VIRTUAL WIRELESS MULTITRACK RECORDING SYSTEM

BACKGROUND OF THE INVENTION

Embodiments of the present invention generally relate to systems and methods for recording and processing audio received from one or more wireless devices. More specifically, the present invention relates to systems and methods for recording and processing audio having one or more tracks received from one or more wireless devices operating in either an asynchronous or synchronous mode.

Many systems and methods have been created to record performance audio. Some such systems include a multi-track audio recorder wired to one or more boom microphones. Typically, one or more performers performing on a sound stage are recorded by one or more boom microphones that are directly wired to the multi-track recorder. The multi-track recorder combines the single track of audio received from each boom microphone to create one multi-track audio file. In many such systems, the received audio and/or the multi-track audio is timestamped with a time reference signal such as a Society of Motion Picture and Television Engineers ("SMPTE") timecode signal containing information regarding the hour, minute, second, frame, type of timecode (i.e., nondrop or drop frame), and user-definable information. Such information allows audio to be more easily matched and/or combined with simultaneously recorded video.

Other such systems include a multi-track audio recorder receiving audio wirelessly from one or more wireless transmitters. Such wireless transmitters may take the form of a

Zaxcom, Inc. Exhibit 2062
RØDE Microphones, LLC, et. al v. Zaxcom, Inc.
Page 1 of 40

body pack that is worn by each performer. Typically, one or more performers performing on a sound stage are recorded by one or more boom microphones that are directly wired to the multi-track recorder. In such systems, the audio may also be captured wirelessly from each performer's wireless transmitter body pack. A wireless link from the wireless transmitter body pack to the multi-track recorder is then used to transmit the audio. By this method, several tracks of audio are created that must be recorded synchronously to each other but also to a master timecode source which is needed for organization in the sound and video editing process.

The audio must be time-stamped to within approximately thirty milliseconds ("ms") accuracy to be useable as a sound versus film time-reference. The individual audio tracks must maintain an accuracy of less than one ms with respect to each other in order to prevent phasing artifacts when the audio is combined into a single audio channel. Using the methods of recording, the audio received from each body pack transmitter as well as the audio from hard-wired boom microphones into a single multi-track recorder easily attains the required time synchronization with the audio tracks to each other and the combined audio tracks' timecode stamp.

SUMMARY OF THE INVENTION

A recording system for wirelessly recording audio is provided in one aspect of the present invention. In one embodiment of the present invention, a recording system wirelessly records live audio events, such as performances, movie takes, etc. having one or

Zaxcom, Inc. Exhibit 2062
RØDE Microphones, LLC, et. al v. Zaxcom, Inc.
Page 2 of 40

more performers. In some such embodiments, each performer dons a dedicated local audio device that receives audio generated by the respective performer through a microphone, headset, or the like. Each local audio device then records the audio received from its respective performer locally within the local audio device and associates individual segments of audio with highly accurate timestamps as compared to the timestamps of all other local audio devices. The accuracy of each local audio device's timestamps relative to each other allow the audio recorded at each local audio device to be combined audibly or electronically after the original performance has ended to form a single audio file (e.g., a multi-track audio file) in accordance with the systems and methods of the present invention.

In one aspect of the present invention, timestamps are generated at each local audio device based upon the timecodes provided by a local, on board timecode generator. In some such embodiments, the local timecode generators of all local audio devices are synchronized to a master timecode source to maintain the accuracy of each local audio device's timestamps relative to all other recording system devices. Synchronization may be performed via hardware or software designed to continuously compare master timecodes to locally generated timecodes such that the speed of each local timecode generator may be adjusted as necessary to keep pace with the master timecode source.

In some embodiments of the present invention, in addition to locally recording audio received from its respective performer, local audio devices also wirelessly transmit the received audio to a master recorder (e.g., a multi-track recorder that combines all received audio into a single audio file). In such embodiments, the ability to synchronize the local timestamps at each local audio device and other components of the recording system (e.g.,

Zaxcom, Inc. Exhibit 2062
RØDE Microphones, LLC, et. al v. Zaxcom, Inc.
Page 3 of 40

master recorder, master receiver, etc.) using the methods of the present invention allows any audio that is not recorded by the master recorder during a live event due to, for example, transmission errors to be recovered by replaying the missed audio and recording the replayed audio in the correct time sequence with respect to the other audio samples. In other words, since the audio samples are stored locally in each local audio device with timestamps that are synchronized with the timestamps of recorder, whenever audio is not recorded at recorder **108**, it may simply be replayed at local audio devices **102** starting at the timecode of the missed audio. Since the local audio device and recorder timestamps are synchronized, the replayed audio may be inserted in the proper time sequence with respect to the other recorded audio samples based upon the synchronized timestamp data. Synchronization is essential to ensure that each performer's audio is synchronized with all other performers' audio and to ensure that the newly recorded replayed audio is in the correct sequence with respect to the previously recorded live audio. Such synchronization must maintain a high accuracy for each performer's timestamps with respect to all other performers' timestamps to prevent the occurrence of phasing artifacts when the multiple audio recordings are combined to create one single recording.

In some embodiments of the present invention, receiver **106** automatically senses an error in transmission caused by, for example, a communication loss, interference, etc. **[[GLENN-IS THIS AUTOMATIC SENSING COMMONLY AVAILABLE OR A NOVEL ASPECT OF THE INVENTION?]]** Upon sensing a transmission error, receiver **106** may transmit a request to RCU **104** requesting playback of the audio recorded locally on local audio devices **102** beginning at a timecode prior to the occurrence of the transmission error. In response, RCU **104** transmits a digital command to all local audio devices **102** to

Zaxcom, Inc. Exhibit 2062
RØDE Microphones, LLC, et. al v. Zaxcom, Inc.
Page 4 of 40

playback the audio stored in the respective local memory **316** that occurred subsequent to the timecode requested by receiver **106** in the manner described below with respect to FIG. 6.

Alternatively, playback may be requested manually by a user of a recording system such as recording system **100**. In this scenario, upon hearing that a transmission error (i.e., a loss of audio data) has occurred, the user manually prompts RCU **104** to transmit a digital command to all local audio devices **102** to playback the audio stored in memory **332** (FIG. 3) that occurred subsequent to a time reference entered at RCU **104** by the user. Such prompting may occur after the audio event ends or immediately upon hearing the transmission error. If the latter option is chosen, prompting playback of a specific segment of the audio event may index the local audio devices to store the requested data in a protected memory location until the end of the audio event to avoid disrupting the live recording. In this scenario, all requested audio shall be replayed after the performance ends. In embodiments of the present invention in which data is recorded in a loop (i.e., when memory is full, new data overwrites previously recorded data), writing the data to a protected memory location removes it from the loop and protects it from being overwritten.

Other objects, features, and characteristics of the present invention, as well as the methods of operation and functions of the related elements of the structure, and the

Zaxcom, Inc. Exhibit 2062
RØDE Microphones, LLC, et. al v. Zaxcom, Inc.
Page 5 of 40

combination of parts and economies of manufacture, will become more apparent upon consideration of the following detailed description with reference to the accompanying drawings, all of which form a part of this specification.

BRIEF DESCRIPTION OF THE DRAWINGS

A further understanding of the present invention can be obtained by reference to the embodiments set forth in the illustrations of the accompanying drawings. Although the illustrated embodiments are exemplary of systems for carrying out the present invention, both the organization and method of operation of the invention, in general, together with further objectives and advantages thereof, may be more easily understood by reference to the drawings and the following description. The drawings are not intended to limit the scope of this invention, which is set forth with particularity in the claims as appended or as subsequently amended, but merely to clarify and exemplify the invention.

