular user.

8 U.S. Patent Number 5,355,302, entitled System For Managing A Plurality Of
9 Computer Jukeboxes, discloses a method and apparatus for managing a plurality of computer
10 jukeboxes at different locations from a central station. Each jukebox includes processor
11 means for controlling the computer jukebox, storage and retrieval means for data, display
12 means for selection menus, audio production means for playing musical records, and a user
13 interface enabling patrons to communicate with the processor means. However, substantially
14 identical information related to a particular user are not synchronized and stored in all the
15 zones in this patent.

16 U.S. Patent Number 6,055,566, entitled Customizable Media Player With
17 Online/Offline Capabilities, discloses an information dissemination system comprises an
18 Internet-connected server adapted for gathering information from plural sources, and sorting
19 the information according to subscriber preferences. The sorted information is transmitted via
20 the Internet to a subscriber's Internet Appliance (IA) as electronic documents, where the
21 documents are downloaded to a connected playback device. However, this patent does not
22 teach a local area network (LAN) for coupling a central storage and interface device with a
23 zone specific storage and interface device.

1 U.S. Patent Number 6,161,142, entitled Method And System For Using A
2 Communication Network To Supply Targeted Streaming Advertising In Interactive Media,
3 discloses a system and method for delivering programmed music and targeted advertising
4 messages to Internet based subscribers includes a software controlled microprocessor based
5 repository in which the dossiers of a plurality of the subscribers are stored and updated,
6 musical content and related advertising are classified and matched. A subscriber has an
7 appropriate microprocessor based device capable of selecting information and receiving
8 information from the Internet. The subscriber receives the programmed music and matched
9 advertisements from the repository over the Internet. However, this patent does not teach
10 user specific synchronization, or LAN based communication network.

11 U.S. Patent Number 6,199,076, entitled Audio Program Player Including A Dynamic
12 Program Selection Controller, discloses an audio program and message distribution system in
13 which a host system organizes and transmits program segments to client subscriber locations.
14 The host organizes the program segments by subject matter and creates scheduled
15 programming in accordance with preferences associated with each subscriber. Program
16 segments are associated with descriptive subject matter segments, and the subject matter
17 segments may be used to generate both text and audio cataloging presentations to enable the
18 user to more easily identify and select desirable programming. However, this patent does not
19 teach user specific information including substantially identical information at each zone
20 wherein the user can access the information without any further communication between the
21 zones. Similarly, U.S. Patent Number 5,926,624, entitled Digital Information Library And
22 Delivery System With Logic For Generating Files Targeted To The Playback Device, does
23 not teach the same either.

1 U.S. Patent Number 5,734,119, entitled Method For Streaming Transmission Of
2 Compressed Music, discloses an Internet high fidelity audio transmission and compression
3 protocol including a system for representing synthesized music in a relatively small file as
4 compared to digital recording. The protocol includes a method for streaming the transmission
5 of a music data file from a Server-Composer computer such that the music can begin being
6 played back as soon as the file begins to arrive at a Client-Player computer. The system
7 includes a graduated resolution improvement feature which allows the music to be recreated
8 exactly as originally composed as the necessary wavetable data is downloading in the
9 background and the music continues to play in the foreground. However, this patent does not
10 teach using LAN as a means for transfer of information.

11 U.S. Patent Number 5,721,815, entitled Media-On-Demand Communication System
12 And Method Employing Direct Access Storage Device, discloses a method of transmission in
13 discrete form between a vender and a user. However, this patent does not teach the user
14 accessing substantially identical information at a plurality of zones. Similarly, U.S. Patent
15 Number 5,572,442, entitled System for Distributing Subscription and On-Demand Audio
16 programming does not teach the same.

17 Therefore it is desirous to have a consumer digitally encode their entire audio, video,
18 and photographic collections to be stored on multimedia storage devices, and have the entire
19 collection synchronized automatically by having the devices communicate to and from each
20 other so that the content is available in multiple locations or zones that the consumer may go.
21 These may include devices such as personal computers located in other rooms or other
22 locations (for example summer home, car, yacht, etc.), or on an online

1 server/website/database. It is preferable to have the content locally stored so that
2 interruptions and skips associated with streaming content over the network does not occur.
3 Additionally, should the storage device such as a hard drive of one unit fails, then the other
4 devices still have complete copies of the content collection to easily replace the failed unit.

5 In light of these considerations, it is an object of this invention to provide a
6 multimedia player device and system which is capable of storing a relatively large amount of
7 digital multimedia programming, whether audio or video, with relatively instant access to any
8 piece of stored data for playback, where the stored data may be replaced with new data when
9 the desires of the user change. It is a further object to provide such a device and system where
10 the transfer of data to the multimedia player device is accomplished through alternative
11 communication means, such that the user can choose from a vast array of data encompassing
12 all formats of audio and video programming and can choose to synchronize the multimedia
13 player device with other multimedia devices on a network for the upload and download of
14 multimedia content from connected network devices.

According to one aspect of this invention there is provided a system for synchronising a multiplicity of devices in a multimedia environment, comprising at least one a system for synchronising a multiplicity of devices in a multimedia environment, comprising at last one central storage and interface device, wherein audio, video and/or photographic information including content information and content management information, relating to at least one user, are stored in digital form, and a plurality of zones each having a zone specific storage and interface device being capable of storing or interfacing with information stored in the central storage and interface device, wherein audio, video and/or photographic information, relating to at least one user, contained within each one of the plurality of zone specific storage and interface devices and the central storage and interface device, are updated in relation with other zone specific storage and interface devices and the central storage and interface device, whereby the at least one user can be situated at anyone of the zones and access substantially identical audio, video and/or photographic information related to the at least one user.

The system may further comprise a local area network (LAN) coupled to at least one zone specific storage and interface device with the central storage and interface device, wherein the interconnections within the LAN is hardwired or wireless.

The system may further comprise a wide area network (WAN) coupling at least one zone specific storage and interface device with the central storage and interface device.

The system may comprise a set of zone specific output devices coupled to each of the zone specific storage and interface device, wherein audio, video

and/or photographic information is outputted, thereby the at least one user is disposed to have substantially identical content information and content management information displayed and manipulated in anyone of the zones.

The system may further comprise an output device coupled to the at least one central storage and interface device, wherein audio, video and/or photographic information is outputted.

