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(54) AUDIO SYSTEM

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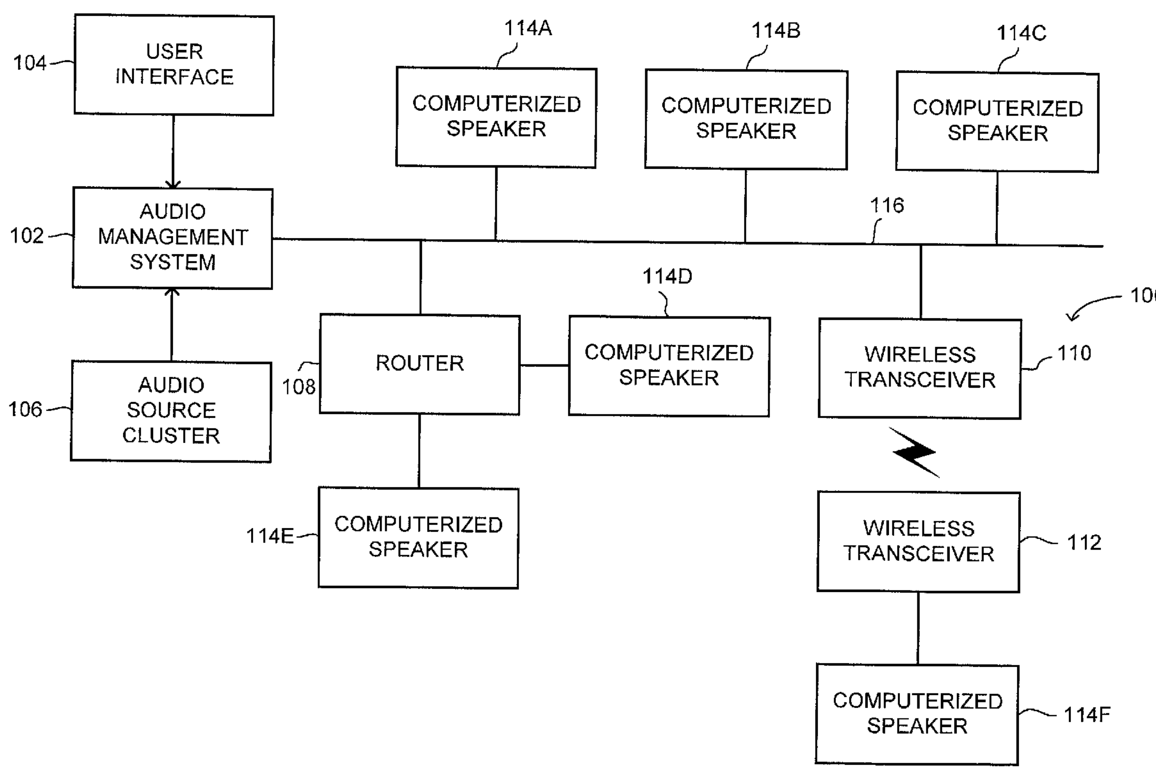
(57) ABSTRACT

Audio system for operating a plurality of speakers, the audio system including an audio management system connected to the speakers via a network, a user interface connected to the audio management system, and an audio source cluster connected to the audio management system, wherein the audio management system operates the speakers by transmitting audio streams and speaker audio control data to the network, and wherein the audio management system determines the speaker audio control data, according to a plurality of parameters.



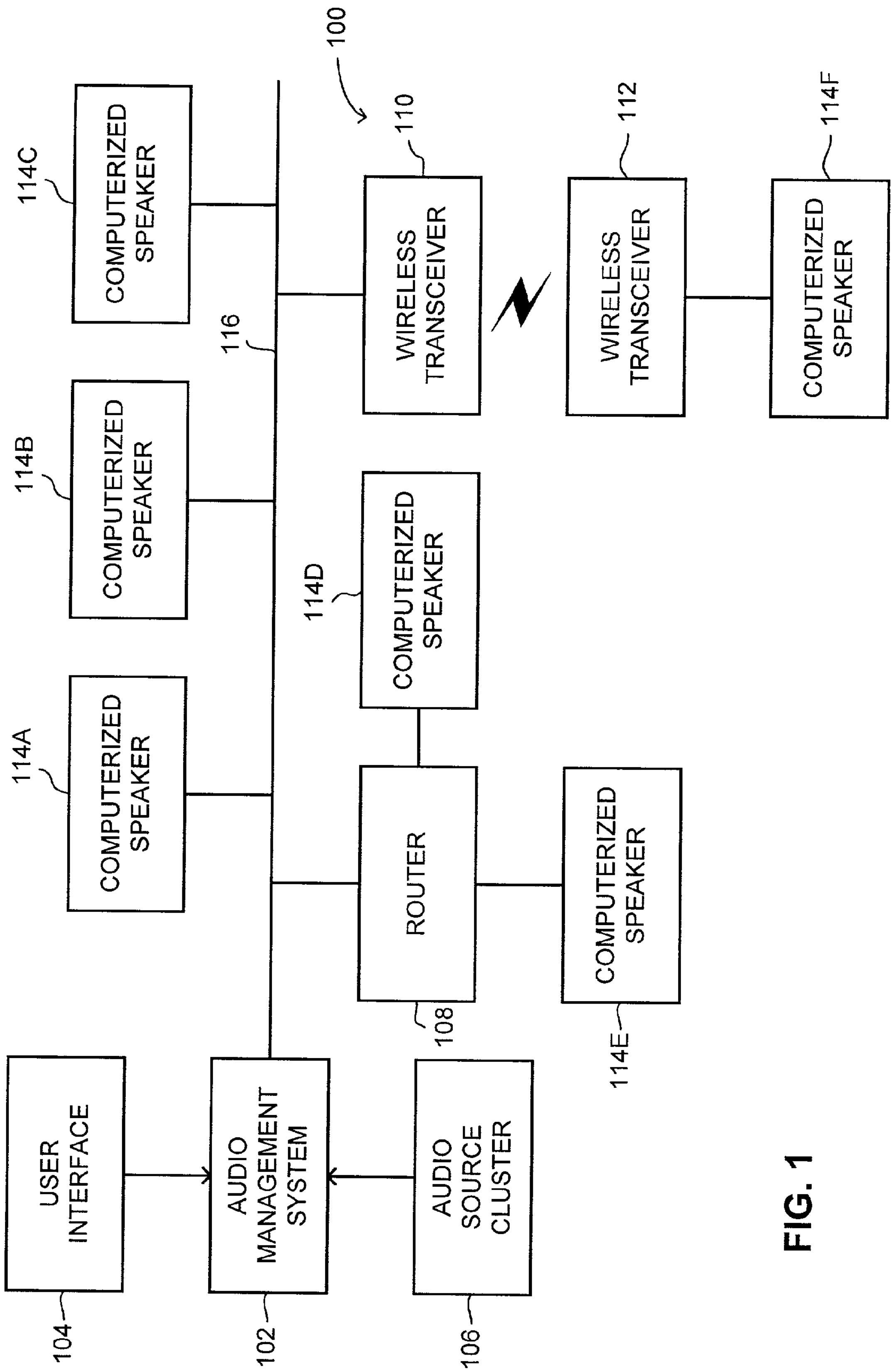

100
COMPUTERIZED SPEAKER
114C
COMPUTERIZED SPEAKER
114B
COMPUTERIZED SPEAKER
114A
116
WIRELESS TRANSCEIVER
110
WIRELESS TRANSCEIVER
112
COMPUTERIZED SPEAKER
114F
COMPUTERIZED SPEAKER
114D
ROUTER
108
COMPUTERIZED SPEAKER
114E
USER INTERFACE
104
AUDIO MANAGEMENT SYSTEM
102
AUDIO SOURCE CLUSTER
106

FIG. 1

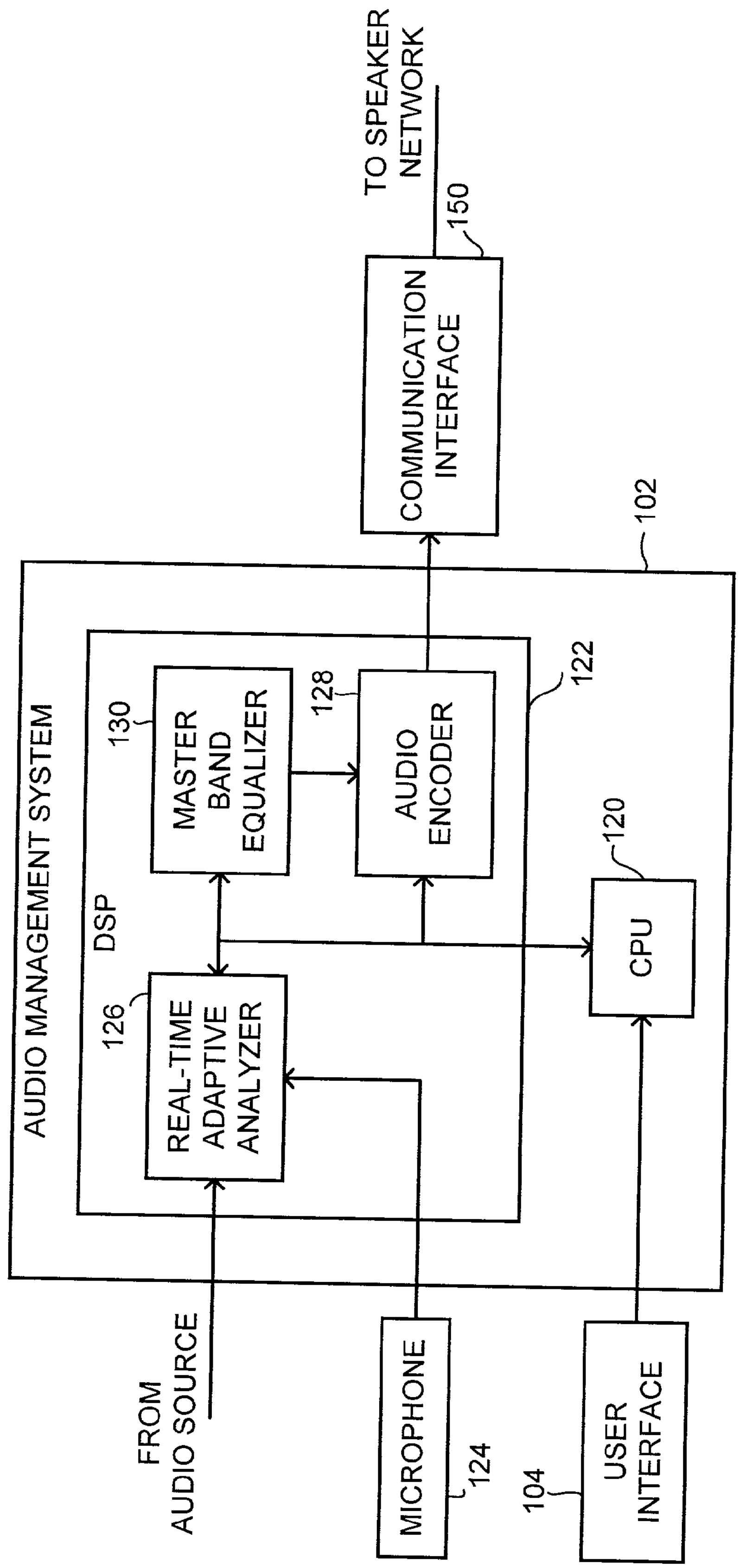

TO SPEAKER NETWORK
150
COMMUNICATION INTERFACE
102
AUDIO MANAGEMENT SYSTEM
DSP
122
130
MASTER BAND EQUALIZER
128
AUDIO ENCODER
120
CPU
126
REAL-TIME ADAPTIVE ANALYZER
FROM AUDIO SOURCE
MICROPHONE
124
104
USER INTERFACE