FIG. 1 is …..

DETAILED DESCRIPTION

As required, a detailed illustrative embodiment of the present invention is disclosed herein. However, techniques, systems and operating structures in accordance with the present invention may be embodied in a wide variety of forms and modes, some of which may be quite different from those in the disclosed embodiment. Consequently, the specific structural and functional details disclosed herein are merely representative, yet in that regard, they are deemed to afford the best embodiment for purposes of disclosure and to

Zaxcom, Inc. Exhibit 2062
RØDE Microphones, LLC, et. al v. Zaxcom, Inc.
Page 6 of 40

provide a basis for the claims herein, which define the scope of the present invention. The following presents a detailed description of one embodiment (as well as some alternative embodiments) of the present invention.

Referring first to FIG. 1, depicted is recording system **100** in accordance with one embodiment of the present invention. Recording system **100** wirelessly records live audio events, such as performances, movie takes, etc. having one or more performers. In one aspect of the present invention, all of the components of recording system **100** are synchronized to allow each component to accurately stamp its recorded audio with the time at which it occurred such that the timestamps (i.e., information stored with an audio sample or audio file conveying the time at which the audio sample or first audio sample of the file occurred) created by each individual component of recording system **100** are highly accurate as compared to the timestamps created by all other components of recording system **100**. This accuracy allows multiple individually recorded audio tracks to be combined into one multi-track audio file electronically post-recording. Furthermore, this accuracy allows recording system **100** to automatically correct for any audio data lost during an original live recording due to wireless transmission problems such as dropout, interference, etc. This automatic correction may be performed either electronically or via synchronized playback of the individually recorded audio tracks. In another aspect of the present invention, the audio recorded by recording system **100** may be recorded asynchronously. In this scenario, the audio is synchronized and/or mixed post-recording to automatically correct for any audio data lost due to wireless transmission problems such as dropout, interference, etc.

In the embodiment of the present invention depicted in FIG. 1, recording system **100**

Zaxcom, Inc. Exhibit 2062
RØDE Microphones, LLC, et. al v. Zaxcom, Inc.
Page 7 of 40

includes local audio devices **102**, remote control unit ("RCU") **104**, receiver **106**, and recorder **108**. In one embodiment, each of the components of recording system **100** include an RF transmitter capable of transmitting one or more of a time reference signal, digital commands, and audio to one or more other components of recording system **100**. Additionally, RCU **104** may be equipped with the capability of remotely controlling local audio devices **102**, receiver **106**, and recorder **108** to perform tasks including, but not limited to, initiating audio playback of all local audio devices **102** starting at the same time reference, as well as recording thereof by receiver **106** and recorder **108**.

Both live and replayed audio transmitted by local audio devices **102** may be received at receiver **106** and recorded by audio recorder **108**. Receiver **106** and recorder **108** may be virtually any commercially available receiver and recorder. Receiver **106** receives the wireless RF signals (e.g., modulated RF carrier signals) generated by all active local audio devices **102** and converts the signals to a format capable of being recorded by a commercially available recording device including, but not limited to, Zaxcom, Inc.'s DEVA® multi-track recorder. In some embodiments, such commercially available recording devices record audio with a locally generated SMPTE-compatible timecode signal.

The ability to synchronize the local timestamps at each local audio device **102** and recorder **108** using the methods of the present invention as discussed in greater detail below allows any audio that is not recorded by recorder **108** during a live event due to transmission errors to be recovered by replaying the missed audio and recording the replayed audio in the correct time sequence with respect to the other audio samples. In other words, since the audio samples are stored locally in each local audio device **102** with

timestamps that are synchronized with the timestamps of recorder **108**, whenever audio is not recorded at recorder **108**, it may simply be replayed at local audio devices **102** starting at the timecode of the missed audio. Since the local audio device and recorder timestamps are synchronized, the replayed audio may be inserted in the proper time sequence with respect to the other recorded audio samples based upon the synchronized timestamp data. Synchronization is essential to ensure that each performer's audio is synchronized with all other performers' audio and to ensure that the newly recorded replayed audio is in the correct sequence with respect to the previously recorded live audio. Such synchronization must maintain a high accuracy for each performer's timestamps with respect to all other performers' timestamps to prevent the occurrence of phasing artifacts when the multiple audio recordings are combined to create one single recording.

In some embodiments of the present invention, receiver **106** automatically senses an error in transmission caused by, for example, a communication loss, interference, etc. **[[GLENN-IS THIS AUTOMATIC SENSING COMMONLY AVAILABLE OR A NOVEL ASPECT OF THE INVENTION?]]** Upon sensing a transmission error, receiver **106** may transmit a request to RCU **104** requesting playback of the audio recorded locally on local audio devices **102** beginning at a timecode prior to the occurrence of the transmission error. In response, RCU **104** transmits a digital command to all local audio devices **102** to playback the audio stored in the respective local memory **316** that occurred subsequent to the timecode requested by receiver **106** in the manner described below with respect to FIG. 6.

Alternatively, playback may be requested manually by a user of a recording system such as recording system **100**. In this scenario, upon hearing that a transmission error (i.e.,

a loss of audio data) has occurred, the user manually prompts RCU **104** to transmit a digital command to all local audio devices **102** to playback the audio stored in memory **332** (FIG. 3) that occurred subsequent to a time reference entered at RCU **104** by the user. Such prompting may occur after the audio event ends or immediately upon hearing the transmission error. If the latter option is chosen, prompting playback of a specific segment of the audio event may index the local audio devices to store the requested data in a protected memory location until the end of the audio event to avoid disrupting the live recording. In this scenario, all requested audio shall be replayed after the performance ends. In embodiments of the present invention in which data is recorded in a loop (i.e., when memory is full, new data overwrites previously recorded data), writing the data to a protected memory location removes it from the loop and protects it from being overwritten.

[[GLENN-IS THIS THE FIRST WIRELESS MICROPHONE TO HAVE LOCAL RECORDING CAPABILITY? ALSO, IS THIS THE FIRST WIRELESS MICROPHONE TO INCLUDE A LOCAL RECEIVER??]]

FIG. 2A depicts a block diagram of one embodiment of RCU **104** in accordance with the present invention. In this embodiment, RCU **104** includes, *inter alia*, RCU timecode generator **204**, RCU power supply **206**, RCU transmitter **208**, RCU local control unit **210**, RCU audio input device **212**, RCU audio input device port **214**, RCU preamp **216**, RCU display **218**, RCU keypad **220**, RCU ADC **222**, RCU DAC **224**, RCU amp **226**, audio output device port **228**, audio output device **230**, external interface **252**, and external interface port **254**.

RCU transmitter **208** allows RCU **104** to transmit a master time reference signal, digital commands, audio, and the like to other devices such as local audio devices **102**,

receiver **106**, and recorder **108**. In one aspect of the present invention, the time reference signal is a SMPTE timecode signal containing information regarding the hour, minute, second, frame, type of timecode (i.e., nondrop or drop frame), and user-definable information (e.g., the transport status of recorder **108**, the name of a scene, etc.). This master time reference signal provides a time reference for all local audio devices **102**, which may use this information for a variety of purposes such as jam synchronizing their respective local timecode generators **304** (FIG. 3), adjusting the speed of the local timecode generators **304** (FIG. 3), timestamping locally recorded audio, etc.