The system may further comprise a server, wherein audio, video and/or photographic information contained within each one of the plurality of zone specific storage and interface devices and the central storage and interface device are stored therein and updated at a predetermined time in relation with other zone specific storage and interface devices as well as the central storage and interface device, coupled to the WAN.

The system may further comprise another device coupled to the central storage and interface device via a network connection other than a LAN or WAN.

Preferably the at least one user is configured to pre-store the audio, video and/or photographic information by way of using a device comprising the central storage and interface device, and the zone specific storage and interface device.

Conveniently the central storage and interface device comprises means for transmitting or receiving information to or from the zone specific storage and interface device.

Preferably the zone specific storage and interface device comprises means for transmitting or receiving information to or from the central storage and interface device.

Conveniently the central storage and interface device is capable of converting analogue information into digital form.

Preferably the zone specific storage and interface device is disposed to be coupled to a personal computer (PC).

Conveniently the zone specific storage and interface device is disposed to be coupled to a wireless mobile device.

Preferably the central storage and interface device is disposed to be coupled to a wireless mobile device via LAN.

Advantageously the central storage and interface device is disposed to be coupled to a wireless mobile device via WAN.

According to another aspect of this invention there is provided a method for synchronising a plurality of devices some of which resides in a plurality of zones comprising the steps of providing the plurality of devices, wherein the devices comprise a plurality of zone specific storage and interface devices, at least one central storage and interface device, and at least one other device providing the plurality of zones, wherein at least one zone specific storage and interface device resides therein, and information content specific to a user is maintained at a substantially identical level among the plurality of devices, determining whether a current synchronisation point exists, if a previous synchronisation point exists, receiving information from a server, if a previous synchronisation point does not exist, sending information to a at least one client by a host, wherein the at least one user is disposed to have control, determining what information is needed by the at least one client, and establishing the resultant state as a synchronisation point.

The method may further comprise a step of determining when to initiate synchronisation.

The method may further comprise a step of requesting synchronisation, accepting synchronisation and if synchronisation is declined, reverting back to the step of determining when to initiate synchronisation.

Preferably the step of determining whether a previous synchronisation point exists involves determining what occurred in the time between the last and the current synchronisation point.

Conveniently the information comprises data, content and/or content management information.

Preferably the information comprises data structures in audio, video and/or picture format.

Conveniently the server comprises the at least one central storage and interface device.

Preferably the server comprises an on-line server having a data-base.

SUMMARY OF THE INVENTION

1
2 The invention is an apparatus and system for providing recorded multimedia
3 programming in digital form in a multimedia player device where the user chooses the
4 particular programming so recorded, and further where the recorded programming is updated
5 or replaced periodically with new content from different networked sources. The system
6 comprises a device capable of digitally recording, storing, downloading and uploading
7 multimedia programming in either audio or video formats and is able to transmit such digital
8 data, either via a hardwired or wireless network, where the data is stored on a combination
9 multimedia receiver/player/data storage device, hereafter referred to as a "digital multimedia
10 device." The digital multimedia device comprises a readable/writable memory storage
11 mechanism (e.g., disk drive, hard drive, memory) capable of receiving network
12 transmissions, and a playback or player feature interfaced with a mobile or fixed radio
13 receiver, television or personal computer.

14 The digital player device provides means to display alphanumeric information related
15 to the data chosen to be played on the device itself or through a television or personal
16 computer monitor connection. As data transmissions are received, the data files are stored in
17 digital form within the storage unit as a buffer to be accessed by the playback mechanism, in
18 effect, delaying the playback of the received data until the multimedia works are needed. The
19 data is then converted to "real time" audio or video, utilizing a digital-to-analog converter in
20 the case of audio files, and played back to the listener. As each file is played back, the user
21 may choose to delete the file, and the storage device would then over-write or replace the file
22 with a new file or files of similar size as downloaded by the user from other network
23 resources. The user could also elect to skip the file and proceed to another file stored on the

1 digital player device. As available memory allows, additional multimedia works will continue
2 to be received and stored until the data storage device is filled to capacity, even as the user
3 simultaneously listens to a previously received file. The digital multimedia device is designed
4 to have a relatively large storage capacity, such that hundreds or thousand of files for
5 example, could be stored on the device at any one time.

6 Another feature of the digital multimedia player provides, as new files are constantly
7 being accessed through a network, such as the world wide web, the data storage unit is
8 refreshing the files stored on the digital multimedia device as old files are either deleted or
9 updated by the user. Since the digital multimedia device is connected to the network, it has
10 direct access to the network which allows the listener to request custom files between the
11 listener and other digital player devices, personal computers and music source databases
12 connected to the network.

13 The instant invention teaches a system and method for synchronizing a multiplicity of
14 devices in a multimedia environment. The system has at least one central storage and
15 interface device, wherein audio, video, and photographic information including content
16 information and content management information, relating to at least one user, are stored in
17 digital form. The system further has a plurality of zones each having a zone specific storage
18 and interface device being capable of storing or interfacing with information stored in the
19 central storage and interface device, wherein audio, video, or photographic information,
20 relating to at least one user, contained within each one of the plurality of zone specific
21 storage and interface devices and the central storage and interface device, are updated in
22 relation with other zone specific storage and interface devices and the central storage and
23 interface device. This results in the at least one user can be situated at anyone of the zones

1 and access substantially identical audio, video, and photographic information related to the at
2 least one user.

3 The method includes providing the plurality of devices, providing the plurality of
4 zones, determining whether a current synchronization point exists, if a previous
5 synchronization point exists, receiving information from a server, if a previous
6 synchronization point does not exist, sending information to a at least one client by a host,
7 wherein the at least one user is disposed to have control, determining what information is
8 needed by the at least one client, and establishing the resultant state as a synchronization
9 point.

BRIEF DESCRIPTION OF THE DRAWINGS

Figure 1 is a network diagram with attached devices capable of storing and transmitting digital multimedia files,

Figure 2 is a diagram of the multimedia device shown connected to a user's home audio/video system,

Figure 3 is a diagram of a typical digital multimedia device connected to the user's home audio/video system,

Figure 4 is a flow chart illustrating the procedure for downloading digital content from a content database,

Figure 5 is a flow chart representing the method for synchronizing and downloading multimedia content and updates from the master to subordinate digital multimedia devices connected to the network,

Figure 6 is a flowchart depicting a synchronization of the instant invention, and

Figure 7 is a block diagram depicting an embodiment of the instant invention.