FIG. 2

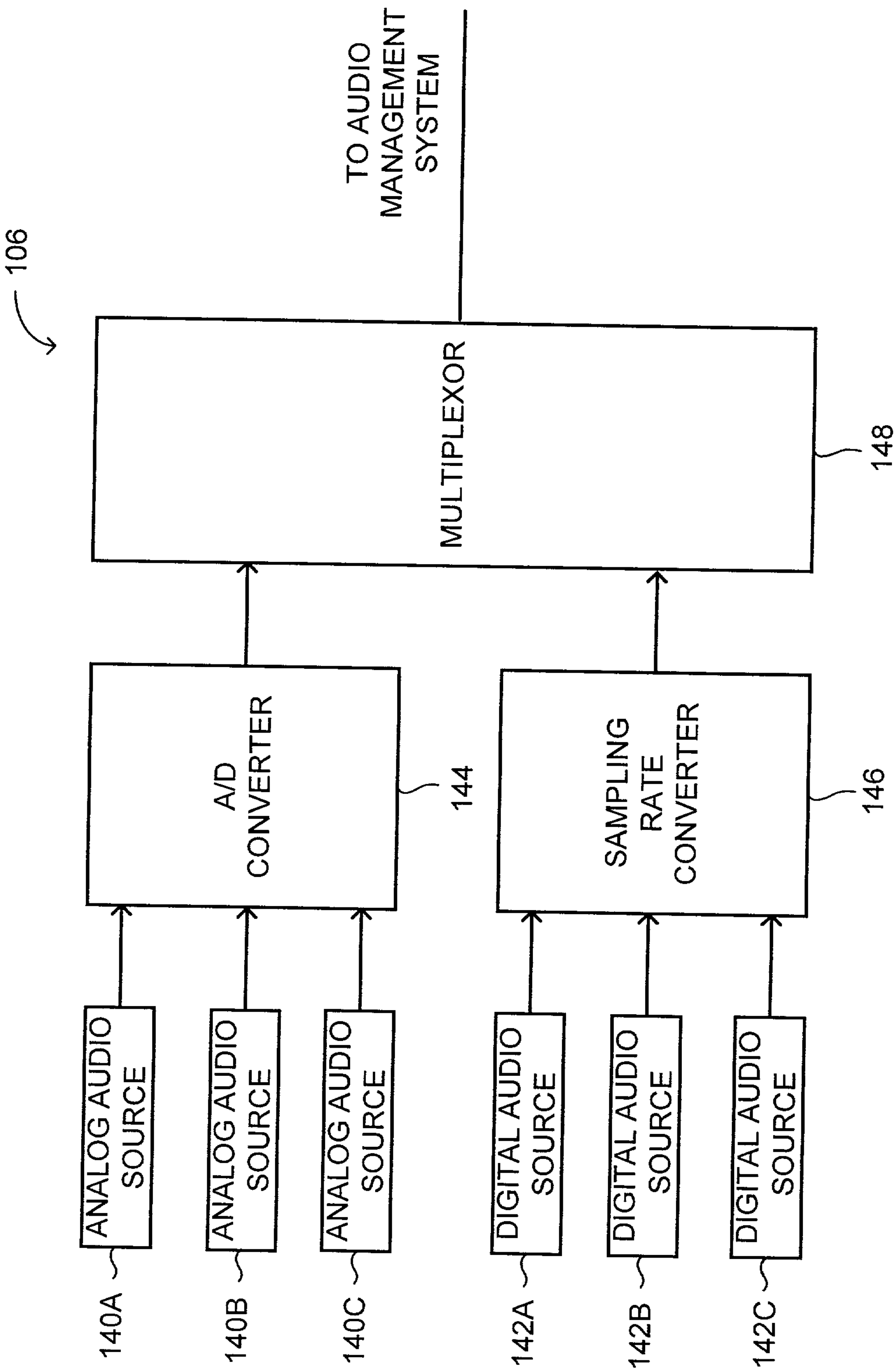

TO AUDIO MANAGEMENT SYSTEM
106
MULTIPLEXOR
148
A/D CONVERTER
144
SAMPLING RATE CONVERTER
146
ANALOG AUDIO SOURCE
140A
ANALOG AUDIO SOURCE
140B
ANALOG AUDIO SOURCE
140C
DIGITAL AUDIO SOURCE
142A
DIGITAL AUDIO SOURCE
142B
DIGITAL AUDIO SOURCE
142C

FIG. 3

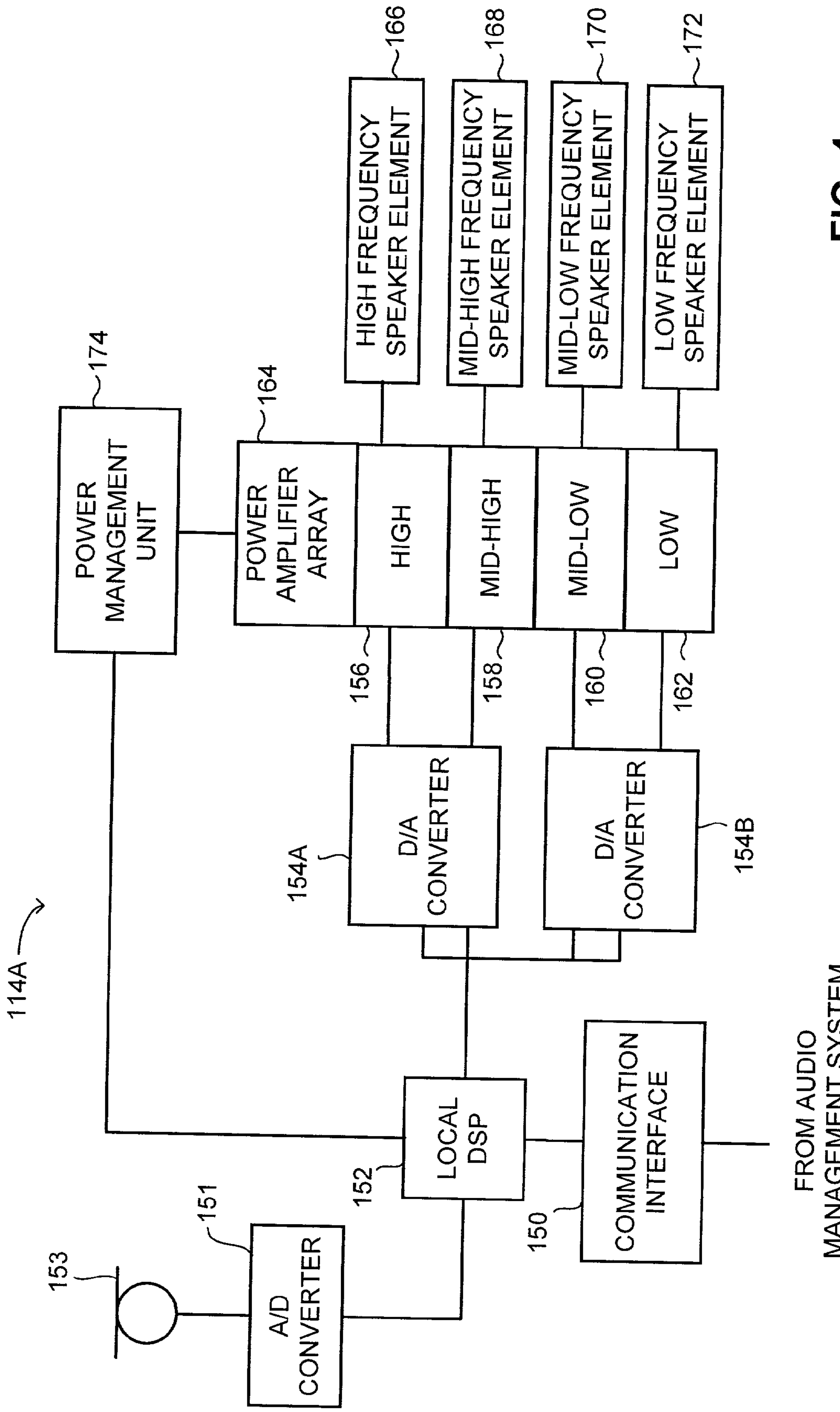

114A
153
151
A/D CONVERTER
152
LOCAL DSP
150
COMMUNICATION INTERFACE
FROM AUDIO MANAGEMENT SYSTEM
174
POWER MANAGEMENT UNIT
154A
D/A CONVERTER
154B
D/A CONVERTER
156
158
160
162
164
POWER AMPLIFIER ARRAY
HIGH
MID-HIGH
MID-LOW
LOW
166
HIGH FREQUENCY SPEAKER ELEMENT
168
MID-HIGH FREQUENCY SPEAKER ELEMENT
170
MID-LOW FREQUENCY SPEAKER ELEMENT
172
LOW FREQUENCY SPEAKER ELEMENT

FIG. 4

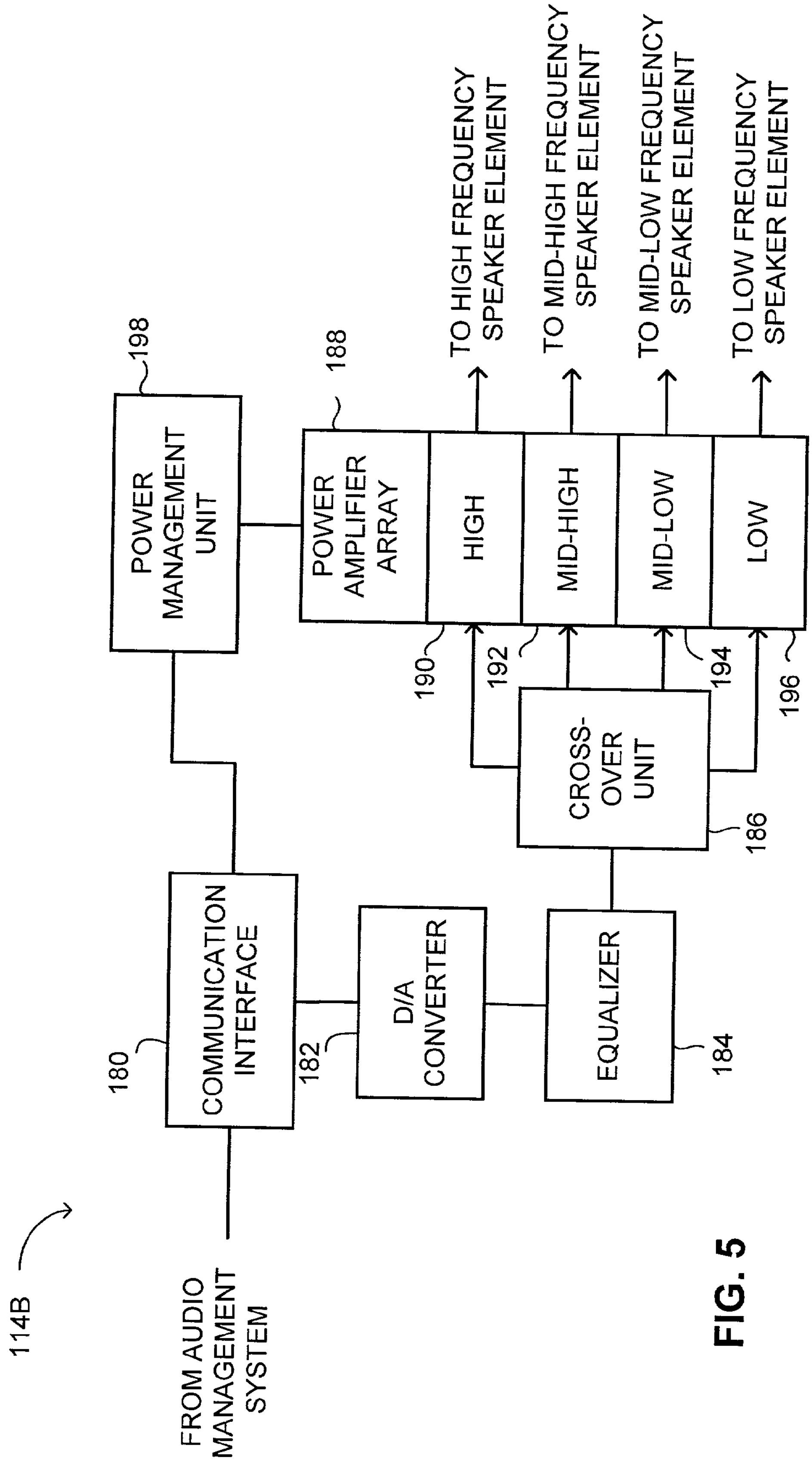

114B
FROM AUDIO MANAGEMENT SYSTEM
180
COMMUNICATION INTERFACE
182
D/A CONVERTER
184
EQUALIZER
186
CROSS-OVER UNIT
198
POWER MANAGEMENT UNIT
188
POWER AMPLIFIER ARRAY
190
HIGH
192
MID-HIGH
194
MID-LOW
196
LOW
TO HIGH FREQUENCY SPEAKER ELEMENT
TO MID-HIGH FREQUENCY SPEAKER ELEMENT
TO MID-LOW FREQUENCY SPEAKER ELEMENT
TO LOW FREQUENCY SPEAKER ELEMENT