[[The master time reference signal may be generated local to RCU **104** by an on-board timecode generator such as RCU timecode generator **204** or may be generated remotely by one of local audio devices **102**, receiver **106**, recorder **108**, or a standalone master timecode generator. In the latter embodiments, RCU timecode generator **204** may be removed from RCU **104** and the master time reference signal may be received by RCU **104** via RCU receiver **202**. The master time reference signal data may then be processed by RCU local control unit **210** prior to transmission via RCU transmitter **208** to the other components of recording system **100**. Or, alternatively, the master time reference signal data may be provided directly to RCU transmitter **208** by RCU receiver **202**.]]

When recording system **100** is operating in a synchronous mode, transmission of the master time reference signal ensures that all of the components of recording system **100** store all locally recorded audio with timestamps that are highly accurate as compared to the timestamps of all other local audio devices **102** and/or all other components of recording system **100**. The timestamps are then used during playback and recording to ensure that the replayed audio from all local audio devices **102** is synchronized with previously recorded

Zaxcom, Inc. Exhibit 2062
RØDE Microphones, LLC, et. al v. Zaxcom, Inc.
Page 11 of 40

audio and with the audio replayed by all other local audio devices **102**. In contrast, when recording system **100** is operating in an asynchronous mode, transmission of the master time reference signal allows the files containing recorded audio to be timestamped with the master time reference information to allow the recorded audio to be accurately synchronized post-recording.

RCU transmitter **208** also allows audio generated locally at RCU **104** to be transmitted to the other components of recording system **100**. Such audio may be received from an audio input device such as audio input device **212** via audio input device port **214**. Audio input device **212** may be any type of commercially available audio input device such as a microphone and audio input device port **214** may be any commercially available audio input device port that is compatible with audio input device **212** and the internal components of RCU **104**. The received audio is then buffered and/or amplified by RCU preamp **216** and is converted from analog to digital by RCU ADC **222** such that the audio may be read in digital form by RCU local control unit **210**. This audio may then be processed and sent via RCU transmitter **208** in either analog or digital form. If the audio is to be sent in analog form, RCU local control unit **210** may be equipped with an on-board DAC or an independent DAC may be incorporated in RCU **104** without departing from the scope of the present invention.

[[Similarly, RCU receiver **202** allows audio received from other components of recording system **100** to be played locally at RCU **104**. Such audio may be received in either analog or digital form at RCU receiver **202**. However, if the audio is to be received in analog form, RCU local control unit **210** may be equipped with an on-board ADC or an independent ADC may be incorporated in RCU **104** without departing from the scope of the

present invention. Thereafter, the audio may be processed or relayed directly to RCU DAC **224**, which converts the audio data back to analog form. The analog audio may then be amplified by RCU amp **226** prior to transmission through audio output device port **228** to audio output device **230**. Audio output device **230** may be any type of commercially available audio output device such as headphones, speakers, and the like, and audio output device port **228** may be any commercially available audio output device port that is compatible with audio output device **230** and the internal components of RCU **104**.]]

RCUs **104** may be handheld units such as the RCU **104** depicted in FIG. 2B. In such an embodiment, display **218** may be a small liquid crystal display ("LCD") or the like and keypad **220** may include a plurality of buttons that allow a user to perform functions such as **[[GLENN-WE WOULD LIKE A USER'S MANUAL FOR THE IFB KEYPAD]]** Additionally, handheld embodiments may include any one of a variety of commercially available batteries to function as power supply **206** without departing from the scope of the present invention.

However, non-handheld embodiments of RCU **104** are also envisioned such as tabletop models, personal computer ("PC") models, etc. Also, RCU **104** may be optionally equipped with external interface **252** (FIG. 2A) to facilitate connection of RCU **104** to a PC, laptop PC, dumb terminal, or the like via external interface port **254**. Such an interface allows a user to control the components of recording system **100** via a graphical user interface or other software that may operate on a larger user interface. Such an interface may provide more features and functions than that available on a portable, handheld device such as **[[GLENN-PLEASE PROVIDE EXAMPLES OF FUNCTIONS YOU CAN PERFORM FROM AN EXTERNAL INTERFACE THAT ARE NOT AVAILABLE FROM THE RCU DIRECTLY, IF ANY]]**

Zaxcom, Inc. Exhibit 2062
RØDE Microphones, LLC, et. al v. Zaxcom, Inc.
Page 13 of 40

Turning next to FIG. 3, depicted is a block diagram of one embodiment of local audio device **102** in accordance with the present invention. In one aspect of the present invention, local audio devices **102** are digital, wireless audio transceivers. Such audio devices may be manufactured in the form of body-packs, such as those typically worn by news announcers, performers, and the like. In the depicted embodiment, local audio device **102** includes, *inter alia*, local receiver **302**, local timecode generator **304**, local power supply **306**, local transmitter **308**, local control unit **310**, local audio input device **312**, local audio input device port **314**, local preamp **316**, local display **318**, local keypad **320**, local ADC **322**, local DAC **324**, local amp **326**, local audio output device port **328**, local audio output device **330**, memory **332**, comparator **334**, oscillator **336**, and counter **338**.

Local transmitter **308** also allows audio generated locally at local audio device **102** to be transmitted to the other components of recording system **100**. Such audio may be received from an audio input device such as local audio input device **312** via local audio input device port **314**. Local audio input device **312** may be any type of commercially available audio input device such as a microphone and local audio input device port **314** may be any commercially available audio input device port that is compatible with local audio input device **312** and the internal components of local audio device **102**. The received audio is then buffered and/or amplified by local preamp **316** and is converted from analog to digital by local ADC **322** such that the audio may be read in digital form by local control unit **310**. This audio may then be processed and sent via local transmitter **308** in either analog or digital form. If the audio is to be sent in analog form, local control unit **210** may be equipped with an on-board DAC or an independent DAC may be incorporated in local audio device **102** without departing from the scope of the present invention.

Zaxcom, Inc. Exhibit 2062
RØDE Microphones, LLC, et. al v. Zaxcom, Inc.
Page 14 of 40

Similarly, local receiver **302** allows audio received from other components of recording system **100** to be played locally at local audio device **102**. Such audio may be received in either analog or digital form at local receiver **302**. However, if the audio is to be received in analog form, local control unit **310** may be equipped with an on-board ADC or an independent ADC may be incorporated in local audio device **102** without departing from the scope of the present invention to allow local control unit **310** to receive the audio in digital form. Thereafter, the audio may be processed or relayed directly to local DAC **324**, which converts the audio data back to analog form. The analog audio may then be amplified by local amp **326** prior to transmission through local audio output device port **328** to local audio output device **330**. Local audio output device **330** may be any type of commercially available audio output device such as headphones, speakers, and the like, and local audio output device port **328** may be any commercially available audio output device port that is compatible with local audio output device **330** and the internal components of local audio device **102**.