1 DETAILED DESCRIPTION OF THE INVENTION

2 The invention will now be described in detail with regard to the best mode and the
3 preferred embodiment. In general, the invention is a system and method comprising a digital
4 multimedia player connected to a network and capable of transferring, receiving, storing,
5 decoding and playing selected data files from plethora of devices connected to the network,
6 including but not limited to, a computer server, a database, a personal computer and other
7 multimedia receiver/player devices capable of transferring, receiving, storing, decoding and
8 playing the data in an multimedia format.

9 With reference to **Figure 1**, a digital multimedia device **104** is connected to network
10 **102** along with a multimedia database **106**. In addition, a portable multimedia player **108**, a
11 personal computer **110**, and "master" digital multimedia player are connected to network **102**.
12 For purposes of this application, a personal computer is any computer, coupled to a network,
13 which receives a program, data or other application from another data source or computer
14 coupled to the network. In the depicted example, multimedia database **106** provides data,
15 such as digital audio and video files in their respective formats, to digital multimedia device
16 **104** and other devices attached to the network capable of receiving audio and video content.
17 The data network system **102** may include additional servers, clients, digital multimedia
18 players and other devices not shown.

19 The network **102** is capable of delivering data files between the network devices, such
20 as multimedia database **106**, personal computer **110**, portable and fixed digital multimedia
21 devices **104**, **112**. Master digital multimedia device **112** can include a database such as the
22 multimedia database **106**. As can be appreciated the multimedia database **106** can be a stand-
23 alone entity coupled to the network **102** as well. Furthermore, Master digital multimedia

1 device 112 has the capability to store video/audio products in digital form. Preferably, the
2 network 102 comprises a transmission network capable of sending data via an extremely high
3 data transfer rate system, and may comprise satellite, radio, microwave, cellular or other
4 known wireless transmissions using terrestrial or satellite means. It is contemplated that such
5 transfer of data could also be accomplished through direct physical connections (e.g.,
6 telephone lines). Network system 102 may be the Internet representing a worldwide
7 collection of networks and gateways that use the TCP/IP suite of protocols to communicate
8 with one another. Of course, network 102 also may be implemented as a number of different
9 types of networks, such as for example, a wireless internet network, an intranet, a local area
10 network (LAN), or a wide area network (WAN). **Figure 1** is intended as an example, and not
11 as an architectural limitation for the present invention.

12 Multimedia database 106 is an online collection of audio and video works which may
13 accessed by network devices on a subscription or free file share basis. The recorded
14 multimedia works may be categorized as to format, such as classical, jazz, contemporary,
15 classic rock, etc. and video works such as, educational, drama, comedy, etc. The digital
16 source database translates and stores the audio and video works in digital form, with the
17 works subdivided into different formats. The content source database is further capable of
18 communication with other network devices to deliver the data stored in the database to a
19 digital player device, personal computer or microprocessor, or portable personal digital
20 player.

21 With regard to **Figure 2**, the digital multimedia device 104 is a powered electronic
22 device designed to be a stand-alone unit for use in the home or office and powered
23 conventionally by electricity, or a unit installed in an automobile in a manner similar to

1 standard cassette, CD, DVD players. **Figure 2** is a diagram showing one example of the
2 digital multimedia device attached to a user's home audio/video system with a menu display
3 on the user's television. The depicted arrangement allows the user to operate the digital
4 multimedia device's playback functions by utilizing a televised menu driven hierarchy
5 providing the user with program selection choices. As well, the user may program the digital
6 multimedia device's download, synchronization and storage functions via the menu driven
7 interface or alternatively keypad on the digital multimedia device.

8 **Figure 3** depicts key internal components providing for the operation of the invention
9 disclosed herein. The digital multimedia device **104** comprises a microprocessor **302**, a
10 network communication unit **306**, an audio video output **308**, a digital to analog converter
11 (D/A unit) **310** which decodes and converts digitized data to an analog signal output, memory
12 unit **312**, preferably consisting of a disk drive, hard drive, memory or flash card system.
13 capable of storing a large amount of multimedia programming in digital format in individual
14 files, but with a readable/writable storage mechanism that will replace or write over the old
15 data file with one or more new ones, until the data storage device is fully loaded. The data
16 memory unit **312** means of the digital multimedia device **104** could be removable or
17 detachable for transport or even interchangeable with other audio/video player devices. The
18 control unit **314** provides means to select and play any particular multimedia work relatively
19 instantaneously which includes keys and commands for start, stop, skip, repeat, shuffle and
20 save keypad buttons, and a visual display means to display alphanumeric information about
21 the work selected. The invention may also be packaged with a standard AM/FM radio,
22 cassette player or compact disc player, such that the listener has other options for digital
23 audio/video recording and audio/video output.

1 **Figure 4** is a flowchart depicting one embodiment of the present invention. The
 2 digital multimedia device **104** allows the user, via the control unit **314** means, to request and
 3 download entirely new recorded data into the digital multimedia device **104** or program the
 4 digital multimedia device **104** to synchronize and update the user's audio/video files
 5 automatically from a multimedia database **106**, a personal computer or other devices
 6 connected to the network. For example, the user instructs the digital multimedia device to
 7 establish network contact with a selected multimedia database in the required communicating
 8 manner(**Step 402**). The user then selects the desired multimedia works to be synchronized
 9 and downloaded for storage on the digital multimedia device from the music multimedia
 10 database(**Step 404**). The selected digital data is downloaded from the music multimedia
 11 database into the data storage memory unit of the digital multimedia device(**Step 406**). The
 12 user may then be prompted to select other content for download. If the user desires to
 13 download more content, the process is repeated(**Step 408**). If the user declines to download
 14 or update content, the session is terminated. As a result, the user may listen to entirely
 15 different audio/video selections from those previously recorded in the digital multimedia
 16 device.