FIG. 5

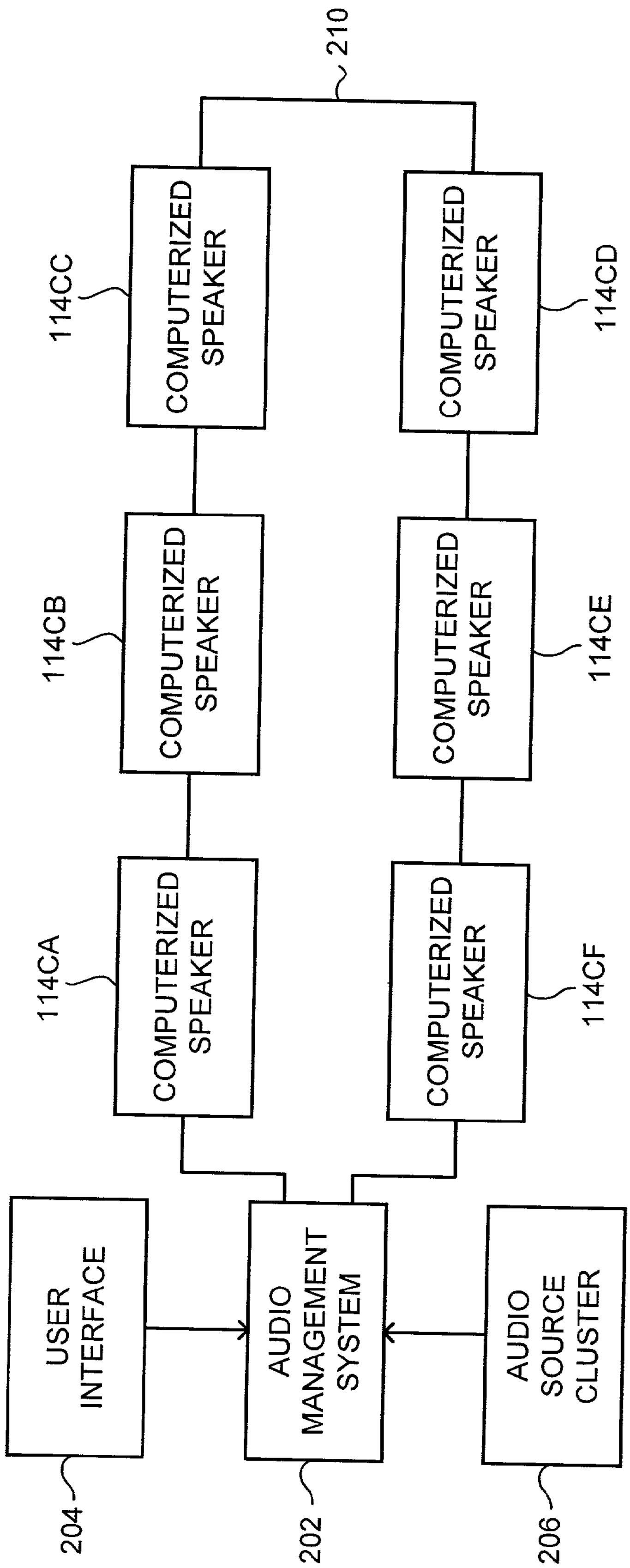
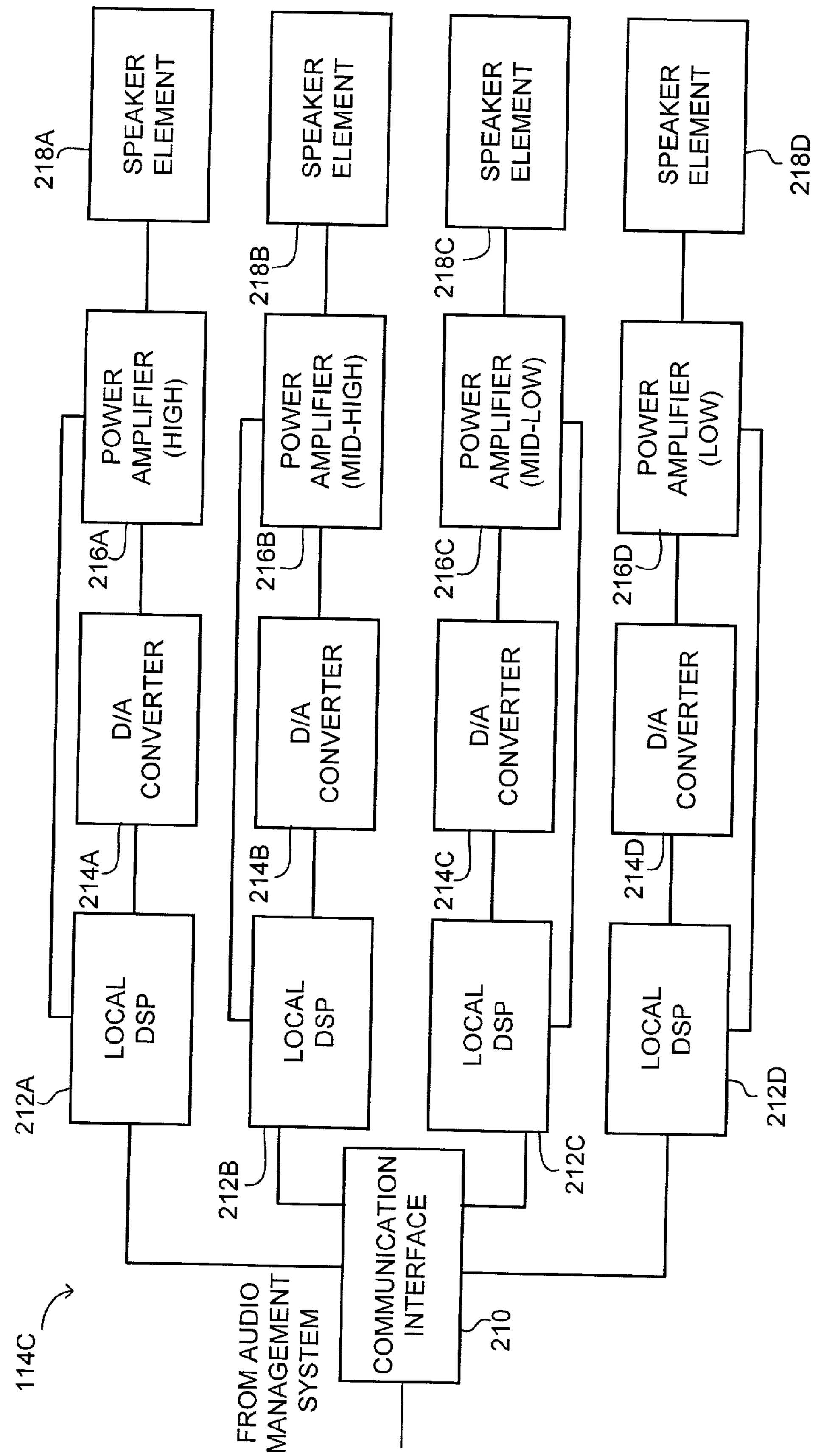

210
200
114CC
COMPUTERIZED SPEAKER
114CD
COMPUTERIZED SPEAKER
114CB
COMPUTERIZED SPEAKER
114CE
COMPUTERIZED SPEAKER
114CA
COMPUTERIZED SPEAKER
114CF
COMPUTERIZED SPEAKER
USER INTERFACE
AUDIO MANAGEMENT SYSTEM
AUDIO SOURCE CLUSTER
204
202
206

FIG. 6



114C
FROM AUDIO MANAGEMENT SYSTEM
COMMUNICATION INTERFACE
210
LOCAL DSP
212A
212B
212C
212D
D/A CONVERTER
214A
214B
214C
214D
POWER AMPLIFIER (HIGH)
POWER AMPLIFIER (MID-HIGH)
POWER AMPLIFIER (MID-LOW)
POWER AMPLIFIER (LOW)
216A
216B
216C
216D
SPEAKER ELEMENT
218A
218B
218C
218D


FIG. 7

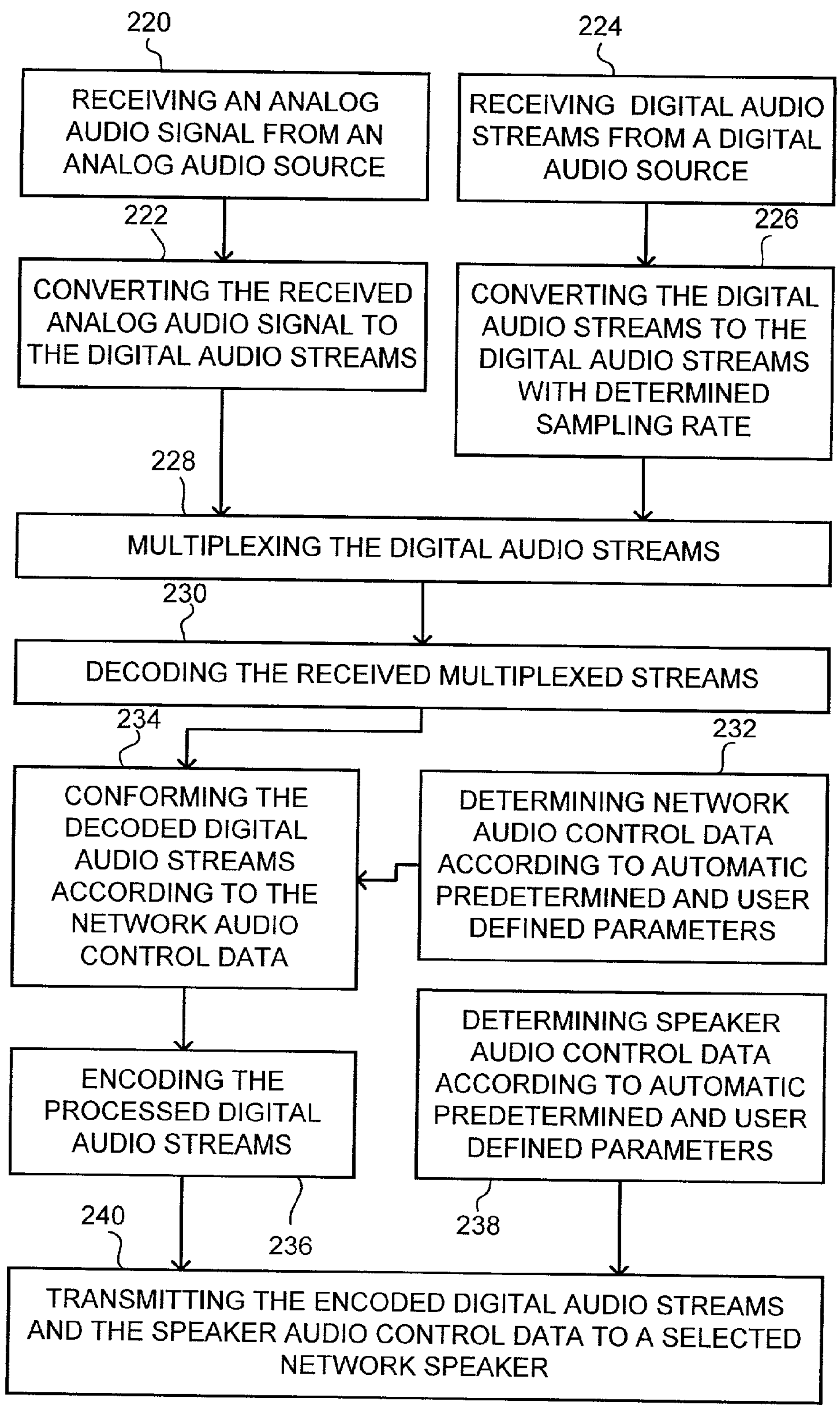

220
RECEIVING AN ANALOG AUDIO SIGNAL FROM AN ANALOG AUDIO SOURCE
224
RECEIVING DIGITAL AUDIO STREAMS FROM A DIGITAL AUDIO SOURCE
222
CONVERTING THE RECEIVED ANALOG AUDIO SIGNAL TO THE DIGITAL AUDIO STREAMS
226
CONVERTING THE DIGITAL AUDIO STREAMS TO THE DIGITAL AUDIO STREAMS WITH DETERMINED SAMPLING RATE
228
MULTIPLEXING THE DIGITAL AUDIO STREAMS
230
DECODING THE RECEIVED MULTIPLEXED STREAMS
234
CONFORMING THE DECODED DIGITAL AUDIO STREAMS ACCORDING TO THE NETWORK AUDIO CONTROL DATA
232
DETERMINING NETWORK AUDIO CONTROL DATA ACCORDING TO AUTOMATIC PREDETERMINED AND USER DEFINED PARAMETERS
ENCODING THE PROCESSED DIGITAL AUDIO STREAMS
DETERMINING SPEAKER AUDIO CONTROL DATA ACCORDING TO AUTOMATIC PREDETERMINED AND USER DEFINED PARAMETERS
240
236
238
TRANSMITTING THE ENCODED DIGITAL AUDIO STREAMS AND THE SPEAKER AUDIO CONTROL DATA TO A SELECTED NETWORK SPEAKER

FIG. 8

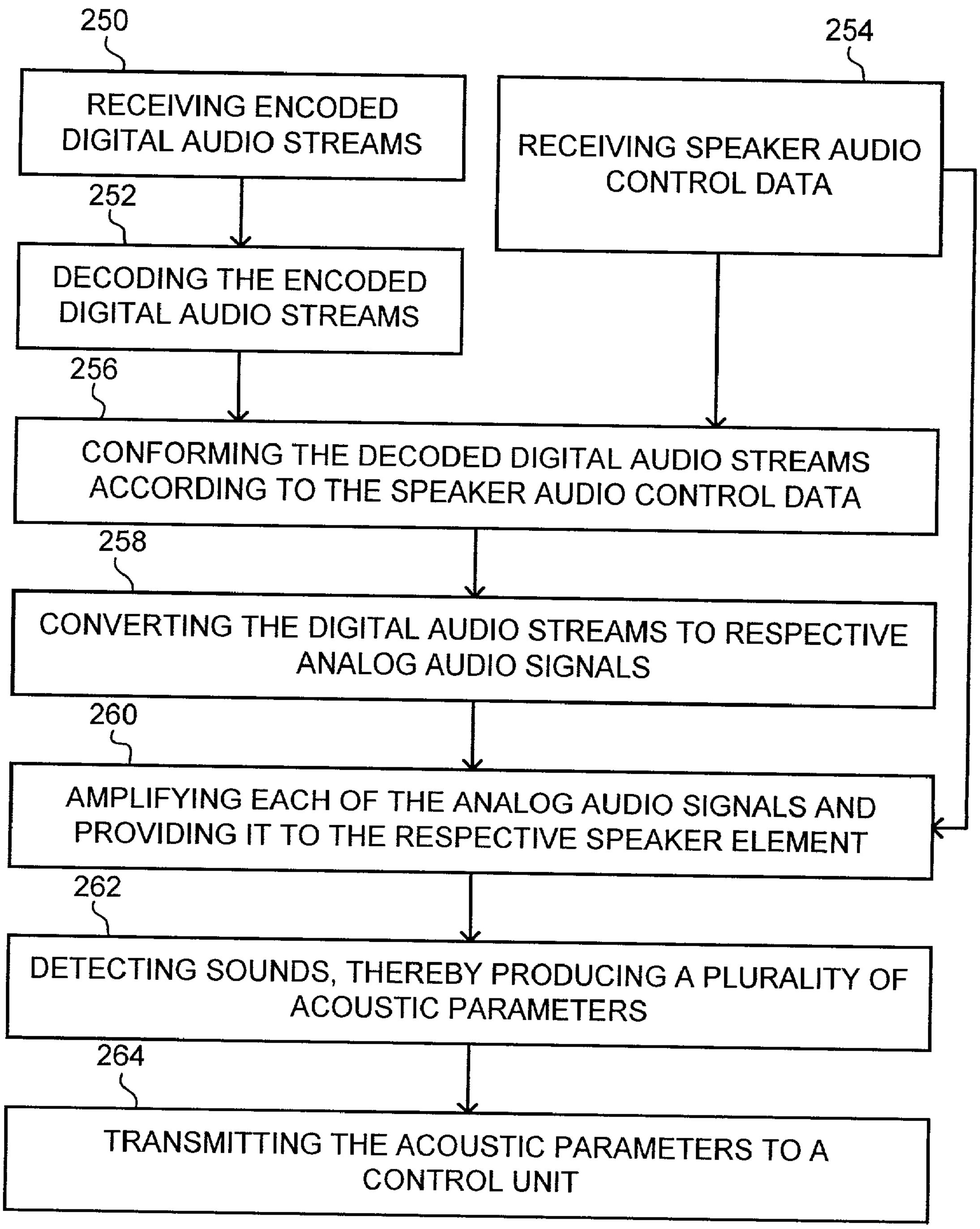

250
RECEIVING ENCODED DIGITAL AUDIO STREAMS
254
RECEIVING SPEAKER AUDIO CONTROL DATA
252
DECODING THE ENCODED DIGITAL AUDIO STREAMS
256
CONFORMING THE DECODED DIGITAL AUDIO STREAMS ACCORDING TO THE SPEAKER AUDIO CONTROL DATA
258
CONVERTING THE DIGITAL AUDIO STREAMS TO RESPECTIVE ANALOG AUDIO SIGNALS
260
AMPLIFYING EACH OF THE ANALOG AUDIO SIGNALS AND PROVIDING IT TO THE RESPECTIVE SPEAKER ELEMENT
262
DETECTING SOUNDS, THEREBY PRODUCING A PLURALITY OF ACOUSTIC PARAMETERS
264
TRANSMITTING THE ACOUSTIC PARAMETERS TO A CONTROL UNIT