Memory **332** of local audio device **102** locally stores audio processed by local control unit **310** in one or more audio files. Local control unit **310** receives recordable audio from other components of recording system **100** or local audio input device **312**, which may be worn by the performer and connects to local audio device **102** at local audio input device port **314**. The locally stored audio files include timestamps (e.g., timestamps may be stored in the header of the audio file) that indicate when, during the live audio event, each segment of audio occurred. The timestamps may be generated based upon timecodes created by local timecode generator **304** or based upon master timecodes received wirelessly from a master timecode source through local receiver **302**. Local timecode generator **304** may be

synchronized with the master timecode generator during recording of the live audio event as described in further detail below with respect to FIG. 5. Or, alternatively, the timestamps may be synchronized post-recording as described in further detail below with respect to FIGS. 9 and 10. Simultaneous with the local recording of audio received from local audio input device **312**, this audio may also be transmitted through local transmitter **308** to receiver **106** and/or recorder **108** to allow live recording of the audio event. In this scenario, receiver **106** and/or recorder **108** may simultaneously record a multi-track recording of all of the single tracks of audio received from local audio devices **102**, which are worn by the performers of the live audio event.

Memory **332** may be virtually any type of commercially available removable or non-removable memory including, but not limited to, flash memory cards, compact flash memory cards, Universal Serial Bus ("USB") thumbdisks, and the like. Use of removable memories **332** facilitates removal and insertion of these memories into a PC or the like for electronic combination or mixing of the recorded audio data. Such electronic mixing may be performed via commercially available software such as Pro Tools or the like and may be performed in addition to or in lieu of live wireless recording of the audio event.

Local audio devices **102** also receive non-audio information (e.g., time reference signals, digital commands, audio, etc.) from other components of recording system **100** via local receiver **302**. During synchronous operation of recording system **100**, a portion of the received data may be used to synchronize local timecode generator **304** to the master timecode generator integral to one of the components of recording system **100** (e.g., RCU **104**, recorder **108**, etc.) using a process such as that described below with respect to FIGS. 4A, 4B, and 5 or an equivalent thereof. Alternatively, during asynchronous operation of

Zaxcom, Inc. Exhibit 2062
RØDE Microphones, LLC, et. al v. Zaxcom, Inc.
Page 16 of 40

recording system **100**, the received data may include master timecodes from the master timecode generator that may be used to timestamp individual audio samples and/or files such that the audio received at multiple local audio devices **102** may be synchronized post-recording using one of the methods discussed below with respect to FIGS. 9 and 10 or an equivalent thereof.

As described in further detail below with respect to FIG. 5, local audio devices **102** operating in the synchronous mode may require one or more of comparator **334**, oscillator **336**, and counter **338**. In one aspect of the present invention, oscillator **336** is a 48 kilohertz ("kHz") voltage controlled oscillator. However, alternate embodiments of oscillator **336** may be substituted without departing from the scope of the present invention including but not limited to a high speed clock divided to produce 48 kHz. In the embodiment of the present invention depicted in FIG. 3, oscillator **336** feeds the sample rate input of local ADC **322**, as well as counter **338**, which provides a time reference for local timecode generator **304**. In this configuration, if local ADC **322** is set to operate at 48 kHz, varying the voltage applied to the clock control input of oscillator **336** will proportionately vary the output of oscillator **336** and, consequently, the sample rate of local ADC **322** and the rate at which local timecode generator **304** keeps time.

When local audio devices **102** such as those depicted in FIG. 3 are used in conjunction with recorders **108** that incorporate a single clock to both regulate the speed of the master timecode generator and control the internal recorder ADC sample rate, comparators **334** help maintain synchronization of local audio devices **102** with each other and with recorder **108** by varying the speed of the respective local timecode generators **304** and the sampling rate of the respective local ADCs **322**. As per an algorithm or hardwired

Zaxcom, Inc. Exhibit 2062
RØDE Microphones, LLC, et. al v. Zaxcom, Inc.
Page 17 of 40

logic that duplicates the sequence depicted in FIG. 5, or an equivalent thereof, comparators **334** compare the timecodes generated by the master timecode generator with timecodes generated by the locally timecode generator and, if necessary, increase or decrease the speed of the respective local timecode generator **304** and the sampling rate of the respective local ADC **322** such that these speeds are synchronized with the speed of the master timecode generator and the ADC of recorder **108**. That is, comparators **334** generate, through software or hardware, the voltage that is applied to the clock control input of the respective oscillator **336** that proportionately varies the sample rate of local ADC **322** and the rate at which local timecode generator **304** keeps time as necessary to maintain synchronization with the sample rate of the ADC of recorder **108** and the master timecode generator, respectively. In this manner, all local audio devices **102** and recorder **108** sample at virtually identical sample rates allowing a wireless recorder **108**, or a wireless recorder/receiver combination, to accurately combine multiple independent tracks of audio, wherein each independent track of audio is received from one of the performer's local audio device **102**.

[[GLENN - IT WOULD BE GOOD TO PROVIDE EXAMPLES OF EACH OF THE COMPONENTS OF THE FIGURES INCLUDING MANUFACTURER AND MODEL #S – THE INVENTION WILL NOT BE LIMITED TO THESE ITEMS, IT WOULD JUST PROVIDE A CLEAR UNDERSTANDING OF HOW TO CREATE THE INVENTION]]

Whenever playback of locally recorded audio is required (e.g., to remedy recording errors caused by transmission losses), RCU **104** transmits a digital command to all local audio devices **102** to playback the audio data stored in the respective memories **332** starting with and subsequent to a specific time reference as indicated by a specific timecode. The

Zaxcom, Inc. Exhibit 2062
RØDE Microphones, LLC, et. al v. Zaxcom, Inc.
Page 18 of 40

digital command is received by local receivers **302**, which transmit or relay the command to their respective local control unit **310**. Thereafter, local control units **310** access the data stored in the respective memory **332** and cause this data to be played or transmitted sequentially via local transmitter **308** starting with the data associated with the requested timecode. The use of timecodes and synchronization of local and master timecode generators, as well as local and recorder audio sampling rates, as discussed herein allows multiple local audio devices **102** to replay audio with the exact timing that occurred during the live audio event.

Local audio devices **102** may be bodypacks such as the local audio device **102** depicted in FIG. 3B. In such an embodiment, display **318** may be a small liquid crystal display ("LCD") or the like and keypad **320** may include a plurality of buttons that allow a user to perform functions such as **[[GLENN-WE WOULD LIKE A USER'S MANUAL FOR THE TRX KEYPAD]]** Additionally, handheld embodiments may include any one of a variety of commercially available batteries to function as power supply **306** without departing from the scope of the present invention.

Referring now to FIG. 4A, illustrated is a flow diagram of one embodiment of a process for operation of a recording system such as recording system **100** in synchronous timecode generator mode in accordance with one embodiment of the present invention. Process **400** begins at **402**. For example, at **402**, one or more performers may each don a local audio device, such as local audio device **102** as described with respect to FIGS. 1, 3A, and 3B. Also, a RCU sound engineer may be equipped with a control unit such as RCU **104**. Process **402** then proceeds to **404**.

At **404**, initialization occurs. During initialization, the local control unit such as local

control unit **310** or other form of central processing unit is reset. Thereafter, the local transmitter, local receiver, ADC, DAC, and local timecode generator clock are initialized. The process then proceeds to **406**, at which the sampling rate of the ADC is set. In some embodiments of the present invention, a sample rate of 48 kHz is incorporated.

Next, at **408**, wireless receive channels are established between the local audio device and one or more wireless devices such as RCUs (e.g., RCU **104**), receivers, and audio recorders. To establish the channel, the local receiver of the audio device receives one or more data packets from the remote wireless device and stores the packets in a designated buffer. For example, when establishing wireless communication with a RCU, the local audio device may receive one or more data packets containing information such as a master timecodes, transport status (i.e., transport mode of an audio recorder), and the like. These packet(s) are then stored in an RX buffer (i.e., a reserved segment of memory used to hold data while it is being processed). Process **400** then proceeds to **410**.