17 Turning to **Figure 5**, an alternative embodiment of the present invention is discussed
 18 in further detail wherein the digital multimedia player **104** automatically performs the
 19 synchronization and download function between "master" and "subordinate" digital
 20 multimedia devices **104,112**. The subordinate digital multimedia device establishes network
 21 contact with a selected master digital multimedia device in the required communicating
 22 manner. The subordinate digital multimedia device then queries the master multimedia
 23 device to determine if content downloads or updates are available(**Step 502**). The master

1 digital multimedia device responds to the query(Step 504). If content updates or downloads
 2 are available, the subordinate digital player device initiates the synchronization and download
 3 of the appropriate files(Step 506) and the digital data is placed in the data storage memory
 4 unit of the subordinate digital multimedia device(Step 508). After download is complete and
 5 at various selectable intervals, the subordinate digital multimedia player may then again
 6 query the master digital multimedia device for available content downloads or updates and
 7 the process is again repeated. It should be noted that the user may access and play stored
 8 audio content while the digital multimedia player while the update synchronization functions
 9 are being performed and that the process may be initiated by the master digital multimedia
 10 device as opposed to the subordinate multimedia device. Likewise, subordinate multimedia
 11 devices may synchronize with other designated master multimedia devices such as a personal
 12 computer, a source database or other networked storage service or devices, and receive
 13 download updates or multimedia files as set forth in this specification.

14 Referring now to **Figure 6**, a flowchart 600 depicting a synchronization of the instant
 15 invention is described. A process(Step 602) determines when to initiate the synchronization.
 16 Once a decision is made to initiate synchronization, a request is made and the depicted
 17 synchronization process starts(Step 604). A first determination is made in that if the request
 18 is rejected, the process reverts back to step 602 to wait for another initiation as to when to
 19 start the synchronization process(Step 606). If the request is accepted, the flowchart600
 20 progresses onward. A second determination(Step 608) is made in that if a previous
 21 synchronization point, or well established point exists, the synchronization process progresses
 22 to receive information from a server(Step 610). The received information from the server
 23 includes files containing audio and video entries. As can be appreciated, the instant invention

1 teaches a set of zones wherein any zone has a zone specific device that contains all the audio
2 and video, as well as text information which a user needs. If the information is not contained
3 within the zone specific device, it is synchronized and made to contain the information.
4 Therefore, if the device already has the information the user needs, an indication of the
5 information may be transferred instead of the whole information, since the whole information
6 is already in the device albeit not specifically used by a particular user. Continuing
7 describing step 610, information may be received from the server regarding files on audio
8 and video information from a database associated with the server. The information may
9 include coded addition, deletion, or modification instructions for a specific device in a zone.
10 In other words, the database includes time sensitive information related to devices residing in
11 different zones. For example, the user may change her preference in a particular zone at
12 some recent time. This change is going to be noticed by the server, which stores related
13 information in the database. A request for transmission of audio, video, or textual files is
14 generated whereby each zone specific device is going to eventually be comprised of
15 information specified by the user(Step 614). As can be appreciated, the information is the
16 most recent information the user wants it to be. As a result, a new synchronization point is
17 achieved. This synchronization resulted in establishing the most recent well-established
18 synchronization point(Step 616). The synchronization process terminates(Step 618).

19 Referring back to step 608, if no previous synchronization exists, a host sends all the
20 necessary information to at least one client, which comprises the relevant devices in each
21 zone respectively(Step 620). The host can be any device in a zone capable of communicating
22 information to other devices residing in different zones. This usually occurs during the initial
23 or setup period wherein different zone devices have not set themselves up for the user

1 specific information. Or this may occur when substantial change in regard to user preference
 2 is initiated. By way of example, change preference from country music to rock-n-roll. Client
 3 then determines what information is needed and in turn requests and receives the needed
 4 information. As can be appreciated, the needed information comprises audio, video and other
 5 graphic and textual files, or segment of files, or indexing information related to the
 6 above(Step 622). The end result is that different zone specific devices in different zones is
 7 going to have substantially identical information, which the user wants. In addition, a
 8 determination is made in regard to change that has occurred since the establishment of the
 9 last synchronization point, or the optimal state(Step 623). Thereupon, the logic progresses
 10 toward step 614.

11 As can be appreciated, a change or alteration that triggers the synchronization process
 12 includes adding or deleting songs, adding or deleting playlists, editing song or playlist
 13 information, and changing system settings and preferences. It is noted that this triggering
 14 occurs when any device in any particular zone is being altered.

15 Referring now to **Figure 7**, a block diagram **700** of an embodiment of the instant
 16 invention is depicted. A central storage and interface device **702**, which can be a
 17 AudioReQuest Pro, produced by ReQuest, Inc. is shown. It is noted that AudioReQuest is a
 18 trademark of ReQuest, Inc. A local area network (LAN) **704** couples central storage and
 19 interface device **702** to at least one zone specific storage and interface device **706**, which
 20 resides in a specific zone (not shown). Network connection **707** couples other zone specific
 21 storage and interface devices, **708**, **710** each residing in a specific zone (not shown). A
 22 personal computer **712** is coupled to LAN **704** as well. In addition, other device **714** such as
 23 an intelligent MP3 player is coupled to LAN **704** as well. An automobile **716** that has

1 AudioReQuest capabilities is also coupled to LAN 704. It is noted that LAN 704 can be
2 based on Ethernet. And LAN 702 can be landline or wireless such as IEEE 802.11 radio
3 frequency or other suitable standard.

4 A wide area network (WAN) 718 such as the Internet couples together a server 720
5 such as an on-line server, and a second central storage and interface device 722, which in turn
6 can be coupled to a second LAN 724 having at least one zone specific storage and interface
7 device 726 coupled thereto. A second automobile 728 is coupled to WAN 718 as well. As
8 can be appreciated, this coupling can be via direct wireless Internet connections, or a cellular
9 connection.

10 A practical example of implementing the instant invention is described infra. In the
11 example, central storage and interface device 702 is AudioReQuest Pro, zone specific storage
12 and interface devices 706, 708, 710 are a set of AudioReQuest Multizone which is produced
13 by ReQuest, Inc.