FIG. 9

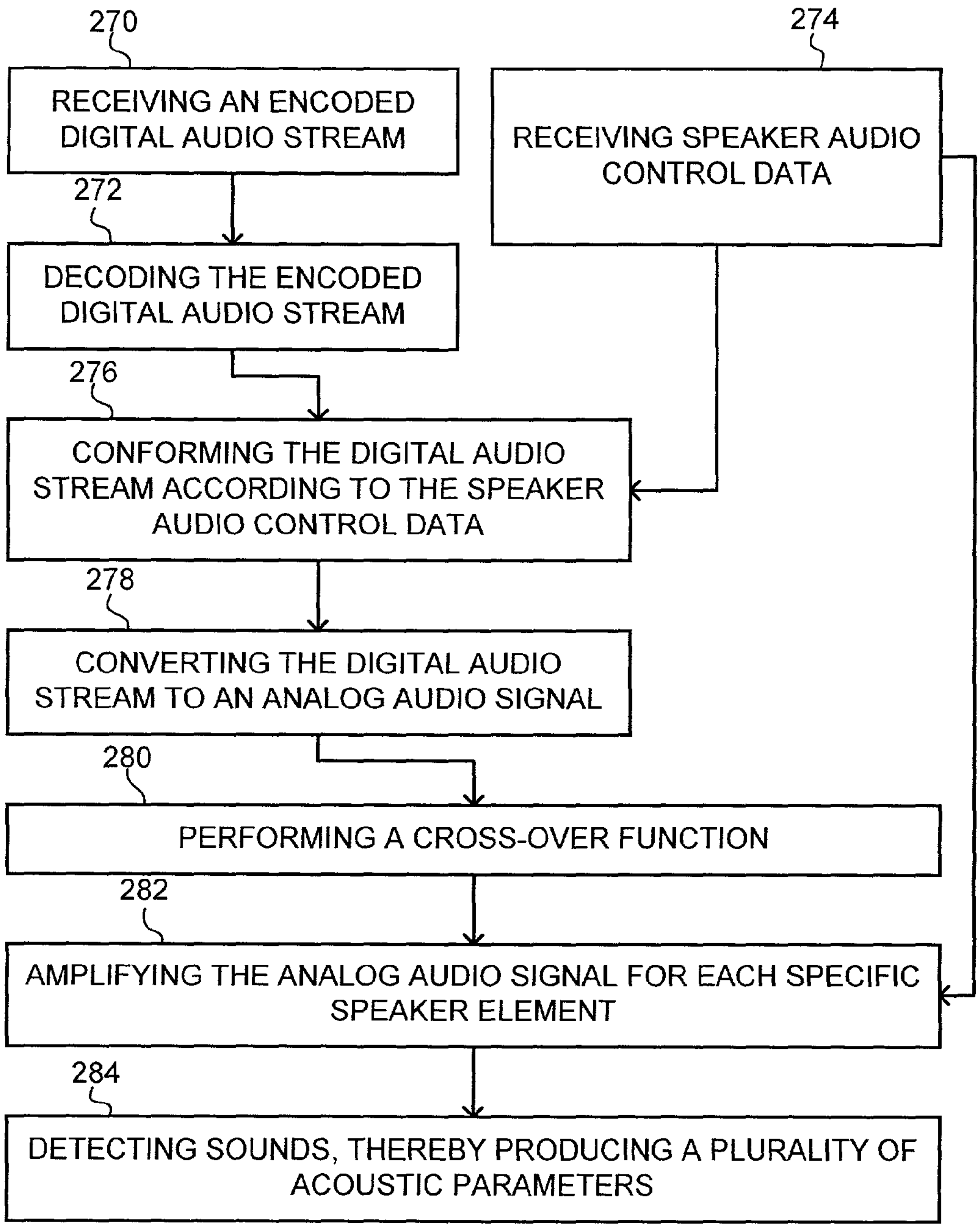

270
RECEIVING AN ENCODED DIGITAL AUDIO STREAM
274
RECEIVING SPEAKER AUDIO CONTROL DATA
272
DECODING THE ENCODED DIGITAL AUDIO STREAM
276
CONFORMING THE DIGITAL AUDIO STREAM ACCORDING TO THE SPEAKER AUDIO CONTROL DATA
278
CONVERTING THE DIGITAL AUDIO STREAM TO AN ANALOG AUDIO SIGNAL
280
PERFORMING A CROSS-OVER FUNCTION
282
AMPLIFYING THE ANALOG AUDIO SIGNAL FOR EACH SPECIFIC SPEAKER ELEMENT
284
DETECTING SOUNDS, THEREBY PRODUCING A PLURALITY OF ACOUSTIC PARAMETERS

FIG. 10

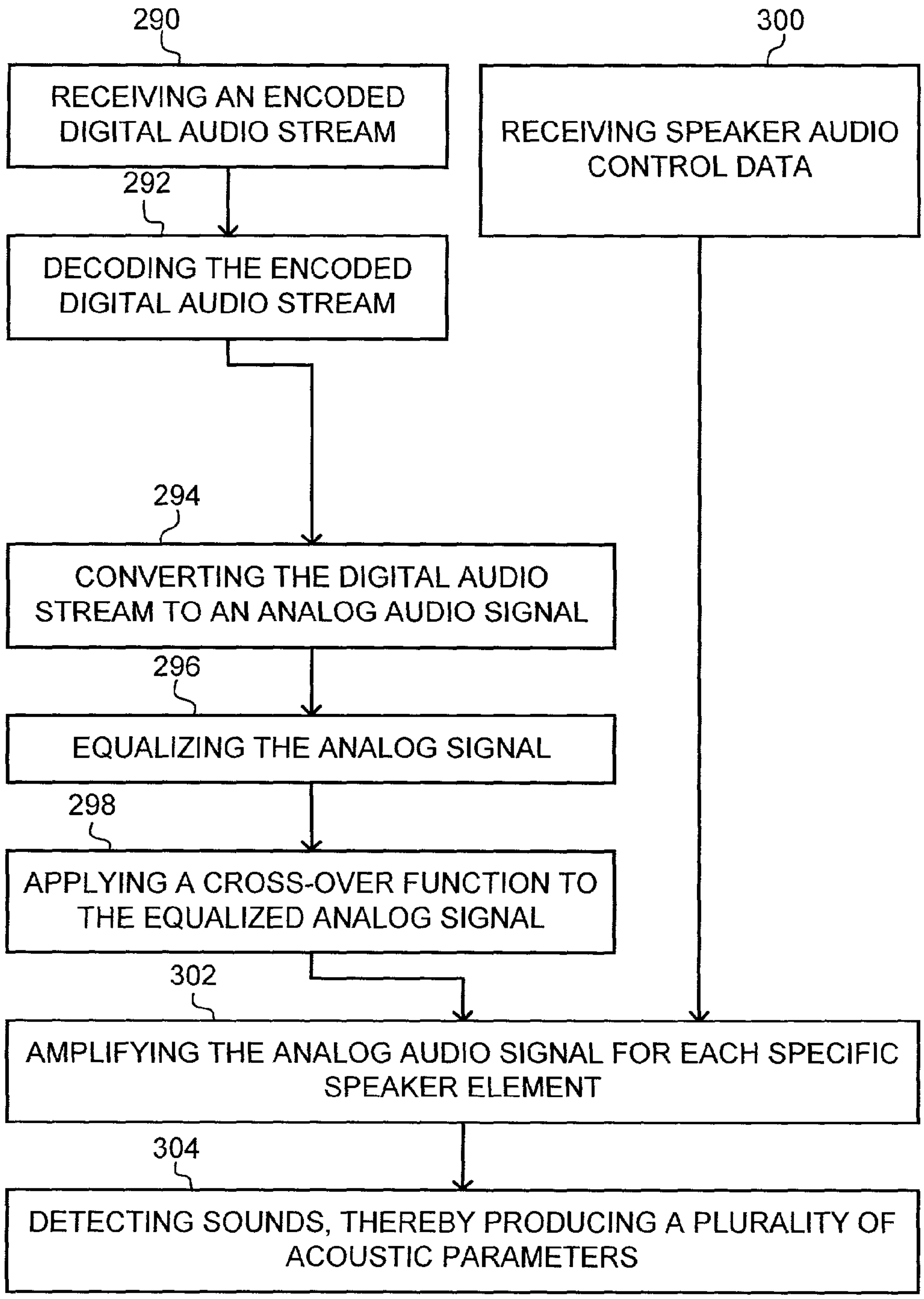

290
RECEIVING AN ENCODED DIGITAL AUDIO STREAM
292
DECODING THE ENCODED DIGITAL AUDIO STREAM
300
RECEIVING SPEAKER AUDIO CONTROL DATA
294
CONVERTING THE DIGITAL AUDIO STREAM TO AN ANALOG AUDIO SIGNAL
296
EQUALIZING THE ANALOG SIGNAL
298
APPLYING A CROSS-OVER FUNCTION TO THE EQUALIZED ANALOG SIGNAL
302
AMPLIFYING THE ANALOG AUDIO SIGNAL FOR EACH SPECIFIC SPEAKER ELEMENT
304
DETECTING SOUNDS, THEREBY PRODUCING A PLURALITY OF ACOUSTIC PARAMETERS

FIG. 11

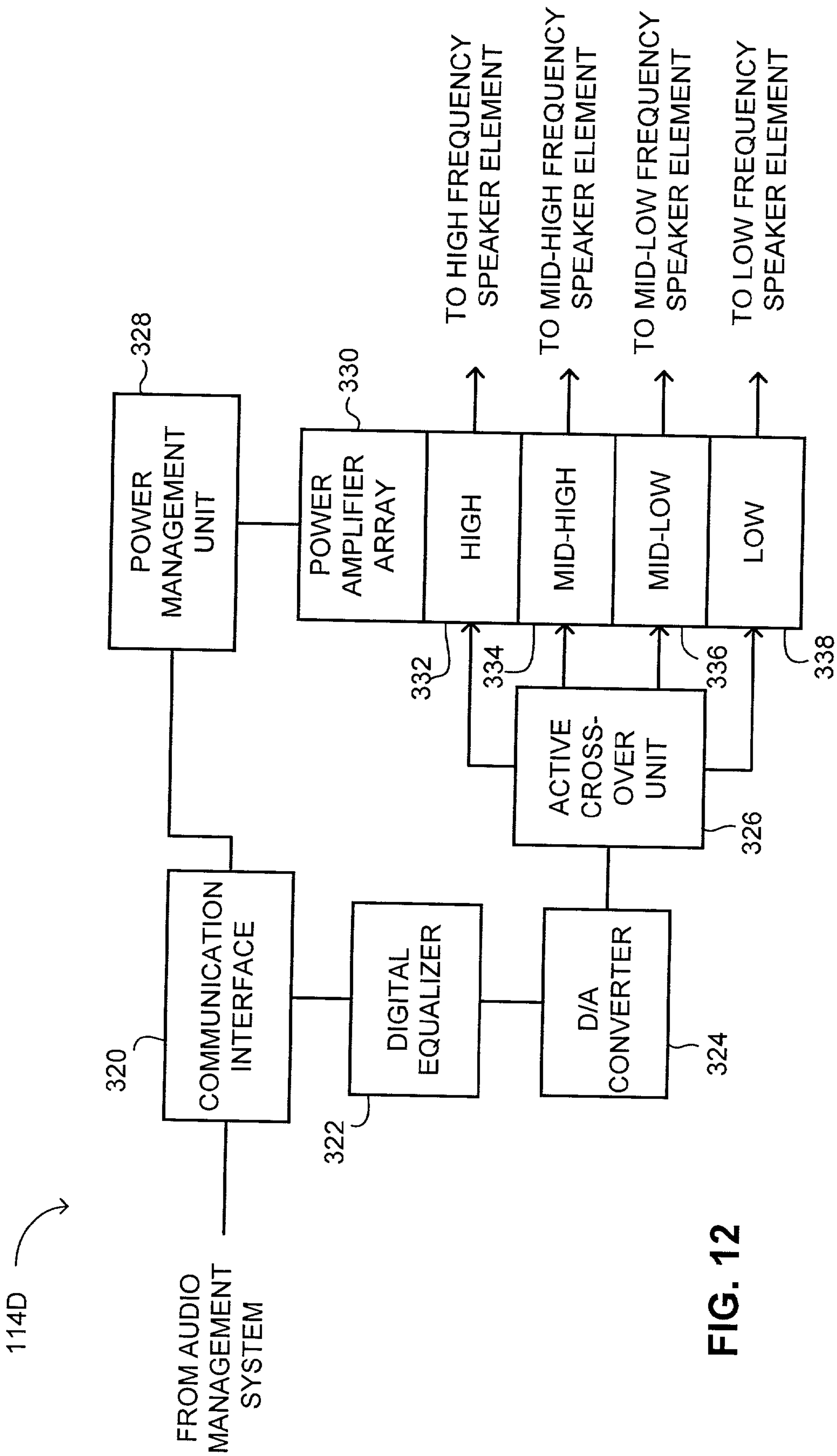

114D
FROM AUDIO MANAGEMENT SYSTEM
320
COMMUNICATION INTERFACE
322
DIGITAL EQUALIZER
324
D/A CONVERTER
326
ACTIVE CROSS-OVER UNIT
328
POWER MANAGEMENT UNIT
330
POWER AMPLIFIER ARRAY
HIGH
MID-HIGH
MID-LOW
LOW
332
334
336
338
TO HIGH FREQUENCY SPEAKER ELEMENT
TO MID-HIGH FREQUENCY SPEAKER ELEMENT
TO MID-LOW FREQUENCY SPEAKER ELEMENT
TO LOW FREQUENCY SPEAKER ELEMENT

FIG. 12

# AUDIO SYSTEM

## FIELD OF THE INVENTION

[0001] The present invention relates to communication systems in general, and to methods and systems for establishing an intelligent sound system, in particular.