At **410**, the local control unit reads the master timecode contained in the RX buffer and jam synchronizes the local timecode generator with the master timecode. The jam sync synchronizes the local audio device with the RCU while allowing the local audio device to supply its own timecode. Local supply of synchronized timecodes ensures proper timing during periods in which the master timecodes cannot be read (e.g., the RCU is temporarily unstable, wireless communication dropouts, etc.).

Next, at **412**, process **400** queries the transport status stored in the RX buffer. If at **412**, the transport status is stop, process **400** returns to **410**. However, if at **412**, the transport status is record, process **400** proceeds to **414**. At **414**, a new audio file is created in memory (e.g., on a flash card) and the newly created file is timestamped. In one aspect

of the present invention, timestamping includes storing the timecode in the file header. Process **400** then proceeds to **416**.

At **416**, the local control unit waits for an audio sample interrupt from the ADC. Once an audio sample interrupt occurs, process **400** proceeds to **418**. At **418**, the audio sample is retrieved from the ADC and stored in the local memory. In one aspect of the present invention, the audio sample is stored in the next available address of the local memory. Next, at **420**, the timecode generator counter is incremented, thereby indicating that the time period for one sample of audio has elapsed.

Process **400** then proceeds to **422**, at which the local control unit transmits the audio sample through the local transmitter to the other wireless devices such as RCUs, receivers, audio recorders, and the like. For example, audio from multiple local audio devices may be transmitted to a multi-track recorder for live recording of the audio event while each local audio device locally records its performer's audio. At **424**, process **400** queries the RF buffer of the local receiver to determine the availability of a new master timecode packet. If at **424**, a new master timecode packet has not been received from the RF receiver, process **400** returns to **416**. However, if at **424**, a new master timecode packet has been received, process **400** proceeds to **426** as depicted in FIG. 4B.

At **426**, process **400** executes a feedback loop algorithm, which modifies the speed of the local timecode generator as necessary to maintain its synchronization with the master timecode generator (e.g., a timecode generator contained within the RCU or master recorder). This algorithm may be implemented using any one of a variety of methods. In one embodiment of the present invention, a feedback loop algorithm, such as process **500** depicted in FIG. 5, modulates a low-pass filtered feedback error voltage that is supplied by

Zaxcom, Inc. Exhibit 2062
RØDE Microphones, LLC, et. al v. Zaxcom, Inc.
Page 21 of 40

the local control unit directly to the local oscillator. The local oscillator then controls the sample rate of the ADC and the speed of the local timecode generator by supplying the feedback error voltage to the ADC's sample rate input and the local timecode generator's clock control input. Alternatively, a comparator independent of the local control unit may perform the comparison of the master timecodes and the local timecodes and may vary the sample rate of the ADC and the speed of the local timecode generator by directly supplying the feedback error voltage to the oscillator. A variety of hardware and software equivalents of this function may be substituted without departing from the scope of the present invention.

Referring now to FIG. 5, the feedback loop algorithm begins at **502**. At **504**, the current local timecode is retrieved from the timecode generator such as local timecode generator **304** and is written to the variable TCgen. Process **500** proceeds to **506**. At **506**, the current master timecode is retrieved from the RX buffer of the local receiver and is written to the variable TCrx and process **500** proceeds to **508**. At **508**, variable TCdiff is calculated by subtracting TCrx from TCgen. Process **500** then proceeds to **510**, at which process **500** compares TCdiff to zero. If, at **510**, TCdiff is less than zero, process **500** proceeds to **512**, at which the feedback error voltage supplied to the local oscillator's DAC by the local control unit is increased above the previously supplied feedback error voltage. The local oscillator's DAC then supplies the new feedback error voltage to the local oscillator, which, in turn, supplies a new clock input voltage to the local timecode generator and a new sample rate input to the ADC. In this manner, the speed of the local timecode generator and the sample rate of the ADC are increased to maintain synchronization with the master timecode generator. However, alternate embodiments of the present invention are envisioned in which only one of either the speed of the local timecode generator or the

Zaxcom, Inc. Exhibit 2062
RØDE Microphones, LLC, et. al v. Zaxcom, Inc.
Page 22 of 40

sample rate of the ADC is modified.

Alternatively, if at **510** TCdiff is not less than zero, process **500** proceeds to **514**, at which TCdiff is analyzed to determine if it is greater than zero. If yes, process **500** proceeds to **516** and the feedback error voltage supplied to the local oscillator's DAC by the local control unit is decreased below the previously supplied feedback error voltage. The local oscillator's DAC then supplies the new feedback error voltage to the local oscillator, which, in turn, supplies a new clock input voltage to the local timecode generator and a new sample rate input to the ADC. In this manner, the speed of the local timecode generator and the sample rate of the ADC are decreased to maintain synchronization with the master timecode generator. However, alternate embodiments of the present invention are envisioned in which only one of either the speed of the local timecode generator or the sample rate of the ADC is modified.

If TCdiff is neither less than zero as determined at **510** or greater than zero as determined at **514**, then TCdiff is equal to zero. In this scenario, the local and master timecode generators are synchronized and, therefore, no adjustment is made to the speed of the local timecode generator. At this point, process **500** ends at **518**.

Although FIG. 5 depicts one method of performing a feedback loop, many variations of this feedback loop may be substituted without departing from the scope of the present invention. For example, the feedback loop may be implemented as a digital phased locked loop that re-samples the audio in a manner that simulates a hardwired feedback loop.

Referring back to FIG. 4B, after execution of the feedback loop algorithm at **426**, process **400** proceeds to **428**. At **428**, the local timecode generator is jam synchronized with the newly received master timecode read from the RX buffer. Next, process **400**

Zaxcom, Inc. Exhibit 2062
RØDE Microphones, LLC, et. al v. Zaxcom, Inc.
Page 23 of 40

optionally proceeds to **430**, at which a timecode is stored as an escape sequence in the next available address of the local memory. The escape sequence stores a master timecode in addition to the locally generated timestamp. This escape sequence may be used post-processing to resample the audio based upon interpolated master timecode data. Process **400** then proceeds to **432**. At **432**, process **400** queries the continuous loop record mode. If at **432** the continuous loop record mode is enabled, process **400** returns to **416** to wait for an audio sample interrupt from the ADC as discussed above. However, if at **432**, the continuous loop record mode has not been enabled, process **400** proceeds to **434**. At **434**, process **400** queries the transport status. If at **434** the transport status is record, process **400** returns to **416** to wait for an audio sample interrupt from the ADC as discussed above. However, if at **434**, the transport status is stop, process **400** returns to **410**, at which process **400** continuously jam synchronizes the local timecode generator with the master timecodes received in the RX buffer until the transport status changes from stop to record at **412**.

Turning next to FIG. 6, illustrated is a flow diagram of one embodiment of a process for recording audio and for replaying and re-recording segments of missed audio in accordance with embodiments of the present invention. Process **600** begins at **602**. For example, at **602**, one or more performers may each don a local audio device, such as local audio device **102** as described with respect to FIG. 2. Process **600** then proceeds to **604**.

