

## **WIRELESS MULTITRACK RECORDING SYSTEM**

### **DETAILED DESCRIPTION**

As required, a detailed illustrative embodiment of the present invention is disclosed  
5 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  
10 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  
15 events, such as performances, movie takes, etc. having one or more speakers. 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 created by each individual component of  
recording system **100** are accurate within 1 millisecond ("ms") or better of the time  
20 references 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** 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 a radio frequency ("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<sup>®</sup> 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 timecode 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 timecode timestamps that are synchronized with the timecode 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 timestamp 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 an accuracy of 1 ms or better for each performer with respect to all other performers 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 timestamp 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 timestamp 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? WE HAVE THE SAME ISSUE IF THIS IS THE FIRST DEVICE 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 receiver **202**,

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  
5 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 Society of Motion Picture and Television Engineers ("SMPTE") timecode signal  
10 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, the name of the director equipped with RCU **104**, 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  
15 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  
20 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 timecode numbers that are accurate within 1 ms or better as compared to all other local audio devices **102** and/or all other components of recording system **100**. The timecode number is then used during playback and recording to ensure that the replayed audio from all local audio devices **102** is synchronized with previously recorded 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 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-CAN WE GET A USER'S MANUAL OR SPECS FOR THE IFB TO ENTER FUNCTIONS?]]** 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]]**

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 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  
5 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.

Similarly, local receiver **302** allows audio received from other components of  
10 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  
15 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  
20 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**.

**DURING ASYNCHRONOUS MODE, NO LOCAL TIME GENERATOR,  
5 COMPARATOR, ETC**

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,  
10 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 card 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.

15 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  
20 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

each audio file's timecodes 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 unrecorded master timecodes of recorded audio samples based upon the recorded master timecodes, and resampling the audio samples to include the interpolated timecode data 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 timecode timestamp. 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 timecodes for the unstamped audio samples (i.e., the audio samples that occur between timestamped audio samples). After the occurrence of each audio sample has been interpolated, the audio samples may then be re-sampled such that the audio samples are now synchronized with the master timecode samples. 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.

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 timecode timestamp that marks the 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 timecode may  
5 be accurately calculated based solely on the audio sample rate and the timecode stamp 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 files, thereby allowing them to be read, edited, etc. used standard, commercially-available editing systems as if the audio file had been generated by a standard multi-track audio recorder. Such condition allows the  
10 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**.

15 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  
20 samples or data points in a data array represented by the variable AudioSample[i]. Process **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 AudioSample[i]. If, at **908**, the value of TimeCodeStart is equal to the value of AudioSample[i], process **900** proceeds to **912**. However, if at **908** the value of TimeCodeStart is not equal to the value of AudioSample[i], 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 AudioSample[i], 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 AudioSample[i+n] is compared to the value of TimeCodeEnd. If at **914**, the value of AudioSample[i+n] is greater than or equal to the value of TimeCodeEnd, process **900** proceeds to **916**. At **916**, the value of AudioSample[i+n] is again compared to the value of TimeCodeEnd. If at **914**, the value of AudioSample[i+n] is greater than the value of TimeCodeEnd, process **900** proceeds to **928**, at which process **900** terminates. However, if at **916**, the value of AudioSample[i+n] is equal to the value of TimeCodeEnd, process **900** proceeds to **922**.

Conversely, if at **914**, the value of AudioSample[i+n] is less than the value of TimeCodeEnd, process **900** proceeds to **918**. At **918**, the value of AudioSample[i+n] is compared to the value of CurrentTimeCodeEscapeSequence. If, at **918**, the value of AudioSample[i+n] 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 AudioSample[i+n] 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 time stamp of AudioSample[i] from the value of the time stamp 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 AudioSample[i+n] 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) 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 were recording continuously, a technician was intermittently recording 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 generated individual audio files (i.e., at least one audio file for each recorded segment such as a movie take), whereas the local audio devices generated one massive audio file. Therefore, a need occurs to break the large local audio devices' files into smaller files that correspond to the segments recorded by the master recorder.

The segmentation method (i.e., the method of segmenting the large local audio device files to match the multiple, smaller master recorder audio files) 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 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 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 timecode of the master recorder whenever the received transport status and 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 timecode timestamp at the beginning of each file.

However, in embodiments of the present invention in which unneeded 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 obtained from the master recorder. Process **1000** then proceeds to **1006**. At **1006**, a y variable is initialized to a value of zero and an x variable is initialized to a value of one. The y variable corresponds to the number of each file contained in the audio file directory copied from the master recorder and the x variable corresponds to the position of each audio sample with an array of audio samples. Process **1000** then proceeds to **1008**, at which the

variable y is increased by one and process **1000** proceeds to **1010**. At **1010**, 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 timecode (e.g., an interpolated timecode).

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 **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 AudioSample[x], 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 as same  
5 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  
10 exist, process **1000** proceeds to **1028**, at which it terminates.

#### **ADD ALTHOUGH SOFTWARE, CAN BE HARDWARE**

**Recording system may include other wireless components; wireless or non-wireless systems or hybrid systems**

15 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 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  
20 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.