14 AudioReQuest Pro (ARQ Pro) is a central storage device that digitally records music
15 from CDs using its built-in CD player. In addition, it can record from analog sources such as
16 radio, Long Playing records (LPs), and audiocassettes through its line-in recording capability.
17 Further, it can record and transfer digital music from the PC and Internet sites. Playlists are
18 created on the system by having a specific user who categorizes song choices for current or
19 later playing. Song and Music Navigation information can be viewed on the built-in LCD, a
20 television interface, or on other devices such as home automation control systems and
21 personal computers through serial and Ethernet connections. Music can be either transferred
22 or streamed to other devices 714 such as digital portable players, streaming network devices,

1 web browsers, personal computers, online websites and servers, and other ReQuest products
2 through network and universal serial bus (USB) connections.

3 AudioReQuest Multizone (ARQ Multi) is an accessory product that has the same
4 storage, playback, and interface as the AudioReQuest Pro. It is noted that the substantial
5 similarity between the AudioReQuest Pro and the AudioReQuest Multizone is achieved by
6 synchronizing of AudioReQuest Pro with AudioReQuest Pro over a network, which may be
7 network 704, 718, 724, or a combination of some or all of them. As can be appreciated,
8 synchronization includes having devices in different zones store substantially identical
9 content of entered music and playlists by a specific user. Different synchronizing schemes
10 are possible (automatic, daily, weekly, etc). It is noted that users can synchronize or make the
11 update on a demand. The ARQ Zone uses an Ethernet connection to a main unit for the
12 synchronization of the music. As can be appreciated, online server 720 includes any main
13 unit such as ARQ pro 702. Any number of ARQ Multizones can work with each ARQ Pro,
14 allowing multiple outputs of the same music collection to be available. In a typical custom
15 home installation, there may be upwards of 20 zones (e.g., rooms) with independent control
16 and output. By way of example, instead of only playing one CD throughout the building,
17 different songs can be played at the same time.

18 The AudioReQuest Pro and ARQ Multizone will have removable hard drives. This is
19 desirable since if any one unit fails, a hard drive from another unit can be inserted and
20 automatically work since they are synchronized such that the content of the hard drives are
21 substantially the same. When an old hard drive is replaced with a new one, it will
22 automatically synchronize and get the entire collection from the ARQ Pro. As can be

1 appreciated, this system provides complete backup redundancy for the entire system,
2 protecting the user's multimedia investment.

3 It is noted that ARQ Multi comprises interfaces for television display, for audio
4 amplification, as well as for infrared connections.

5 Two AudioReQuest Pros can synchronize over the Internet, so that if a customer has
6 their main home in Maine and a second home in Hawaii, both locations will have the same
7 music collection available. Should a new CD be loaded in Hawaii, it will become available
8 in Maine after it synchronizes, and vice versa. In addition, car and other mobile devices can
9 also synchronize over wired or wireless connections. Furthermore, an online file server
10 which exists either through a proprietary system or a public service, can be used to
11 synchronize their music collections from their Pro, so that music can be streamed to their web
12 browser, wireless cell phone, or other devices, anywhere on the Internet.

13 It is noted that the above listed devices such as zone specific storage and interface
14 devices 706, 708, 710, or PC 712 can be located in separate zones respectively. Or, some can
15 co-exist in a zone. One of the purpose of the zone is to give a user substantially exclusive or
16 reclusive enjoyment of information shared by zone specific storage and interface devices 706,
17 708, 710, as well as by central storage and interface device 702, and other devices.

18 As can be appreciated, this invention teaches a complete system for providing
19 recorded multimedia programming in digital form that is updated on all devices owned by an
20 user, over a network. The main device digitally records, stores, plays, downloads, and
21 uploads multimedia programming in audio, video, or picture format or other suitable formats
22 (analog or digital). Furthermore, the main device is able to transmit such data, either via a
23 hardwired or wireless network, to other devices that are capable of playing, storing, download

1 and upload such data. The system allows multiple devices to synchronize its internal
2 collection with each other, so that the end result is that all the devices have the same content
3 and content management means (playlists, settings, etc.). Also devices having accessory
4 playback only feature would synchronize the data, content, and content management with the
5 main recorder device. Therefore, a complete copy of the content is stored locally in a device
6 within a zone or any zone, so that the output can be played in multiple zones or rooms in a
7 networked building or in multiple locations traveling through a wide area network such as the
8 Internet. The instant invention also contemplates other device for mobile applications such as
9 car, boat, airplane, and other transportation, that would synchronize through either hardwired
10 or wireless means resulting in storing the content locally. The instant invention further
11 contemplates a set of digital multimedia devices comprises a readable/writable memory
12 storage mechanism (e.g. disk drive, hard drive, memory) which are capable of transmitting
13 and/or receiving network transmissions, and a playback or player feature interfaced with a
14 mobile or fixed radio receiver, television, or personal computer.

15 Although preferred embodiments of the present invention have been described in the
16 foregoing Detailed Description and illustrated in the accompanying drawings, it will be
17 understood that the invention is not limited to the embodiments disclosed, but is capable of
18 numerous rearrangements, modifications, and substitutions of steps without departing from
19 the spirit of the invention. Accordingly, the present invention is intended to encompass such
20 rearrangements, modifications, and substitutions of steps as fall within the scope of the
21 appended claims.

CLAIMS:

1. A system for synchronising a multiplicity of devices in a multimedia environment, comprising at least one central storage and interface device, wherein audio, video and/or photographic information including content information and content management information, relating to at least one user, are stored in digital form, and a plurality of zones each having a zone specific storage and interface device being capable of storing or interfacing with information stored in the central storage and interface device, wherein audio, video and/or photographic information, relating to at least one user, contained within each one of the plurality of zone specific storage and interface devices and the central storage and interface device, are updated in relation with other zone specific storage and interface devices and the central storage and interface device, whereby the at least one user can be situated at anyone of the zones and access substantially identical audio, video and/or photographic information related to the at least one user.
2. The system of Claim 1 further comprising a local area network (LAN) coupled to at least one zone specific storage and interface device with the central storage and interface device, wherein the interconnections within the LAN is hardwired or wireless.
3. The system of Claim 1 or 2 further comprising a wide area network (WAN) coupling at least one zone specific storage and interface device with the central storage and interface device.

4. The system of any one of the preceding Claims further comprising a set of zone specific output devices coupled to each of the zone specific storage and interface device, wherein audio, video and/or photographic information is outputted, thereby the at least one user is disposed to have substantially identical content information and content management information displayed and manipulated in anyone of the zones.