At **604**, a master unit, such as RCU **104**, receiver **106**, or recorder **108** transmits master timecodes to each local audio device, and process **600** proceeds to **606**. At **606**, each local audio device synchronizes (e.g., jam syncs) its respective on board local timecode generator with the master timecodes received from the master unit, thereby synchronizing all local audio device timecode generators with the master timecode

Zaxcom, Inc. Exhibit 2062
RØDE Microphones, LLC, et. al v. Zaxcom, Inc.
Page 24 of 40

generator contained within the master unit. Process **600** then proceeds to **608**. At **608**, local audio devices begin locally recording audio received from an audio input device. This audio is stored in the memory of the respective local audio device with timestamps generated by the local timecode generator. Each local audio device also simultaneously transmits its received audio to recorders or receiver/recorder combinations such as receivers **106** and recorders **108** in real time. The audio received from each of the local audio devices (e.g., the local audio device of each performer) may be combined to create a single multi-track audio file that is stored with master timestamps generated by the receiver/recorder's internal master timecode generator.

Process **600** then proceeds to **610**. At **610**, process **600** queries the initiation of audio replay. The initiation of audio replay may be manual or automatic. For example, if a user detects a loss of audio, the user may manually initiate audio replay beginning at the specific timecode reference at which the transmission error occurred. Alternatively, if a loss of audio is automatically detected by the receiving equipment, a playback request may be sent from the receiving equipment to the controlling unit such as a remote control unit. In response, such controlling unit may command the local audio devices to replay or retransmit the missed audio to the receiving equipment beginning at the timecode at which the loss of data occurred.

If, at **610**, audio replay is not initiated either manually or automatically, process **600** returns to **608**. However, if, at **610**, audio replay is initiated, process **600** proceeds to **612**. At **612**, a controlling unit, such as RCU **104**, sends a signal to the local audio devices requesting playback of the stored audio starting at a specific timecode.

Next, at **614**, each local audio device processes the playback command and

Zaxcom, Inc. Exhibit 2062
RØDE Microphones, LLC, et. al v. Zaxcom, Inc.
Page 25 of 40

synchronizes playback to the timecode contained in the playback command. In addition, at least one local audio device transmits the synchronization data to the receiving equipment (e.g., receiver **106**, recorder **108**, etc.) to synchronize recording of the replayed audio. Process **600** then proceeds to **616**. However, in alternate embodiments of the present invention, the receiving equipment and the local audio devices may simultaneously receive the synchronization and time reference data from the transmitting equipment (e.g., the controlling unit).

At **616**, one or more local audio devices transmit, or replay, its respective stored audio starting with the audio that corresponds to the time specified by the timecode. The receiving equipment simultaneously records the replayed audio from each of the local audio devices and stores it within the previously recorded audio according to its timecode data. That is, due to the highly accurate synchronization of all of the components of the recording system, the receiving equipment may insert the replayed audio data that was not recorded during the live audio event due to wireless transmission errors into the original recording at the nearly the exact time at which the missed audio originally occurred, thereby compensating for any transmission losses. Process **600** then proceeds to **618**. At **618**, one or more local audio devices continue to replay audio while the receiving equipment records the audio.

At **620**, process **600** queries the status of audio replay. If, at **620**, the audio has been fully replayed, process **600** proceeds to **608**. At **608**, the local audio devices may record a new audio event or may replay a different segment of recorded data. Otherwise, if, at **620**, all requested audio has not been replayed or re-recorded, process **600** returns to **618**.

Referring now to FIG. 7, illustrated is a flow diagram of one embodiment of a process

Zaxcom, Inc. Exhibit 2062
RØDE Microphones, LLC, et. al v. Zaxcom, Inc.
Page 26 of 40

for operation of a recording system such as recording system **100** in asynchronous timecode generator mode in accordance with one embodiment of the present invention. Process **700** begins at **702**. For example, at **702**, one or more performers may each don a local audio device, such as local audio device **102** as described with respect to FIGS. 1, 3A, and 3B. Also, a RCU sound engineer may be equipped with a control unit such as RCU **104**. Process **702** then proceeds to **704**.

At **704**, initialization occurs. During initialization, the local control unit such as local control unit **310** or other form of central processing unit is reset. Thereafter, the local transmitter, local receiver, ADC, DAC, and clock are initialized. The process then proceeds to **706**, at which the sampling rate of the ADC is set. In some embodiments of the present invention, a sample rate of 48 kHz is incorporated.

Next, at **708**, wireless receive channels are established between the local audio device and one or more wireless devices such as RCUs (e.g., RCU **104**), receivers, and audio recorders. To establish the channel, the local receiver of the audio device receives one or more data packets from the remote wireless device and stores the packets in a designated buffer. For example, when establishing wireless communication with a RCU, the local audio device may receive one or more data packets containing information such as a timecode, transport status (i.e., transport mode of an audio recorder), and the like. These packet(s) are then stored in an RX buffer. Process **700** then proceeds to **710**.

At **710**, the local control unit reads the transport status and the master timecode contained in the RX buffer. Next, at **712**, process **700** queries the transport status. If at **712**, the transport status is stop, process **700** returns to **710**. However, if at **712**, the transport status is record, process **700** proceeds to **714**. At **714**, a new audio file is created

Zaxcom, Inc. Exhibit 2062
RØDE Microphones, LLC, et. al v. Zaxcom, Inc.
Page 27 of 40

in memory (e.g., on a flash card) and the timecode is stored in the header of the newly created file. Process **700** then proceeds to **716**.

At **716**, the local control unit waits for an audio sample interrupt from the ADC. Once an audio sample interrupt occurs, process **700** proceeds to **718**. At **718**, the audio sample is retrieved from the ADC and stored in the local memory. In one aspect of the present invention, the audio sample is stored in the next available address of the local memory. Process **700** then proceeds to **720**, at which the local control unit transmits the audio sample through the local transmitter to the other wireless devices such as RCUs, receivers, audio recorders, and the like.

At **722**, process **700** queries the RF buffer of the local receiver to determine the availability of a new master timecode packet. If at **722**, a new master timecode packet has not been received from the RF receiver, process **700** returns to **716**. However, if at **722**, a new master timecode packet has been received, process **700** optionally proceeds to **724**. At **724**, the timecode is stored as an escape sequence in the next available address of the local memory. Process **700** then proceeds to **726**. At **726**, process **700** queries the continuous loop record mode. If at **726** the continuous loop record mode is enabled, process **700** returns to **716** to wait for an audio sample interrupt from the ADC as discussed above. However, if at **726**, the continuous loop record mode has not been enabled, process **700** proceeds to **728**. At **728**, process **700** queries the transport status. If at **728** the transport status is record, process **700** returns to **716** to wait for an audio sample interrupt from the ADC as discussed above. However, if at **728**, the transport status is stop, process **700** returns to **710**, at which process **700** continuously reads the transport status and master timecodes from the RX buffer until the transport status changes from stop to record at **712**.

Zaxcom, Inc. Exhibit 2062
RØDE Microphones, LLC, et. al v. Zaxcom, Inc.
Page 28 of 40

Operation of the present invention in asynchronous mode allows one or more components of local audio devices such as local audio devices **102** (e.g., local timecode generator, comparator, counter, etc.) to be eliminated in embodiments in which the local audio devices utilize master timecodes generated by the master timecode generator rather than locally generated timecodes.

Referring next to FIG. 8, depicted is multi-disk drive **800** for reading and/or reformatting audio files recorded on a plurality of local audio device memories (e.g., memories **332**). In its simplest form, such as the embodiment depicted in FIG. 8, multi-disk drive **800** includes a plurality of individual disk drives **802** (e.g., flash memory card drives, compact flash memory card drives, USB thumbdisk drives, etc.). Also included is a plurality of disk drive ejector buttons **804**, if required, to allow users to eject a memory located in a specific disk drive **802**. Multi-disk drive **800** may be equipped with an integral user interface or may be connected to an external interface (e.g., a personal computer) to allow the audio files contained on each memory to be manipulated and/or read.