## BACKGROUND OF THE INVENTION

[0002] Systems and methods for broadcasting an audio program from a set of remote speakers are known in the art. Such audio systems generally include an audio data source, a system for processing and selecting an audio signal, an amplifier for amplifying the selected signal and a plurality of speakers for reproducing the amplified signal at an audio device. Traditionally, the audio program system configures with multiple speakers cabled together. This configuration may include multiple networks or multiple zones with a common signal per zone.

[0003] U.S. Pat. No. 5,406,634 issued to Anderson et al., and entitled "Intelligent Speaker Element Unit for Speaker Element System Network", is directed to an audio broadcast system. This system provides some of the advantages of the intelligence at the remote speaker element such as, selection of remote speakers, selection of channels and speakers and mixing of selected channels in the selected remote speaker element. The speaker element may be adjusted to its environment and all of these tasks may be accomplished remotely. The system includes a central processing unit (CPU), an analog audio source, a digital audio source, analog to digital converters (A/D), a multiplexor (MUX), a transmitter, a receiver, and a set of intelligent speakers connected to the transmitter and the receiver via separate buses. The intelligent speaker element includes a communications receiver, a communications transmitter, a digital signal processor (DSP), a digital to analog converter (D/A), an audio amplifier, a power supply and an address. The DSP processes the audio data in accordance with the control data. The control data includes an address for selecting the speaker element. When the DSP detects the address of the speaker element in the control command, the DSP is enabled to process audio data digitally based on parameters in the control data. The processed digital audio data is then converted to an analog signal, which is amplified and sent to the speaker element.

[0004] A system which provides a switching network for interconnective and active demonstration of consumer electronic products, is described in U.S. Pat. No. 5,646,602 issued to Gertz et al., which is entitled "Switching Network for Selective Product Demonstration". This reference is directed to a switching network for selective product demonstration. The switching network provides features such as advanced noise suppression system, combining source, source mute and speaker element de-selection and a multiple speaker element selection ability to allow simultaneous demonstration of two or more speaker element pairs. The switching network further provides an automatic volume calibration, automatic sound pressure limiting and a built-in mechanism for indicating unauthorized disconnection of a product from the switching network. In this system, different product combinations may be compared instantaneously at identical volume levels and annoying sudden bursts of loud music are eliminated. The switching network includes a plurality of nodes: source nodes, amplifier nodes and speaker nodes. Each product on display is connected to a node. The total number of sources, amplifiers and speakers that may be interconnected through the source, amplifier and speaker nodes is essentially unlimited, but for practical purposes it can be up to 99. The switching network also includes a control means for supervising all network functions. The control means includes a micro-controller at each node to take advantage of the distributed intelligence architecture.

[0005] Transmission and reception systems for performing transmission and reception of various types of signals through a transmission line are described in U.S. Pat. No. 5,287,547 issued to Hidaka, and entitled "Transmission and Reception System". The basic configuration of the transmission and reception system (**FIG. 1**) includes transmitting means **1**, receiving means **2**, transmission request signal output means **3**, decoding means **4** and information signal output control means **5**.

[0006] This system provides a transmission and reception capability in which an information signal from transmitting means **1** provided at a first location, is transmitted through a transmission line to be received by receiving means **2** provided at the second location. According to this system, an information signal sent through the transmission line is received by receiving means **2**. The transmission request signal output means **3**, provided at the second location includes the capability to encode a transmission request signal and to send the encoded transmission request signal to the transmission line. Decoding means **4** is also provided at the first location for receiving and decoding the transmission request signal. Information signal output control means **5** causes transmitting means **1**, to output the information signal to the transmission line. At this time, information signal output control means **5** controls transmitting means **1** to cause transmitting means **1** to output an information signal to be received by receiving means **2**.

[0007] U.S. Pat. No. 5,182,552 issued to Paynting, and entitled "Multiple Zone Audio System" is directed to a program signal processing system, which is monitored and controlled from one of many remote locations. The program sources are located at a central location. However, the program signal monitored at a remote location may originate from the program source at which the signal is being monitored (the central location or other locations), or from a different program source. The monitored signals are audio signals consist of a compact disc player, a radio tuner, a television, an audio signal tuner and an audio type player. The circuitry at the central location includes a central controller, and at least two independent selectors, which can be independently selected from the input sources for routing to an output channel. The circuitry may include at least two volume controllers responsive to the central controller. In order to assist control of the system, commands which are issued to the system from a remote interface, are verified by a verification signal added to the output signal sent to the output channels. The program signal processing system (**FIG. 1**) includes a plurality of program sources **100**, a plurality of program signal selectors **110**, a plurality of verify and processing blocks **120**, a plurality of remote interfaces **140** and a central controller **130**. Central controller **130** monitors and controls the operation of the program signal processing system, and supports remote interfaces

140 to control the system. In the embodiment shown in **FIG. 2**, the amplitude of the signal delivered by zone 1 and zone 2 remote speakers **260** is controlled by volume control circuits **231** and **232**. Program signals from system **200** drive the speakers via amplifiers and speaker wires. Multiple remote controllers may be used with system **200**. An arbitration scheme provides the controllers with exclusive control of volume and source selection for one zone, as well as the source functions for the selected source.

[**0008**] A method and apparatus for configuring plural multi-media audio cards as a local area network, are described in U.S. Pat. No. 5,519,641 issued to Beers et al., and entitled "Method and Apparatus for Configuring Plural Multimedia Audio Cards as a Local Area Network". According to this method, a plurality of computers can be configured as a LAN (local area network) through the use of an audio card, cables, and a communication protocol. This system makes use of the line-in/line-out connectors for each right and left stereo channel of the audio card, to provide a communication network. The audio and data information can be transmitted simultaneously over the local area network. The distributed computers are connected in a master/slave configuration. All line-in ports of the slave systems are connected together, and the master system lineout port is connected to the slave systems line-in ports, for each channel. All lineout ports of the slave systems are connected together and the master system line-in port is connected to the slave systems lineout ports, for each channel. Only one slave lineout can actively transmit data at any time. A communication protocol is provided wherein the master system provides a clock signal on the control channel. During communication, if either the master or slave system recognize their address on the packet received on the line-in channel, then the information is decoded and processed by the digital signal processor on the audio card. The audio data can be output to the system as a play file or the like, while the data information may be sent directly to the host CPU in the system, either to the master or the slave, for display on the computer screen. This system also provides a multi-tasking program for controlling an onboard processor on the audio card. This will allow the simultaneous transmission and/or receipt of audio and data information and audio play or record by the systems interconnected in the local area network audio configuration.

[**0009**] U.S. Pat. No. 5,818,948 issued to Gulick, and entitled "Architecture for a Universal Serial Bus-Based PC Speaker Controller", is directed to a system for transmission of audio signals to speakers, through a Universal Serial Bus (USB). A powered loudspeaker, compatible with a USB specification is described in this system. Powered speaker **108** (**FIG. 5**) includes a power supply **110**, an amplifier **114**, a USB controller **112** and a speaker **116**. The USB controller is connected to a USB host interface, for example, of a PC. It is noted that a single USB port can host up to 127 devices and hence can host a plurality of such powered speakers, each receiving information through the USB bus, decoding it, and providing it to the internal amplifier **114** which is powered by the power supply unit **110**. This system provides tone control, volume control and balance between left and right outputs (in a stereo version). Furthermore, power management function is provided. The system can place itself into a sleep mode via software from the host.

## SUMMARY OF THE PRESENT INVENTION

[**0010**] It is an object of the present invention to provide a novel method and system for operating a plurality of digital speakers connected to a digital audio network. In accordance with the present invention, there is thus provided an audio system for operating a plurality of speakers. The audio system includes an audio management system, a user interface and an audio source cluster. The audio management system is connected to the speakers via a network. The audio management system is connected to the user interface and to the audio source cluster. The audio management system operates the speakers by transmitting audio streams and speaker audio control data to the network. The audio management system determines the speaker audio control data, according to a plurality of parameters.

[**0011**] In accordance with another aspect of the present invention, there is thus provided a method for operating an audio system. The audio system includes an audio management system, a user interface, an audio source cluster and a plurality of speakers. The speakers are connected to the audio management system via a network, and the audio management system is connected to the user interface and to the audio source cluster. The method includes the steps of determining network audio control data, conforming audio streams, encoding the conformed audio streams and determining speaker audio control data. The network audio control data is determined according to automatic predetermined parameters and user defined parameters. The audio streams are conformed according to the network audio control parameters. The speaker audio control data is determined according to the automatic predetermined parameters and the user defined parameters.

[**0012**] In accordance with a further aspect of the present invention, there is thus provided a method for operating an audio system. The audio system includes an audio management system, a user interface, an audio source cluster and a plurality of speakers, wherein each of the speakers includes a speaker element. The speakers are connected to the audio management system via a network, and the audio management system is connected to the user interface and to the audio source cluster. The method includes the steps of amplifying an analog audio signal, providing the amplified analog audio signal to each of the speaker elements, detecting sounds from each of the speaker elements and producing an acoustic parameter respective of the sounds.

[**0013**] In accordance with another aspect of the present invention, there is thus provided a method for operating an audio system. The audio system includes an audio management system, a user interface, an audio source cluster and a plurality of speakers, wherein each of the speakers includes a speaker element. The speakers are connected to the audio management system via a network, and the audio management system is connected to the user interface and to the audio source cluster. The method includes the steps of performing a cross-over function on an analog audio signal, amplifying the analog audio signal for each of the speaker elements, detecting sounds from each of the speaker elements and producing an acoustic parameter respective of the sounds.

[**0014**] In accordance with a further aspect of the present invention, there is thus provided a method for operating an audio system. The audio system includes an audio management system, a user interface, an audio source cluster and a plurality of speakers, wherein each of the speakers includes a speaker element. The speakers are connected to the audio management system via a network, and the audio management system is connected to the user interface and to the audio source cluster. The method includes the steps of performing a cross-over function on an equalized analog audio signal, amplifying the equalized analog audio signal for each of the speaker elements, detecting sounds from each of the speaker elements, and producing an acoustic parameter respective of the sounds.

## BRIEF DESCRIPTION OF THE DRAWINGS

[**0015**] The present invention will be understood and appreciated more fully from the following detailed description taken in conjunction with the drawings in which:

[**0016**] **FIG. 1** is a schematic illustration of an audio system, constructed and operative in accordance with a preferred embodiment of the present invention;

[**0017**] **FIG. 2** is a schematic illustration in detail of the audio management system of **FIG. 1**, constructed and operative in accordance with another preferred embodiment of the present invention;

[**0018**] **FIG. 3** is a schematic illustration in detail of the audio source cluster of **FIG. 1**, constructed and operative in accordance with a further embodiment of the present invention;

[**0019**] **FIG. 4** is a schematic illustration in detail of one of the computerized speakers of the system of **FIG. 1**, constructed and operative in accordance with another preferred embodiment of the present invention;

[**0020**] **FIG. 5** is a schematic illustration in detail of another computerized speaker of the system of **FIG. 1**, constructed and operative in accordance with another preferred embodiment of the present invention;

[**0021**] **FIG. 6** is a schematic illustration of a ring type configured audio system, constructed and operative in accordance with a further preferred embodiment of the present invention;

[**0022**] **FIG. 7** is a schematic illustration in detail of a further computerized speaker of the system of **FIG. 1**, constructed and operative in accordance with another preferred embodiment of the present invention;

[**0023**] **FIG. 8** is an illustration of a method for managing audio, operative in accordance with another preferred embodiment of the present invention. The method of **FIG. 8** can be used for operating audio management system **100** of **FIG. 1**;

[**0024**] **FIG. 9** is an illustration of a method for operating a computerized speaker **114**A of **FIG. 4**, operative in accordance with another preferred embodiment of the present invention;

[**0025**] **FIG. 10**, which is an illustration of a method for operating a computerized speaker **114**D of **FIG. 12**, operative in accordance with another preferred embodiment of the present invention.

[**0026**] **FIG. 11** is an illustration of a method for operating a computerized speaker **114**B of **FIG. 5**, operative in accordance with another preferred embodiment of the present invention;

[**0027**] **FIG. 12** is a schematic illustration in detail of another computerized speaker of the system of **FIG. 1**, constructed and operative in accordance with another preferred embodiment of the present invention.

## DETAILED DESCRIPTION OF PREFERRED EMBODIMENTS

[**0028**] The present invention overcomes the disadvantages of the prior art by providing a digital networked audio system and a method for operating the same, with digital speakers, which include internal and ambient dynamic audio processing, amplification and psycho-acoustic manipulation (i.e., creating a sensation of acoustic depth through surround sound techniques) of the received audio stream. The invention preferably makes use of a digital communication link, which is based on digital isochronous communication concept. A digital isochronous communication channel guarantees predetermined bandwidth (for example, at least at 44.1 k-bit/sec) for each network address (i.e., digital speaker device) in the network. The use of digital isochronous communication eliminates any noticeable delay in network traffic. It will be appreciated by those skilled in the art that such delay exists when using conventional networking protocols such as Voice Over IP (VoIP), and the like.