5. The system of any one of the preceding Claims further comprising an output device coupled to the at least one central storage and interface device, wherein audio, video and/or photographic information is outputted.

6. The system of any one of the preceding claims further comprising a server, wherein audio, video and/or photographic information contained within each one of the plurality of zone specific storage and interface devices and the central storage and interface device are stored therein and updated at a predetermined time in relation with other zone specific storage and interface devices as well as the central storage and interface device, coupled to the WAN.

7. The system of any one of the preceding Claims further comprising another device coupled to the central storage and interface device via a network connection other than a LAN or WAN.

8. The system of any one of the preceding Claims wherein the at least one user is configured to pre-store the audio, video and/or photographic information by way of using a device comprising the central storage and interface device, and the zone specific storage and interface device.

9. The system of any one of the preceding Claims wherein the central storage and interface device comprises means for transmitting or receiving information to or from the zone specific storage and interface device.
10. The system of any one of the preceding Claims wherein the zone specific storage and interface device comprises means for transmitting or receiving information to or from the central storage and interface device.
11. The system of any one of the preceding Claims wherein the central storage and interface device is capable of converting analogue information into digital form.
12. The system of any one of the preceding Claims wherein the zone specific storage and interface device is disposed to be coupled to a personal computer (PC).
13. The system of any one of the preceding Claims wherein the zone specific storage and interface device is disposed to be coupled to a wireless mobile device.
14. The system of any one of the preceding Claims wherein the central storage and interface device is disposed to be coupled to a wireless mobile device via LAN.
15. The system of any one of the preceding Claims wherein the central storage and interface device is disposed to be coupled to a wireless mobile device via WAN.

16. A method for synchronising a plurality of devices some of which resides in a plurality of zones comprising the steps of providing the plurality of devices, wherein the devices comprise a plurality of zone specific storage and interface devices, at least one central storage and interface device, and at least one other device providing the plurality of zones, wherein at least one zone specific storage and interface device resides therein, and information content specific to a user is maintained at a substantially identical level among the plurality of devices, determining whether a current synchronisation point exists, if a previous synchronisation point exists, receiving information from a server, if a previous synchronisation point does not exist, sending information to a at least one client by a host, wherein the at least one user is disposed to have control, determining what information is needed by the at least one client, and establishing the resultant state as a synchronisation point.

17. The method of Claim 16 further comprising a step of determining when to initiate synchronisation.

18. The method of Claim 17 further comprising a step of requesting synchronisation, accepting synchronisation and if synchronisation is declined, reverting back to the step of determining when to initiate synchronisation.

19. The method of Claim 17 or 18 wherein the step of determining whether a previous synchronisation point exists involves determining what occurred in the time between the last and the current synchronisation point.

20. The method of any one of claims 16 to 19 wherein the information comprises data, content and/or content management information.

21. The method of any one of Claims 16 to 20 wherein the information comprises data structures in audio, video and/or picture format.
22. The method of any one of Claims 16 to 21 wherein the server comprises the at least one central storage and interface device.
23. The method of any one of Claims 16 to 22 wherein the server comprises an on-line server having a data-base.



INVESTOR IN PEOPLE

Application No: GB 0213763.6
Claims searched: 1-23

Examiner: Geoff Western
Date of search: 6 January 2003

Patents Act 1977 : Search Report under Section 17

Documents considered to be relevant:

Category	Relevant to claims	Identity of document and passage or figure of particular relevance
X	1-15	"Novell iFolder synchronizes information access across the net; provides anytime, anywhere access", press release, 19 March 2001, at www.novell.com/news/press/archive/2001/03/pr01021.html
X	1-15	US 5978805 A (CARSON) N.b. columns 1-4
X	1-15	WO 2001/016804 A2 (QUALCOMM) N.b. page 3
X	1-15	US 6000000 A (HAWKINS et al) N.b. columns 1-3
X	1-15	US 5729735 A (MEYERING) See whole document
X,E	1-15	WO 2002/077862 A (EDWARDS et al) N.b. pages 1-3

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	Filing Date	2013-04-17
	First Named Inventor	MILLINGTON; Nicholas A.J.
	Art Unit	2442
	Examiner Name	SURVILLO, Oleg
	Attorney Docket Number	13-0402

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	Attorney Docket Number	13-0402	

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CERTIFICATION STATEMENT			
Please see 37 CFR 1.97 and 1.98 to make the appropriate selection(s):			
<p>That each item of information contained in the information disclosure statement was first cited in any communication from a foreign patent office in a counterpart foreign application not more than three months prior to the filing of the information disclosure statement. See 37 CFR 1.97(e)(1).</p>			
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A signature of the applicant or representative is required in accordance with CFR 1.33, 10.18. Please see CFR 1.4(d) for the form of the signature.			
Signature	/Benjamin M. Urban/	Date (YYYY-MM-DD)	2016-10-13
Name/Print	Benjamin M. Urban	Registration Number	71,591
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7. A record from this system of records may be disclosed, as a routine use, to the Administrator, General Services, or his/her designee, during an inspection of records conducted by GSA as part of that agency's responsibility to recommend improvements in records management practices and programs, under authority of 44 U.S.C. 2904 and 2906. Such disclosure shall be made in accordance with the GSA regulations governing inspection of records for this purpose, and any other relevant (i.e., GSA or Commerce) directive. Such disclosure shall not be used to make determinations about individuals.
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9. A record from this system of records may be disclosed, as a routine use, to a Federal, State, or local law enforcement agency, if the USPTO becomes aware of a violation or potential violation of law or regulation.