In one aspect of the present invention, the memory of each local audio device such as local audio device **102** may be removed after completion of a live performance, videotaping, etc. Each memory may then be inserted into a corresponding one of disk drives **802**. Thereafter, all of the individual audio files may be combined to provide one comprehensive audio file. Or, alternatively, each audio file may be individually reformatted or otherwise manipulated prior to creation of one comprehensive audio file.

In embodiments of the present invention in which the recording system recorded the live audio event in asynchronous mode, or in which long periods (e.g., 8 hours) of recording occurred, multi-disk drive **800** may be used to resample the audio samples to ensure that

Zaxcom, Inc. Exhibit 2062
RØDE Microphones, LLC, et. al v. Zaxcom, Inc.
Page 29 of 40

each audio file's timestamps are properly synchronized. One example of such as process is illustrated in the flowchart of FIG. 9.

Referring now to FIG. 9, illustrated is a flow diagram of one embodiment of a process for interpolating timestamps for unstamped audio samples (i.e., audio samples that are not associated with a master timecode timestamp) based upon the timestamps of stamped audio samples (i.e., audio samples that are associated with a master timecode timestamp), and resampling the audio samples to include the interpolated timestamps in accordance with embodiments of the present invention. After recording of an audio event, the audio data stored in the memory of the local audio device (e.g., memory **332**) will typically be stored as an audio sample stream wherein approximately one out of every one thousand to one hundred thousand samples includes a timestamp generated by a remote master timecode generator. However, the interval between timestamped audio samples may be greater than the aforementioned interval if the wireless timecode link was less reliable than a standard wireless link.

The resampling process depicted in FIG. 9, and equivalents thereof, analyze the occurrence of the relatively sparse timestamped audio samples to generate a linear interpolation or a best fit curve. This curve is then used to interpolate timestamps for the unstamped audio samples. After the timestamp of each audio sample has been interpolated, the audio samples may then be re-sampled such that the audio samples are now synchronized with samples generated by the master timecode generator. In one aspect of the present invention, the audio samples are resampled based upon the calculated curve to simulate the condition of an ADC whose sample rate input was driven directly by the master timecode generator's source.

Zaxcom, Inc. Exhibit 2062
RØDE Microphones, LLC, et. al v. Zaxcom, Inc.
Page 30 of 40

If all of the audio from all local audio devices is resampled in this manner, each resulting resampled audio file appears as if it was originally sampled with an accurate audio sample clock derived from the master timecode source. This resampling allows each audio file to include a single timestamp that marks the master timecode of the first audio sample of the audio file. Furthermore, since the audio files now appear as if they have been sampled by an extremely accurate audio sample clock, each audio sample's timestamp may be accurately calculated based solely on the audio sample rate and the timestamp of the first audio sample of the audio file. This condition allows the audio files to be formatted and/or stored as a standard timecoded broadcast .WAV file, thereby allowing them to be read, edited, etc. used standard, commercially-available editing systems. That is, the files may be processed in the same manner as if the audio file had been generated by a standard multi-track audio recorder. Such condition allows the present invention to be easily integrated with other industry standard recording equipment.

One such resampling process is illustrated in FIG. 9. Process **900** begins at **902**. For example, at **902**, one or more local audio device memories may be removed from its respective local audio device and may be inserted into a multi-disk drive **800**, or an equivalent thereof. Process **902** then proceeds to **904**.

At **904**, process **900** determines the desired starting and ending timecodes and stores this data in the variables TimeCodeStart and TimeCodeEnd, respectively. The desired starting and ending timecodes may be input by a user or may be suggested or automatically determined by the algorithm. Process **900** then proceeds to **906**. At **906**, a variable, i, is initialized to a value of zero. The variable i corresponds to the position of audio samples or data points in a data array represented by the variable AudioSample[i]. Process

Zaxcom, Inc. Exhibit 2062
RØDE Microphones, LLC, et. al v. Zaxcom, Inc.
Page 31 of 40

900 then proceeds to **908**.

At **908**, process **900** begins an iterative search for the audio file that matches the desired starting timecode of the output file by comparing the value of TimeCodeStart with the value of the timecode of AudioSample[i]. If, at **908**, the value of TimeCodeStart is equal to the value of the AudioSample[i] timecode, process **900** proceeds to **912**. However, if at **908** the value of TimeCodeStart is not equal to the value of the AudioSample[i] timecode, process **900** proceeds to **910**. At **910**, the variable i is increased by a value of one thereby allowing the value located in the next position of the audio sample array to be compared to the value of TimeCodeStart when process **900** returns to **908**.

If the value of TimeCodeStart is equal to the value of the AudioSample[i] timecode, process **900** proceeds to **912**. At **912**, a variable, n, is initialized to a value of one. The variable n is added to the variable i to allow process **900** to continue to traverse the audio sample array while maintaining the location of the audio sample at the starting timecode, which is represented by the variable AudioSample[i]. Process **900** then proceeds to **914**. At **914**, the value of the AudioSample[i+n] timecode is compared to the value of TimeCodeEnd. If at **914**, the value of the AudioSample[i+n] timecode is greater than or equal to the value of TimeCodeEnd, process **900** proceeds to **916**. At **916**, the value of the AudioSample[i+n] timecode is again compared to the value of TimeCodeEnd. If at **914**, the value of the AudioSample[i+n] timecode is greater than the value of TimeCodeEnd, process **900** proceeds to **928**, at which process **900** terminates. However, if at **916**, the value of the AudioSample[i+n] timecode is equal to the value of TimeCodeEnd, process **900** proceeds to **922**.

Conversely, if at **914**, the value of the AudioSample[i+n] timecode is less than the

Zaxcom, Inc. Exhibit 2062
RØDE Microphones, LLC, et. al v. Zaxcom, Inc.
Page 32 of 40

value of TimeCodeEnd, process **900** proceeds to **918**. At **918**, the value of the AudioSample[i+n] timecode is compared to the value of CurrentTimeCodeEscapeSequence. If, at **918**, the value of the AudioSample[i+n] timecode is not equal to the value of TimeCodeEscapeSequence, process **900** proceeds to **920** where the variable n is increased by one and process **900** returns to **914**. However, if at **918**, the value of the AudioSample[i+n] timecode is equal to the value of TimeCodeEscapeSequence, process **900** proceeds to **922**.

At **922**, the average time period "T" that elapsed between the audio samples that occurred between AudioSample[i] and AudioSample[i+n] may be calculated by subtracting the value of the timecode of AudioSample[i] from the value of the timecode of AudioSample[i+n] and dividing by n, wherein n is now equivalent to the number of audio samples that occurred between the current timestamped audio sample and the previous timestamped audio sample. Process **900** then proceeds to **924**. At **924**, AudioSamples[i] through AudioSamples[i+n] are re-sampled at any desired sample rate based upon the value of T as calculated in **922**, or any other desired sample rate, using an audio resampling algorithm (e.g., linear interpolation). Process **900** then proceeds to **926**, at which the variable i is set to a value equal to the current value of i plus the current value of n and process **900** returns to **912**. The iterative process continues until the value of the AudioSample[i+n] timecode is greater than the value of TimeCodeEnd, whereby process **900** proceeds to **928**, at which process **900** terminates.