[**0029**] The digital speakers are a set of network devices, identified by a unique address, and can attach/detach themselves without requiring reboot of the network, or re-initialization. This is due to the fact that isochronous protocols funnel the information regardless of the existence of the network device as a network link. Hence, such a communication channel does not exhibit packet loss. The digital samples can be packetized (as in Internet Protocols) or can be transmitted without packet overhead (e.g., Header/Footer/Parity). It is noted that the present invention can make further use of conventional networking protocols such as Ethernet, ATM, and the like, for conveying control data between the network devices and the audio management system.

[**0030**] The speed of sound has a considerable effect on large-scale sound systems employing a large number of speakers, which are spread over a large area. A listener located at a certain location within that area, receives the sound from each speaker at a delay, which is proportional to his distance from that speaker. For example, in an open field concert, which employs front-end speakers and rear-end speakers, all of which are directed at the audience, the rear-end listeners would hear the front-end speakers, at a delay from the rear-end speakers. The delay D equals the distance between the front-end speakers and the rear-end speakers, divided by the speed of sound.

[**0031**] According to one aspect of the invention, the sound produced by the rear-end speakers is delayed by D, with reference to the sound produced by the front-end speakers. Hence, the rear-end listeners receive both sounds at the same phase. The digital isochronous communication protocols enable a unique capability to program the delay element associated with each of the devices as a fixed delay. This can be used to accurately calculate all delays associated with the digital communication link. Hence, the use of digital isochronous communication protocols nullifies dynamic delay in the audio system, and eliminates the need for echo cancellation procedures, and protocols associated with various communication links.

[0032] Digital isochronous protocols, for use with the present invention to funnel digital audio data to the remote speakers, can be selected form the list consisting of low level Universal Serial Bus (USB), high speed IEEE1394, Short Distance Wireless Blue-tooth format, and the like. The open network topology, used for connecting the digital speakers, can be ring, star, or a combination of ring and star.

[0033] In addition, these protocols and their specifications provide tools and services, which facilitate interoperability and development of distributed applications on communication networks, such as home networks. For example, the API used in IEEE1394 is based on the Home Audio/Video Interoperability (HAVi) Specification 1.0.

[0034] Reference is now made to **FIG. 1**, which is a schematic illustration of an audio system, generally referenced **100**, constructed and operative in accordance with a preferred embodiment of the present invention. Audio system **100** includes an audio management system **102**, a user interface **104**, an audio source cluster **106**, a router **108**, wireless transceivers **110** and **112**, and a plurality of computerized speakers **114A**, **114B**, **114C**, **114D**, **114E**, **114F** and network **116**.

[0035] Audio management system **102** is connected to audio source cluster **106**, to user interface **104** and to network **116**. Network **116** is further connected to computerized speakers **114A**, **114B**, **114C**, to router **108** and to wireless transceiver **110**. Router **108** is further connected to computerized speakers **114D** and **114E**. Wireless transceiver **112** is connected to computerized speaker **114F**. Wireless transceiver **110** is wirelessly coupled to wireless transceiver **112**.

[0036] Network **116** provides at least one isochronous channel, which maintains fixed bandwidth. The number of network speakers, connected to the network, is virtually unlimited, according to the available network addresses. According to a preferred embodiment of the present invention, audio data transmitted over network **116**, utilizes digital isochronous communication links (channels), where the following is a short introduction therefor.

[0037] Audio management system **102** receives audio streams in digital and analog form from audio source cluster **106**. User interface **104** provides user commands to audio management system **102** for processing the received audio streams.

[0038] Audio management system **102** determines network audio control data according to automatic predetermined and user defined parameters. Audio management system **102** processes the audio streams according to the user commands, network audio control data and produces speaker audio control data, respective of various audio parameters, such as volume, balance, equalization, harmonic envelope, echo and audio effects, and the like. Audio management system **102** transmits the audio streams and the speaker audio control data to remote computerized speakers via network **116**. Router **108** provides scalable coupling of additional computerized speakers (such as the ones referenced **114D** and **114E**) to network **116**. Computerized speaker **114F** is wirelessly connected to network **116** via transceivers **110** and **112**.

[0039] Reference is now made to **FIG. 2**, which is a schematic illustration in detail of audio management system **102** of **FIG. 1**, constructed and operative in accordance with another preferred embodiment of the present invention. Audio management system **102** includes a central processing unit (CPU) **120** and a digital signal processor (DSP) **122**. DSP **122** includes a real-time adaptive analyzer **126**, an audio encoder **128** and a master band equalizer **130**.

[0040] CPU **120** is connected to user interface **104** (**FIG. 1**), DSP **122**, real-time adaptive analyzer **126**, audio encoder **128** and to master band equalizer **130**. Real-time adaptive analyzer **126** is further connected to a microphone **124**, audio encoder **128**, master band equalizer **130** and to CPU **120**. Microphone **124** can be placed at various locations within the system. Audio encoder **128** is further connected to master band equalizer **130** and to communication interface **150**.

[0041] Audio management system **102** receives a plurality of digital and analog audio streams from audio source cluster **106** (**FIG. 1**), and processes and transforms the digital audio streams according to control parameters (predetermined parameters of the speaker audio control data and of the network audio control data). Control parameters, such as volume, balance, equalization, harmonic envelope, echo and audio effects, and the like, are either defined by the user or determined automatically, using feedback mechanisms.

[0042] CPU **120** controls and manages the general operation of audio management system **102**. User interface **104** receives control parameter selection from a user operating the system. User interface **104** provides the received control parameters to CPU **120**, which in turn updates the settings of the modules connected thereto.

[0043] DSP **122** decodes the multiplexed digital audio streams received from audio source cluster **106** of **FIG. 1**. The received digital audio streams may be provided in any conventional or proprietary format, such as Dolby Logic, Dolby Pro-Logic, Dolby Digital 5.1 channels (AC3 format), Digital Theatrical Sound (DTS Format), Ultimate Computerized Pro-Audio Sound System (UCPASS) multi-channel encoded audio source, and the like. The system can be adapted to support any future sound-encoding format. The decoding operation can include any operation required to produce a digital representation of the audio streams, which can then be audio analyzed. These operations can include de-multiplexing, separating the various audio streams, converting from one digital format to another, and the like.

[0044] DSP **122** incorporates an audio decoding layer, which expands the compressed audio streams. The audio decoding layer employs the pulse code modulation (PCM) scheme. DSP **122** funnels each of the PCM decoded streams to a selected speaker. This operation can be performed in the process of encoding the streams. DSP **122** sends a control sequence to the selected speaker, which configures that speaker to operate according to selected format. The DSP **122** can further produce an indication to the user, via the user interface. According to one embodiment of the invention, this audio decoding layer supports MPEG-3, also known in the industry as MP3 format.

[0045] Audio source cluster **106** (**FIG. 1**) provides the digital and analog audio streams to real time adaptive analyzer **126**. CPU **120** provides control parameters to real time adaptive analyzer **126**. Real time adaptive analyzer **126** further receives audio feedback from microphone **124**.

Thereby, the extracted sound parameters from PCM highway are funneled to the channels of audio management system **102**. Real time adaptive analyzer **126** analyzes the digital audio streams according to the control parameters and the audio feedback, determines the difference between them and updates the settings of the master band equalizer **130**, in real time.

[**0046**] Master band equalizer **130** transforms the digital audio streams according to the settings provided by real time adaptive analyzer **126** and initial equalization parameters, which are set by the listener. For example, master band equalizer **130** can receive a plurality of equalization parameter sets, each set for a selected band pass filter, of the plurality of band pass filters which are defined for the system. For example, the system can operate according to a total of sixty-two band pass filters, at a resolution of one sixth of an octave each. Each of the parameter sets can include parameters such as quality sharpness of the band-pass filter, gain of the filter (e.g., +/−18 db), and the like.

[**0047**] Master band equalizer **130** can operate in two alternative modes, an automatic mode and a manual mode. In the automatic mode, master band equalizer **130** receives a parameter set for each band-pass filter, directly from the real-time adaptive analyzer. In the manual mode, user interface **104** updates various parameters directly, as desired by the user operating the system. In the manual mode, the parameter set of each of the band-pass filters can be selected by placing the mouse cursor on the slide control knobs depicted in a selected window of the user interface. Dragging each control knob up or down with the mouse, changes the values of these parameters.

[**0048**] It is noted that a master-bass and a master-treble control provide a general-purpose compatibility of a plurality of amplifiers certified on the audio market. The master-bass control is a proprietary air-band, controlled by a plurality of band-pass filter switches. The master-bass control and master-treble control, provide audio streams in logarithmic scales of 0-100 Hz and 2.5-22 kHz, respectively, to the respective amplifier. The master-bass control and the master-treble control are selected band pass filters, which provide audio streams to respective amplifiers with logarithmic scale at 0-100 Hz and logarithmic scale at 10 kHz-22 kHz accordingly. User interface **104** provides data to the user and receives user selections from the user, via a variety of input and output devices such as video, audio (e.g., sound and voice recognition, and the like), tactile (e.g., keyboard, mouse, general pointing devices), and the like. In the automatic mode, user interface **104** provides initial settings of the control parameters. Real time adaptive analyzer **126** compares the feedback stream with the transmitted audio streams so as to determine the difference there between two and provide the required update in the control parameters. The predetermined policies set limits for the actual update, which is finally introduced to the audio control data parameters. It is noted that the update can either be of network control parameters or of the speaker element control parameters, or both.

[**0049**] Audio encoder **128** receives the equalized digital audio streams, encodes them and transmits the encoded streams to communication interface **150**. Alternatively, audio management system **102** can disable the audio encoding operation of audio encoder **128**, and transmit the digital audio streams to the computerized speakers, without encoding. CPU **120** provides speaker element control parameters to audio encoder **128**, which encodes them according to a predetermined format, and provides the encoded speaker element control parameters to communication interface **150**, for further transmitting to the digital speaker elements. The corrected speaker element control parameters (delay, volume, equalization, cross, and the like) can be dynamically adapted for maintaining an optimum operating condition. An optimum operating condition can be an optimum frequency response, optimum phase relations throughout the audible spectrum, and the like.

[**0050**] In the present invention, real time adaptive analyzer **126** maintains a status and a configuration of "last" operation. The user can always recall a previous setting of the audio system. Furthermore, the audio system can automatically recall a plurality of user defined sound parameters (or parameter sets—profiles), such as Concert, Classical-Hall, Jazz, Stadium, Pop, Rock, and the like from a preset menu and a plurality of user program registers. These predefined profiles can be stored in a storage unit (not shown) or received from a user operated storage device, such as a diskette, a memory smart card, and the like.

[**0051**] Reference is further made to **FIG. 3**, which is a schematic illustration in detail of the audio source cluster **106** of **FIG. 1**, constructed and operative in accordance with a further embodiment of the present invention. Audio source cluster **106** includes a plurality of analog audio sources **140**A, **140**B and **140**C, a plurality of digital audio sources **142**A, **142**B and **142**C, an analog to digital converter (A/D) **144**, a sampling rate converter **146** and a multiplexor **148**.

[**0052**] A/D converter **144** is connected to analog audio sources **140**A, **140**B and **140**C and to multiplexor **148**. Sampling rate converter **146** is connected to digital audio sources **142**A, **142**B and **142**C and to multiplexor **148**. Multiplexor **148** is further connected to audio management system **102** (**FIG. 1**). An audio signal can either be an analog audio signal (when received from analog audio sources **140**A, **140**B and **140**C) or a digital audio signal (when received from digital audio sources **142**A, **142**B and **142**C).

[**0053**] A/D converter **144** receives the analog audio signal from each of analog audio sources **140**A, **140**B and **140**C, converts these analog audio signals to the digital audio streams and transmits that digital audio streams to multiplexor **148**. Sampling rate converter **146** receives digital audio streams from digital audio sources **142**A, **142**B and **142**C, converts a sampling rate of the digital audio streams to a selected digital-sampling rate. In the present example, the selected sampling rate is 96.0 kb/sec. Sampling rate converter **146** provides the digital audio streams at the selected sampling rate, to multiplexor **148**. Multiplexor **148** multiplexes the digital audio streams, received from sampling rate converter **146** and A/D converter **144** and provides the multiplexed audio streams to audio management system **102** (**FIG. 1**).

[**0054**] Reference is now made to **FIG. 8**, which is an illustration of a method for managing audio, operative in accordance with another preferred embodiment of the present invention. The method of **FIG. 8** can be used for operating audio management system **100** of **FIG. 1**.

[**0055**] In step **220**, an analog audio signal is received from an analog audio source. The analog source can be an analog output of any known media device, such as a CD player, a sound card, a television set, a radio device, and the like. It is noted that such a device can provide a plurality of related analog signals, such as in stereo, surround sound, multiple language translation, and the like. With reference to **FIG. 3**, A/D converter **144** receives analog signals, from either of the analog audio sources **140**A, **140**B and **140**C.

[**0056**] In step **222**, the received analog audio signal is converted to digital audio streams. With reference to **FIG. 3**, A/D converter **144** converts the received audio signals to digital audio streams.

[**0057**] In step **224**, digital audio streams are received from a digital audio source. The digital source can be any known media device, such as a CD player, a DVD player, a digital Set-Top Box, a Digital Console, and the like, having a digital audio stream output. With reference to **FIG. 3**, sampling rate converter **146** receives the digital streams, from either of the digital audio sources **142**A, **142**B and **142**C.

[**0058**] In step **226**, the received digital audio streams are converted according to a selected sampling rate. With reference to **FIG. 3**, sampling rate converter **146** converts received digital audio streams, to a selected digital sampling rate (in the present example, the selected sampling rate is 96.0 kb/sec).

[**0059**] In step **228**, the digital audio streams are multiplexed. With reference to **FIG. 3**, multiplexor **148** multiplexes the digital audio streams received from A/D converter **144** and the digital audio streams received from sampling rate converter **146**.