Electronic Acknowledgement Receipt	
EFS ID:	27176079
Application Number:	13864249
International Application Number:	
Confirmation Number:	9806
Title of Invention:	METHOD AND APPARATUS FOR PLAYING AND PROVIDING AUDIO CONTENT IN A SYNCHRONY GROUP
First Named Inventor/Applicant Name:	Nicholas A. J. Millington
Customer Number:	107361
Filer:	Benjamin Michael Urban
Filer Authorized By:	
Attorney Docket Number:	13-0402 (MBHB 15-518-CON)
Receipt Date:	13-OCT-2016
Filing Date:	17-APR-2013
Time Stamp:	14:24:12
Application Type:	Utility under 35 USC 111(a)

Payment information:

Submitted with Payment	no				
File Listing:					
Document Number	Document Description	File Name	File Size(Bytes)/ Message Digest	Multi Part /.zip	Pages (if appl.)
1	Foreign Reference	JP2000149391.PDF	502149 26b08e237f101717605ce2a98520fa4c7b63d0fb	no	10
Warnings:					

Information:					
2	Foreign Reference	JP63269633.PDF	873462	no	7
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4	Foreign Reference	WO2000019693.PDF	1693361	no	38
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6	Foreign Reference	WO2007135581A2.PDF	1027402	no	19
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8	Non Patent Literature	ZX135_INSTALLATION_MANUAL_2003.pdf	4609544	no	44
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9	Non Patent Literature	LG_RJP-201M_REMOTE_2010.PDF	20283650	no	24
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37	Other Reference-Patent/App/Search documents	Sonos_DM_Holdings_EXHIBIT2 _INVALIDITY_CHARTS_8588949 _PATENT.PDF	809899	no	112
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44	Other Reference-Patent/App/Search documents	Sonos_DM_Holdings_EXHIBIT9_INVALIDITY_CHARTS_9202509_PATENT.PDF	739299	no	163
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53	Other Reference-Patent/App/Search documents	Sonos_v_DM_Defendants_1st_AMENDED_ANSWER_TO_3RD_AMENDED_COMPLAINT_9-7-2016.pdf	22634863	no	23
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55	Other Reference-Patent/App/Search documents	Sonos_v_DM_Exhibit_A_2nd_AMENDED_ANSWER_TO_3rd_AMENDED_COMPLAINT_9-9-2016.pdf	7167661	no	43
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58	Information Disclosure Statement (IDS) Form (SB08)	13-0402_MBHB15-518- CON_IDS_second_13OCT2016. pdf	1057309 4bee3d60f9b0f2d7501a1c87dfdd56a23b0 74c7b	no	10
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<p>This Acknowledgement Receipt evidences receipt on the noted date by the USPTO of the indicated documents, characterized by the applicant, and including page counts, where applicable. It serves as evidence of receipt similar to a Post Card, as described in MPEP 503.</p> <p><u>New Applications Under 35 U.S.C. 111</u> If a new application is being filed and the application includes the necessary components for a filing date (see 37 CFR 1.53(b)-(d) and MPEP 506), a Filing Receipt (37 CFR 1.54) will be issued in due course and the date shown on this Acknowledgement Receipt will establish the filing date of the application.</p> <p><u>National Stage of an International Application under 35 U.S.C. 371</u> If a timely submission to enter the national stage of an international application is compliant with the conditions of 35 U.S.C. 371 and other applicable requirements a Form PCT/DO/EO/903 indicating acceptance of the application as a national stage submission under 35 U.S.C. 371 will be issued in addition to the Filing Receipt, in due course.</p> <p><u>New International Application Filed with the USPTO as a Receiving Office</u> If a new international application is being filed and the international application includes the necessary components for an international filing date (see PCT Article 11 and MPEP 1810), a Notification of the International Application Number and of the International Filing Date (Form PCT/RO/105) will be issued in due course, subject to prescriptions concerning national security, and the date shown on this Acknowledgement Receipt will establish the international filing date of the application.</p>					



Espacenet

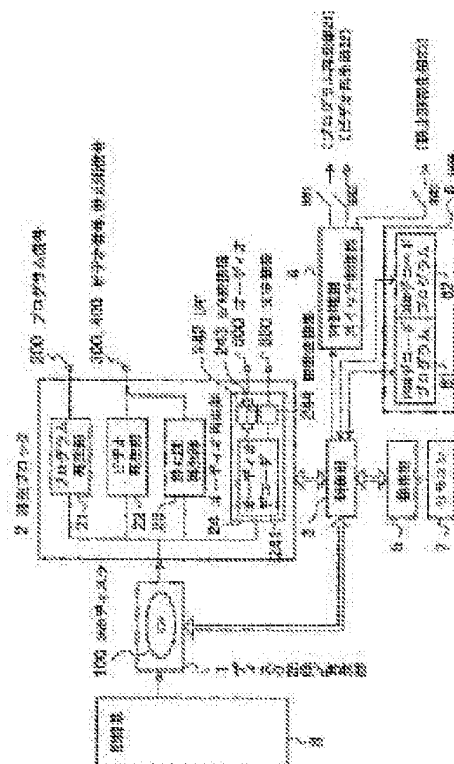
Bibliographic data: JP2000149391 (A) — 2000-05-30

DISK REPRODUCING DEVICE

Inventor(s): TANAKA YOSHIKI $\pm$ (TANAKA YOSHIKI)
Applicant(s): VICTOR COMPANY OF JAPAN $\pm$ (VICTOR CO OF JAPAN LTD)
Classification: - **international:** **G11B19/00; G11B19/12;** (IPC1-7): G11B19/00;
G11B19/12
- **cooperative:**
Application number: JP19980313579 19981104
Priority number (s): JP19980313579 19981104

Abstract of JP2000149391 (A)

PROBLEM TO BE SOLVED: To save electric power by supplying an electric power only to a necessary reproducing system according to kinds of data to be reproduced from a disk. SOLUTION: When a DVD disk is loaded on a disk recording/reproducing part 1, a control unit 3 judges whether the disk is a DSD disk or other disks by a disk identifier, and when it is confirmed as a DSD disk an internal power source switch control part switches off a power source to a program reproducing part, a video reproducing part, and a still picture reproducing part by making off internal power source switches SW1, SW2, SW3. The reproduced data read from the disk recording/reproducing part 1 is reproduced by an audio reproduction part as an audio signal. Since the power supply to unnecessary reproduction systems is switched off when the DSD disk is reproduced like the above, electric power saving is possible.