A similar interpolation algorithm, such as the algorithm depicted in FIG. 10, may be incorporated to break down single large audio files (e.g., an audio file recording the filming of multiple movie takes over a continuous eight hour period as a single eight-hour audio file)

Zaxcom, Inc. Exhibit 2062
RØDE Microphones, LLC, et. al v. Zaxcom, Inc.
Page 33 of 40

into smaller, more useful files (e.g., one audio file per take). These smaller files will allow the audio recorded locally by the local audio devices to be more easily matched or synchronized with the individual audio files recorded by a master recorder such as recorder **108**.

In one use of an embodiment of the present invention, multiple local audio devices store audio samples with wirelessly-received timecode and transport status samples continuously for the entire duration of the work day (e.g., an 8 hour period). In a typical scenario, while the local audio devices is recording continuously, a technician intermittently records segments of the live, eight-hour audio event. For example, in a film setting, each segment would typically represent a movie 'take' and might range from one to five minutes in duration. Consequently, the master recorder generates individual audio files (i.e., at least one audio file for each recorded segment such as a movie take), whereas the each local audio device generates one massive audio file. Therefore, there is a need for a method of segmenting each large local audio file into smaller audio files that correspond to the segments recorded by the master recorder.

The segmentation method (i.e., the method of segmenting the large local audio devices' files to match the multiple, smaller master recorder's audio file) requires knowledge of which portions of the single local audio device audio file are important and which portions can be discarded. This information can be inferred from the transport status of the master recorder since it is typically operated by someone with this knowledge. Therefore, when the transport status of the master recorder changes from stop to record, it can be inferred that a new master recorder audio file begins, and, subsequently, when the transport status of the master recorder changes from record to

Zaxcom, Inc. Exhibit 2062
RØDE Microphones, LLC, et. al v. Zaxcom, Inc.
Page 34 of 40

stop, it can be inferred that the same master recorder audio file has ended. In addition, when the transport status of the master recorder remains in the stop mode, it can be inferred that the audio recorded by the local audio device during this time period may be discarded. This audio may be discarded post-processing as per algorithms such as that depicted in FIG. 10 or during live recording.

In embodiments of the present invention in which such data is discarded during live recording, the transport status and master timecode of the master recorder are wirelessly transmitted to the local audio devices. This information may be processed by the local audio devices to allow them to create a new audio file with the current master timecode of the master recorder whenever the received transport status and master timecode indicate that the transport status has changed from stop to record. Similarly, the local audio devices may end the newly created audio file when the received transport status indicates that it has changed from record to stop. In this scenario, the resulting local audio device files will automatically be segmented and will each be marked with a master timestamp at the beginning of each file.

However, in embodiments of the present invention in which unimportant audio is not discarded during live recording and, therefore, one large audio file is created, the large audio file may be segmented as per a process such as process **1000** as illustrated in FIG. 10. Process **1000** begins at **1002** at which one or more local audio devices have continuously recorded a lengthy quantity of audio data. Process **1000** then proceeds to **1004**.

At **1004**, a copy of the audio file directory containing the segmented audio files that correspond to the same time period as the local audio device's single large audio file is

Zaxcom, Inc. Exhibit 2062
RØDE Microphones, LLC, et. al v. Zaxcom, Inc.
Page 35 of 40

obtained from the master recorder. Process **1000** then proceeds to **1006**. At **1006**, a variable y is initialized to a value of zero. The variable y corresponds to the number of each file contained in the audio file directory copied from the master recorder. Process **1000** then proceeds to **1008**, at which the variable y is increased by one and a variable x is initialized to a value of one. The variable x corresponds to the position of each audio sample within a particular file. Process **1000** then proceeds to **1010**, at which the copied audio file directory is queried to determine if a file[y] (i.e., the file named with the number that corresponds to the value of y) exists in the audio file directory. If no, process **1000** proceeds to **1028** and terminates.

If file[y] does exist, process **1000** proceeds to **1012**, at which process **1000** determines the starting and ending timecodes for file[y] and stores them in the variables TimeCodeStart and TimeCodeEnd, respectively. Process **1000** then proceeds to **1014**, at which process **1000** compares the value of TimeCodeStart to the value of the timecode associated with AudioSample[x] stored in the memory of the local audio device. If at **1014** the value of TimeCodeStart is not equal to the value of the timecode associated with AudioSample[x], process **1000** proceeds to **1016**. At **1016**, the variable x is increased by one and process **1000** returns to **1014**. In this manner, TimeCodeStart is compared to each consecutive AudioSample[x] until the AudioSample timestamped with a value equal to TimeCodeStart is found. In some embodiments of the present invention, process **1000**, or an equivalent thereof, is performed after process **900**, or an equivalent thereof, to ensure that each of the audio samples has a timestamp (e.g., an interpolated timestamp).

When the AudioSample[x] having a timecode equivalent to TimeCodeStart is found at **1014**, process **1000** proceeds to **1018**. At **1018**, AudioSample[x] is extracted and process

Zaxcom, Inc. Exhibit 2062
RØDE Microphones, LLC, et. al v. Zaxcom, Inc.
Page 36 of 40

1000 proceeds to **1020**, at which the variable x is increased by one and process **1000** proceeds to **1022**. At **1022**, process **1000** compares the value of TimeCodeEnd to the value of the timecode associated with AudioSample[x]. If at **1022**, the value of TimeCodeEnd is not equal to the value of the AudioSample[x] timecode, process **1000** returns to **1018**, whereupon audio samples are consecutively extracted until the timecode of the current AudioSample[x] equals TimeCodeEnd. If, at **1022**, the value of TimeCodeEnd is equal to the value of the timecode of AudioSample[x], process **1000** proceeds to **1024**, at which the final AudioSample[x] of the segmented audio file is extracted and the audio file is saved at **1026**.

Process **1000** then proceeds to **1008**, at which the variable y is increased by one and process **1000** proceeds to **1010** at which the audio file directory is queried to determine the existence of file [y]. If file[y] exists, process **1000** proceeds to **1012** and it continues thereafter as described above. However, if at **1010**, it is determined that file[y] does not exist, process **1000** proceeds to **1028**, at which it terminates.

Although several processes have been disclosed herein as software, it is appreciated by one of skill in the art that the same processes, functions, etc. may be performed via hardware or a combination of hardware and software. Similarly, although the present invention has been disclosed with respect to wireless systems, these concepts may be applied to hardwired systems and hybrid hardwired and wireless systems without departing from the scope of the present invention.

While the present invention has been described with reference to one or more embodiments, which embodiments have been set forth in considerable detail for the purposes of making a complete disclosure of the invention, such embodiments are merely

Zaxcom, Inc. Exhibit 2062
RØDE Microphones, LLC, et. al v. Zaxcom, Inc.
Page 37 of 40

exemplary and are not intended to be limiting or represent an exhaustive enumeration of all aspects of the invention. The scope of the invention, therefore, shall be defined solely by the following claims. Further, it will be apparent to those of skill in the art that numerous changes may be made in such details without departing from the spirit and the principles of the invention.

Zaxcom, Inc. Exhibit 2062
RØDE Microphones, LLC, et. al v. Zaxcom, Inc.
Page 38 of 40

CLAIMS

VIRTUAL WIRELESS MULTITRACK RECORDING SYSTEM

ABSTRACT

Zaxcom, Inc. Exhibit 2062
RØDE Microphones, LLC, et. al v. Zaxcom, Inc.
Page 40 of 40