[**0060**] In step **230**, the multiplexed digital audio streams are decoded. The decoding operation can include any operation required to produce a digital representation of the audio streams, which can be audio analyzed. These operations can include de-multiplexing, separating the various audio streams, converting from one digital format to another, and the like. With reference to **FIG. 2**, DSP **122** decodes the multiplexed digital audio streams received from audio source cluster **106** of **FIG. 1**. It is noted that DSP **122** can include a dedicated decoding module for these purposes (not shown).

[**0061**] In step **232**, network audio control data is determined according to automatic predetermined parameters (by using a feedback mechanism) and user defined parameters. The network audio control data can include a plurality of various audio parameters, such as volume, balance, equalization, harmonic envelope, echo and audio effects, and the like. These parameters can continuously be updated by a feedback mechanism. Alternatively, these parameters can be set by a user. With reference to **FIG. 2**, the user provides CPU **120** with his selection of the settings for the audio parameters via user interface **104**. The user selections can further include audio feedback policies, surround settings, and the like. CPU **120** provides these settings to real-time adaptive analyzer **126**, which further receives feedback signal from microphone **124**. Real-time adaptive analyzer **126** analyzes the received audio streams with respect to the parameters and the feedback signal and updates the settings of the master band equalizer **130**.

[**0062**] In step **234**, the decoded digital audio streams are conformed according to the determined network audio control data. This procedure is preferably performed in real time. With reference to **FIG. 2**, master band equalizer **130** conforms the digital audio streams according to audio data provided by real-time adaptive analyzer **126**. For example, if a selected frequency is set to be at a predetermined volume level and the feedback signal indicates that it was produced beyond that level (with respect to other frequencies), then, this frequency can be somewhat diminished from now on so as to achieve the original settings.

[**0063**] In step **236**, the processed digital audio streams are encoded. It is noted that this step of encoding can be eliminated. With reference to **FIG. 2**, audio encoder **128** encodes the digital audio streams received from master band equalizer **130**. Alternatively, audio management system **102** can disable audio encoding operation of audio encoder **128** and so to transmit the digital audio streams to the computerized speakers, without encoding.

[**0064**] In step **238**, speaker audio control data is determined according to automatic predetermined parameters (by using a feedback mechanism) and user defined parameters. The speaker audio control data can include a plurality of audio parameters, such as volume, balance, equalization, harmonic envelope, echo and audio effects, phase, time-delay, and the like. These parameters can continuously be updated by a feedback mechanism. Alternatively, these parameters can be set by a user. With reference to **FIG. 2**, the user provides CPU **120** with his selection of the settings for the audio parameters via user interface **104**. The user selections can further include audio feedback policies, surround settings, and the like. CPU **120** provides these settings to real-time adaptive analyzer **126**, which further receives feedback signal from microphone **124**. Real-time adaptive analyzer **126** analyzes the audio streams received with respect to the parameters and the feedback signal and updates the settings of the master band equalizer **130**. Thus, either audio management system **102** or the user can perform high speed switching of each of the speaker elements. For example, the user can convert a two-way speaker to two separate speakers.

[**0065**] In step **240**, the digital audio streams (encoded or otherwise) and the speaker audio control data are transmitted to a selected network speaker. With reference to **FIG. 2**, communication interface **150** transmits each of the digital audio streams to a selected network speaker **114**A, **114**B, **114**C, **114**D, **114**E and **114**F (**FIG. 1**). In addition, communication interface **150** further transmits respective speaker audio control data, received from the audio management system **102**, to each of the network speakers **114**A, **114**B, **114**C, **114**D, **114**E and **114**F. It is noted that each of the digital audio streams can be transmitted to each of the speakers and furthermore, to selected sections of different speakers (e.g., the high portion of the stream being transmitted to the high section of one speaker and the low portion of that same stream being transmitted to the low section of another speaker).

[**0066**] Reference is now made to **FIG. 4**, which is a schematic illustration of computerized speaker **114**A of **FIG. 1**, constructed and operative in accordance with a further preferred embodiment of the present invention. Computerized speaker **114**A includes a communication interface **150**, a local DSP **152**, an analog to digital (A/D) converter **151**, a microphone **153**, digital to analog converters (D/A) **154**A and **154**B, a power amplifier array **164**, a power management unit **174**, a high frequency speaker element **166**, a mid-high frequency speaker element **168**, a mid-low frequency speaker element **170** and a low frequency speaker element **172**. Power amplifier array **164** includes a high frequency bandwidth amplifier **156**, a mid-high frequency bandwidth amplifier **158**, a mid-low frequency bandwidth amplifier **160** and a low frequency bandwidth amplifier **162**.

[**0067**] Local DSP **152** is connected to communication interface **150**, A/D converter **151**, D/A converters **154**A and **154**B, and to power management unit **174**. Microphone **153** is connected to A/D converter **151**. Power amplifier array **164** is connected to D/A converters **154**A and **154**B and to power management unit **174**.

[**0068**] High frequency bandwidth amplifier **156** is connected to D/A converter **154**A and to high frequency speaker element **166**. Mid-high frequency bandwidth amplifier **158** is connected to D/A converter **154**A and to mid-high frequency speaker element **168**. Mid-Low frequency bandwidth amplifier **160** is connected to D/A converter **154**B and to mid-low frequency speaker element **170**. low frequency bandwidth amplifier **162** is connected to D/A converter **154**B and to low frequency speaker element **172**.

[**0069**] Communication interface **150** receives digital audio streams and speaker audio control data from audio management system **102 (FIG. 2)**, and provides the streams and data to local DSP **152**. Local DSP **152** processes the received digital audio streams according to speaker audio control data, thereby producing four digital audio streams (such as streams from different channels of a stereo audio system, and the like). It is noted that the digital audio streams can include predetermined and already separated digital audio streams, one for each band (i.e., high, mid-high range, mid-low and low), or a single full band stream, on which the local DSP **152** applies a cross-over function.

[**0070**] Local DSP **152** provides each of the processed audio streams to the respective D/A input port. Local DSP **152** processes the speaker element control data, thereby producing amplification commands, designated for each of the amplifiers of power amplifier array **164**, and applies amplification commands to power management unit **174**. An amplification command can be an on-off command, volume control, percentage of the maximum amplifier gain during operation, and the like.

[**0071**] D/A converter **154**A receives the high frequency digital audio stream and the mid-high frequency digital audio stream from local DSP **152**, converts each of these streams to a respective analog audio signal. D/A converter **154**A, furthermore directs the high frequency analog signal to high frequency bandwidth amplifier **156** and the mid-high frequency analog signal to mid-high frequency bandwidth amplifier **158**. D/A converter **154**B receives the mid-low frequency digital audio stream and the low frequency digital audio stream from local DSP **152**. D/A converter **154**B converts these streams to respective analog audio signals and directs the mid-low frequency analog signal to mid-low frequency bandwidth amplifier **160** and the low frequency analog signal to low frequency bandwidth amplifier **162**.

[**0072**] Power management unit **174** provides high electric power, required to operate power amplifier array **164**. Power amplifier array **164** amplifies the analog audio signals separately and provides each amplified signal to the respective speaker element. High frequency bandwidth amplifier **156** receives the high frequency audio signal from D/A converter **154**A and amplification commands from power management unit **174**. High frequency bandwidth amplifier **156** amplifies the high frequency audio signal, according to the amplification commands and provides the amplified signal to high frequency speaker element **166**.

[**0073**] Mid-high frequency bandwidth amplifier **158** receives the mid-high frequency audio signal from D/A converter **154**A and amplification commands from power management unit **174**. Mid-high frequency bandwidth amplifier **158** amplifies the mid-high frequency audio signal, according to the amplification commands and provides the amplified signal to mid-high frequency speaker element **168**.

[**0074**] Mid-Low frequency bandwidth amplifier **160** receives the mid-low frequency audio signal from D/A converter **154**B and amplification commands from power management unit **174**. Mid-Low frequency bandwidth amplifier **160** amplifies the mid-low frequency audio signal, according to the amplification commands and provides the amplified signal to mid-low frequency speaker element **170**.

[**0075**] Low frequency bandwidth amplifier **162** receives the low frequency audio signal from D/A converter **154**B and amplification commands from power management unit **174**. Low frequency bandwidth amplifier **162** amplifies the low frequency audio signal, according to the amplification commands and provides the amplified signal to low frequency speaker element **172**.

[**0076**] It is noted that the amplification commands can be dynamic and can further be influenced by equalization and cross-over processes. Each specific speaker element: **166**, **168**, **170** or **172** receives the amplified analog audio signal and produces respective sound signals.

[**0077**] When a computerized speaker unit such as the one referenced **114**A, is connected to a network, it initiates a "handshake" procedure with the various nodes residing on the network, and provides a list of settings and characteristics thereof. For example, the computerized speaker can provide information relating to the physical structure thereof, frequency responses, amplification ability, harmonic distortion values, protocol support, programmable aspects (e.g., internal equalizer and cross-over modules), cross over configuration (e.g., 1-way, 2-way, 3-way and N-way), and the like.

[**0078**] In most cases, the computerized speaker would perform the handshake procedure with the audio management system, which operates as a communication server. The audio management system receives the settings of the computerized speaker that was just introduced to the network, and updates the main program with these parameters.

[**0079**] A/D converter **151** receives analog signals from microphone **153**, converts them to digital signals and sends the digital signals to local DSP **152**. Microphone **153** continuously transmits audio feedback signal to local DSP **152**. Local DSP **152** compares the signals from communication interface **150** and A/D converter **151**, and controls the operation of speaker elements **166**, **168**, **170** and **172** according to the outcome of this comparison.

[**0080**] Reference is now made to **FIG. 9**, which is an illustration of a method for operating a computerized speaker **114**A of **FIG. 4**, operative in accordance with another preferred embodiment of the present invention. In step **250**, encoded digital audio streams are received. It is noted that the digital audio streams can be received in an un-encoded format, if transmitted as such. Further more, the received streams can include one stream or more, which can further be directed to a specific speaker element as will be described in detail herein below. With reference to **FIG. 4**, communication interface **150** receives encoded digital audio streams, transmitted by audio management system **102** of **FIG. 1**.

[**0081**] In step **252**, the encoded digital audio streams are decoded. It is noted that this step can be eliminated when the digital audio streams are received in an un-encoded format. With reference to **FIG. 4**, local DSP **152** encodes the digital audio streams received from communication interface **150**.

[**0082**] In step **254**, speaker audio control data is received. The received speaker audio control data can include a plurality of predetermined parameters such as volume, equalization, active cross-over, and the like. With reference to **FIG. 4**, local DSP **152** receives the speaker audio control data from communication interface **150**.

[**0083**] In step **256**, the decoded digital audio streams are conformed according to the speaker audio control data. With reference to **FIG. 4**, local DSP **152** conforms the digital audio streams according to the speaker audio control data. For example, local DSP **152** equalizes the digital audio streams, applies acoustic filters and functions, adds audible effects, and the like. In addition, the DSP **152** further applies a cross-over function on the final streams, according to the acoustic specification of the speaker elements, the acoustic design of the speaker chambers and override parameters which can be induced by the audio management system **102** (**FIG. 1**).

[**0084**] In step **258**, the digital audio streams are converted to respective analog audio signals. With reference to **FIG. 4**, each of the D/A converters converts the received digital audio streams in the separate channel. D/A converter **154**A receives the high frequency digital audio stream and the mid-high frequency digital audio stream from local DSP **152** and converts these streams to analog audio signals of the high frequency and mid-high frequency, accordingly. D/A converter **154**B receives the mid-low frequency digital audio stream and the low frequency digital audio stream from local DSP **152** and converts these streams to analog audio signals of the mid-low frequency and low frequency, accordingly.

[**0085**] In step **260**, each of the analog audio signals is amplified respective of its destination speaker element. With reference to **FIG. 4**, power amplifier array **164** amplifies the analog audio signals received from D/A converters **154**A and **154**B. It is noted that local DSP **152** produces amplification commands, designated for each of the amplifiers of power amplifier array **164**, according to the speaker audio control data and frequency limitation for each separate amplifier. The amplified audio signals are provided to each separate speaker, which produce sound.

[**0086**] In step **262**, sounds are detected, thereby a plurality of acoustic parameters is produced. Microphone **124** (**FIG. 2**) can continuously transmit audio feedback to audio management system **102** (**FIG. 1**), and microphone **124** provides feedback audio data for continuous management of sound system **100** (**FIG. 1**) in an automatic mode.

[**0087**] In step **264**, the acoustic parameters are transmitted to a control unit. With reference to **FIG. 2**, microphone **124** transmits these acoustic parameters to DSP **122**. Alternatively, if a computerized speaker includes an integrated microphone, then the DSP thereof can transmit acoustic parameters detected by that microphone to DSP **122**, via network **116**.

[**0088**] Reference is further made to **FIG. 5**, which is a schematic illustration in detail of computerized speaker **114**B of **FIG. 1**, constructed and operative in accordance with a further preferred embodiment of the present invention. Computerized speaker **114**B includes a communication interface **180**, a digital to analog converter (D/A) **182**, an equalizer **184**, a cross-over unit **186**, a power amplifier array **188** and a power management unit **198**. Power amplifier array **188** includes a high frequency bandwidth amplifier **190**, a mid-high frequency bandwidth amplifier **192**, a mid-low frequency bandwidth amplifier **194** and a low frequency bandwidth amplifier **196**.

[**0089**] Communication interface **180** is connected to D/A converter **182**, power management unit **198** and to network **116** (**FIG. 1**). Equalizer **184** is connected to D/A converter **182** and to cross-over unit **186**. Power amplifier array **188** is connected to cross-over unit **186** and to power management unit **198**.

[**0090**] Communication interface **180** receives data from audio management system **102** (**FIG. 1**) via network **116** (**FIG. 1**). This data includes a digital audio stream and amplification commands. Communication interface **180** provides the received digital audio stream to D/A converter **182**, which in turn converts the digital audio stream to analog audio signal and further provides it to equalizer **184**. Communication interface **180** provides the amplification commands to power management unit **198**. Equalizer **184** equalizes the analog signal, according to predetermined equalization parameters, and provides the equalized audio signal to cross-over unit **186**.