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特開2000-149391

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19/00	5 0 1	19/00	5 0 1 H

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(71)出願人 000004329

日本ビクター株式会社

神奈川県横浜市神奈川区守屋町3丁目12番地

(72)発明者 田中 美昭

神奈川県横浜市神奈川区守屋町3丁目12番地 日本ビクター株式会社内

(74)代理人 100083806

弁理士 三好 秀和 (外9名)

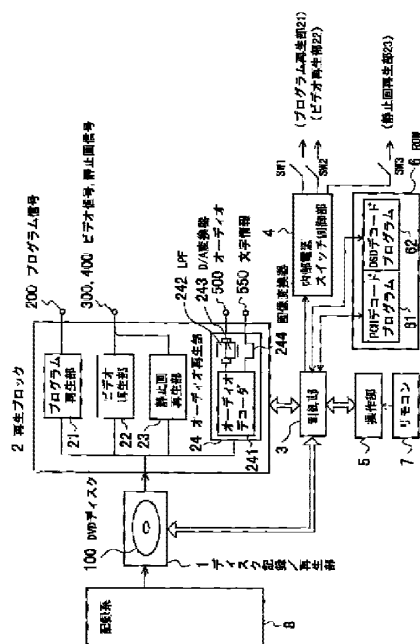
Fターム(参考) 5D066 HA01

(54)【発明の名称】 ディスク再生装置

(57)【要約】

【課題】 ディスクから再生するデータの種別に応じて必要な再生系のみ電源を供給して省電力化を図ること。

【解決手段】 DVDディスクがディスク記録/再生部1にセットされると、制御部3はディスク識別子より、DSDディスクか、その他のディスクかを判定し、DSDディスクであった場合は、内部電源スイッチ制御部により、内部電源スイッチSW1、SW2、SW3をオフして、プログラム再生部、ビデオ再生部、静止画再生部への電源をオフにする。ディスク記録/再生部1から読み出された再生データはオーディオ再生部によりオーディオ信号となって再生される。上記のようにDSDディスクを再生する際、不要な再生系への電源がオフとなるため、省電力化を図ることができる。



【特許請求の範囲】

【請求項1】 コンピュータプログラム、ビデオデータ、静止画データ及びオーディオデータの一部又は全部を記録した第1のディスク、オーディオデータ及び静止画データを記録した第2のディスク、オーディオデータのみを記録した第3のディスクを互換再生するために、コンピュータプログラムを再生するプログラム再生部と、ビデオデータを再生するビデオ再生部と、静止画データを再生する静止画再生部と、オーディオデータを再生するオーディオ再生部とを有するディスク再生装置において、

再生するディスクが上記第3のディスクであるか否かを判定する判定手段と、

この判定手段によって上記第3のディスクであると判定された場合、前記プログラム再生部、ビデオ再生部、静止画再生部への電源の供給を停止するパワーセーブ手段とを備えたことを特徴とするディスク再生装置。

【請求項2】 コンピュータプログラム、ビデオデータ、静止画データ及びオーディオデータの一部又は全部を記録した第1のディスク、オーディオデータ及び静止画データを記録した第2のディスク、オーディオデータのみを記録した第3のディスクを互換再生するために、コンピュータプログラムを再生するプログラム再生部と、ビデオデータを再生するビデオ再生部と、静止画データを再生する静止画再生部と、オーディオデータを再生するオーディオ再生部とを有するディスク再生装置において、

ディスクから再生されるデータの種別を判別する判別手段と、

前記プログラム再生部とビデオ再生部と静止画再生部とオーディオデータ再生部の中の前記判別手段によって判別されたデータを再生しない再生部への電源の供給を停止するパワーセーブ手段とを備えたことを特徴とするディスク再生装置。

【請求項3】 コンピュータプログラム、ビデオデータ、静止画データ及びオーディオデータの一部又は全部を記録した第1のディスク、オーディオデータ及び静止画データを記録した第2のディスク、オーディオデータのみを記録した第3のディスクを互換再生するために、コンピュータプログラムを再生するプログラム再生部と、ビデオデータを再生するビデオ再生部と、静止画データを再生する静止画再生部と、オーディオデータを再生するオーディオ再生部とを有するディスク再生装置において、

再生するディスクが上記第3のディスクであるか否かを判定する判定手段と、

この判定手段によって上記第3のディスクであると判定された場合、前記プログラム再生部、ビデオ再生部、静止画再生部への電源の供給を停止する第1のパワーセーブ手段と、

前記判定手段によって再生するディスクが上記第3のディスク以外の他のディスクである場合、このディスクから再生されるデータの種別を判別する判別手段と、

前記プログラム再生部とビデオ再生部と静止画再生部とオーディオデータ再生部の中の前記判別手段によって判別されたデータを再生しない再生部への電源の供給を停止する第2のパワーセーブ手段とを備えたことを特徴とするディスク再生装置。

【発明の詳細な説明】

【0001】

【発明の属する技術分野】本発明は、各種のDVDディスクを再生するディスク再生装置に関する。

【0002】

【従来の技術】従来より、DVDディスクには、ビデオデータを含むAvdディスク、オーディオデータと静止画に特化したAディスク及びオーディオデータのみをDSDディスク（1ビット信号であるDSDを用いてデータを記録しているディスク）とがある。

【0003】しかも、これらのディスクにはコンピュータプログラムデータが記録されている場合があるため、上記した全てのディスクを互換再生するディスク再生装置はオーディオデータの再生部と、静止画の再生部と、ビデオデータの再生部と、コンピュータプログラムデータの再生部とが設けられ、上記した各種のディスクに記録されている全てのデータを再生することができるようになっている。

【0004】

【発明が解決しようとする課題】ところで、上記したディスク再生装置で、DSDディスクを再生する場合、DSDディスクには、オーディオデータしか入っていないので、オーディオデータの再生部のみ動作させれば良いが、同時に、静止画の再生部と、ビデオデータの再生部と、コンピュータプログラムデータの再生部も動作してしまうと、その分、無駄な電力を消費してしまうという問題が出てくる。特に、ディスク再生装置が携帯型で、電源がバッテリーなどの場合、無駄な電力消費は重大な問題で、バッテリー電力の消耗を早め、充電、又は電池交換からの装置の再生可能時間を短くしてしまうという問題を生じる。

【0005】本発明は、上述の如き従来の課題を解決するためになされたもので、その目的は、ディスクから再生するデータの種別に応じて必要な再生系のみ電源を供給して省電力化を図ることができ、特に携帯型のディスク再生時間を延ばすことができるディスク再生装置を提供することを目的としている。

【0006】

【課題を解決するための手段】上記目的を達成するために、第1の発明の特徴は、コンピュータプログラム、ビデオデータ、静止画データ及びオーディオデータの一部又は全部を記録したAvdディスク、オーディオデータ