[**0091**] Cross-over unit **186** applies a cross-over function to the equalized analog signal, according to predetermined cross-over parameters respective of the structure of computerized speaker **114**B, thereby producing a plurality of analog audio signals. Cross-over unit **186** provides each of the analog audio signals to the respective amplification unit in power amplifier array **188**, which in turn amplifies the analog audio signals according to amplification commands received from power management unit **198**. Power amplifier array **188**, then provides the amplified signal to the respective speaker elements (not shown) connected thereto. It is noted that computerized speaker **114**B can be manufactured and sold without the speaker elements, which can then be added later, according to the preferences of the final user or an end-user system manufacturer. Accordingly, the equalization parameters and cross-over parameters can be modified according to the selected speaker elements, at that later stage.

[**0092**] Reference is now made to **FIG. 11**, which is an illustration of a method for operating computerized speaker **114**B of **FIG. 5**, operative in accordance with another preferred embodiment of the present invention. In step **290**, an encoded digital audio stream is received. It is noted that the digital audio stream can be received in an un-encoded format, if transmitted as such. With reference to **FIG. 5**, communication interface **180** receives an encoded digital audio stream, transmitted by audio management system **102** of **FIG. 1**.

[**0093**] In step **292**, the encoded digital audio stream is decoded. It is noted that this step can be eliminated when the digital audio stream is received in an un-encoded format. With reference to **FIG. 5**, communication interface **180** decodes the digital audio stream, received from audio management system **102** of **FIG. 1**.

[**0094**] In step **294**, the digital audio stream is converted to analog audio signal. With reference to **FIG. 5**, D/A converter **182** converts the digital audio stream, received from communication interface **180** to analog audio signal.

[**0095**] In step **296** the analog audio signal is equalized according to pre-set equalization parameters. It is noted that the equalization parameters can be pre-set in the equalization unit of the computerized speaker according to the acoustic and electronic characteristics of the elements comprising the computerized speaker. With reference to **FIG. 5**, equalizer **184** equalizes the analog audio signal received from D/A converter **182**

[**0096**] In step **298**, a cross-over function is applied to the equalized analog audio signal. The cross-over function is predetermined according to the acoustic and electronic characteristics of the elements comprising the computerized speaker. For example, the cross-over function can be a band-pass, a band-cross, a high-pass, a low-pass digital filter, and the like. With reference to **FIG. 5**, cross-over unit **186** applies a cross-over function to the analog audio signal, received from equalizer **184**.

[**0097**] In step **300**, speaker audio control data is received. The received speaker audio control data includes amplification commands, designated for each of the amplifiers within the digital speaker. It is noted that a single amplification command can be provided to power management unit **198**, and be directed to all the amplifiers in a specific digital speaker. With reference to **FIG. 5**, communication interface **180** receives the speaker audio control data from audio management system **102** of **FIG. 1**.

[**0098**] This amplification command can be provided once, from power management unit **198** to the power amplifier array **188**, for evenly distributing among the amplifiers. A plurality of parameters within the power amplifier array can be controlled. The parameter can be on-off, volume, percentage of maximum gain, and the like.

[**0099**] In step **302**, the analog audio signal is amplified for each specific speaker element. With reference to **FIG. 5**, each of the amplifiers of power amplifier array **188**, amplifies a respective analog audio signal, received from cross-over unit **186**. The amplified audio signals are provided to each separate speaker, which in turn produces sounds.

[**0100**] In step **304**, sounds are detected, thereby producing a plurality of acoustic parameters. Microphone **124** (**FIG. 2**) detects sounds, either in the vicinity of a selected computerized speaker or in a predetermined location. These detected sounds can be used for the purpose of feedback by audio management system **102** (**FIG. 1**).

[**0101**] Reference is further made to **FIG. 6**, which is a schematic illustration of a ring type configured audio system, generally referenced **200**, constructed and operative in accordance with a further preferred embodiment of the present invention. Audio system **200** includes an audio management system **202**, a user interface **204**, an audio source cluster **206**, a network **210** and a plurality of computerized speakers **114**CA, **114**CB, **114**CC, **114**CD, **114**CE and **114**CF.

[**0102**] Audio management system **202** is connected to audio source cluster **206**, user interface **204** and to network **210**. Audio management system **202** is further connected to computerized speakers **114**CA, **114**CB, **114**CC, **114**CD, **114**CE and **114**CF via network **210**. Network **210** and computerized speakers **114**CA, **114**CB, **114**CC, **114**CD, **114**CE and **114**CF form a ring networking structure.

[**0103**] Audio management system **202** receives a plurality of digital audio streams from audio source cluster **206**. User interface **204** provides user commands to audio management system **202** for processing the received audio streams. Examples regarding user commands are volume, balance, equalization, harmonic envelope, echo, audio effects, and the like. Audio management system **202** processes the digital audio streams according to the user commands and transmits the audio streams and the speaker audio control data to remote computerized speakers via network **210**.

[**0104**] Reference is further made to **FIG. 7**, which is a another version of schematic illustration of computerized speaker **114**C, of **FIG. 1**, constructed and operative in accordance with another preferred embodiment of the present invention. Computerized speaker **114**C includes a communication interface **210**, a plurality of local DSPs **212**A, **212**B, **212**C, **212**D, a plurality of digital to analog converters (D/A) **214**A, **214**B, **214**C, **214**D, a plurality of power amplifiers **216**A, **216**B, **216**C and **216**D, and plurality of speaker elements **218**A, **218**B, **218**C and **218**D.

[**0105**] Local DSP **212**A is connected to communication interface **210** and to D/A converter **214**A. Local DSP **212**B is connected to communication interface **210** and to D/A converter **214**B. Local DSP **212**C is connected to communication interface **210** and to D/A converter **214**C. Local DSP **212**D is connected to communication interface **210** and to D/A converter **214**D.

[**0106**] Power amplifier (high) **216**A is connected to D/A converter **214**A and to speaker element **218**A. Power amplifier (mid-high) **216**B is connected to D/A converter **214**B and to speaker element **218**B. Power amplifier (mid-low) **216**C is connected to D/A converter **214**C and to speaker element **218**C. Power amplifier (low) **216**D is connected to D/A converter **214**D and to speaker element **218**D.

[**0107**] Communication interface **210** receives digital audio streams and speaker audio control data, in the form of addressable packets, from audio management system **102** (**FIG. 1**), and provides each of the packets to at least one of the local DSPs **212**A, **212**B, **212**C and **212**C, according to address data embedded in each packet. It is noted that the packet address can include a two stage addressing structure, where the first stage includes the communication interface address and the second stage includes the local DSP address. Alternatively, if each of the local DSP units in the entire set of computerized speakers is unique, then the address can be just of the final destination of the local DSP.

[**0108**] As presented above, a computerized speaker such as computerized speaker **114**C includes a plurality (N) of sound processing and production sections, each including a DSP **212**N, a D/A converter **214**N, a power amplifier **216**N and a speaker elements **218**N, connected in series. N denotes the index of a reference in a series of similar references.

Each of these sections operates in a similar manner. The following description addresses an N section.

[0109] Local DSP **212**N processes the received packets addressed thereto and extracts the received digital audio stream and the speaker audio control data embedded therein. Local DSP **212**N produces a digital audio stream according to the speaker audio control data and provides it to D/A converter **214**N. D/A converter **214**N converts the digital audio stream to an analog audio signal and provides it to power amplifier **216**N. Local DSP **212**N provides amplification commands to power amplifier **216**N, which in turn amplifies the analog audio signal, received from D/A converter **214**N. Power amplifier **216**N provides the amplified audio signal to speaker element **218**N, which in turn produces sound signals.

[0110] Reference is further made to **FIG. 12**, which is a schematic illustration in detail of computerized speaker **114**D of **FIG. 1**, constructed and operative in accordance with a further preferred embodiment of the present invention. The number of the computerized speakers, which can be connected to system **100**, is virtually unlimited. The system supports a plurality of computerized speaker types and further supports plug and play functionalities, which enable seamless introduction of new types of computerized speakers thereto.

[0111] Computerized speaker **114**D includes a communication interface **320**, a digital equalizer **322**, digital to analog converter (D/A) **324**, an active cross-over unit **326**, a power amplifier array **330**, and a power management unit **328**. Power amplifier array **330** includes a high frequency bandwidth amplifier **332**, a mid-high frequency bandwidth amplifier **334**, a mid-low frequency bandwidth amplifier **336** and a low frequency bandwidth amplifier **338**.

[0112] Communication interface **320** is connected to digital equalizer **322**, power management unit **328** and to network **116** (**FIG. 1**). D/A converter **324** is connected to digital equalizer **322** and to active cross-over unit **326**. Power amplifier array **330** is connected to active cross-over unit **326** and to power management unit **328**.

[0113] Communication interface **320** receives data from audio management system **102** (**FIG. 1**) via network **116** (**FIG. 1**). This data includes a digital audio stream, a speaker audio control data and amplification commands. Communication interface **320** provides the received digital audio stream to digital equalizer **322**. Communication interface **320** further provides the amplification commands to power management unit **328**.

[0114] Digital equalizer **322** equalizes the digital audio stream, according to predetermined equalization parameters of the speaker audio control data, and provides the transformed audio stream to D/A converter **324**. D/A converter **324** converts the digital audio stream to analog audio signal and further provides it to active cross-over unit **326**. Active cross-over unit **326** applies a cross-over function to the analog signal, according to predetermined cross-over parameters respective of the structure of computerized speaker **114**D, thereby producing a plurality of analog audio signals. Active cross-over unit **326** provides each of the analog audio signals to the respective amplification unit in power amplifier array **330**, which in turn amplifies it according to amplification commands received from communication interface **320** and provides the amplified signal to the respective speaker elements (not shown) connected thereto.

[0115] An active cross-over unit, according to the present invention, applies a cross-over function to an audio signal, according to predetermined cross-over parameters determined by real time adaptive analyzer **126** (**FIG. 2**), in the main program. Thereby, the active cross-over unit produces a plurality of band specific audio signals. The active cross-over unit provides each of the band specific audio signals to a respective audio channel, for further amplification and production as sound. It is noted that the audio channel may further include D/A conversion, when the band specific audio signals are in digital format. The cross-over unit can further dynamically change the band frequency limitation of the specific audio signals.

[0116] For example, the mid-high frequency bandwidth amplifier of a specific computerized speaker receives from the active cross-over unit the audio signals of the mid-high and the high frequency bands, and the mid-low frequency bandwidth amplifier receives the audio signals of the mid-low and of the low frequency bands. The high frequency bandwidth amplifier and the low frequency bandwidth amplifier of this computerized speaker receive the audio signals of the high and of the low frequency bands, respectively, without the band frequency limitation changes.

[0117] The physical structure of the speaker, the position of the speaker and user predetermined sound parameters determine the band frequency limitation of the produced audio signals. Hence, such an active cross-over unit optimizes sound production, respective of the structure of the computerized speaker and the various speaker elements, which it includes.

[0118] Furthermore, the active cross-over unit can perform out-of-phase and in-phase manipulations of the audio source signals, thereby correcting some phase inequalities, which may be exhibited between the different elements of the computerized speaker. By applying dynamic digitized frequency allocation for each of the speaker elements, the active cross-over unit provides a low level form of equalization to the produced analog signal, (e.g., the active cross-over unit can eliminate some inequalities of the speaker passive elements). The active cross-over unit reduces the amount of energy wasted on the element (such as a passive cross speaker element) at all audio power levels.

[0119] The technology provided by the present invention, allows to perform automatic testing and calibration of each of the sound producing elements thereof (i.e., computerized speakers and computerized speaker elements). The audio management system can automatically adjust the settings of each of the computerized speakers, as well as manipulate the audio streams provided thereto (i.e., network audio control data and speaker audio control data).

[0120] For example, the automatic testing and calibration procedures can set the system for a selected acoustic center. The acoustic center of the audio system is a point with balance of each frequency band sound produced by all computerized speakers. It is noted that the frequency changing and a delay alignment of the produced sound by the computerized speaker can correct the acoustic center.

[0121] Reference is now made to **FIG. 10**, which is an illustration of a method for operating computerized speaker **114**D of **FIG. 12**, operative in accordance with another preferred embodiment of the present invention. In step **270**, an encoded digital audio stream is received. It is noted that the digital audio stream can be received in an un-encoded format, if transmitted as such. With reference to **FIG. 12**, communication interface **320** receives encoded digital audio stream, transmitted by audio management system **102** of **FIG. 1**.

[0122] In step **272**, the encoded digital audio stream is decoded. It is noted that this step can be eliminated when the digital audio stream is received in an un-encoded format. With reference to **FIG. 12**, communication interface **320** encodes the digital audio stream received from audio management system **102** of **FIG. 1**.

[0123] In step **274**, speaker audio control data is received. The received speaker audio control data includes predetermined equalization parameters and amplification commands. With reference to **FIG. 12**, communication interface **320** receives the speaker audio control data from audio management system **102** of **FIG. 1**.

[0124] In step **276** the digital audio streams are conformed according to equalization parameters. It is noted that the equalization parameters can be predetermined or dynamically updated via the network. With reference to **FIG. 12**, digital equalizer **322** equalizes the digital audio stream received from communication interface **320**. Digital equalizer **322** can be pre-programmed or dynamically programmed by means of speaker audio control data, received by communication interface **320**.

[0125] In step **278**, the digital audio stream is converted to an analog audio signal. With reference to **FIG. 12**, D/A converter **324** converts the digital audio stream, received from digital equalizer **322** to analog audio signal.

[0126] In step **280**, a cross-over function is applied on the analog audio signal. The cross-over function is pre-set according to the acoustic and electronic characteristics of the elements comprising the computerized speaker. With reference to **FIG. 12**, active cross-over unit **326** applies active cross-over function to the analog audio signal received from D/A converter **324**.

[0127] In step **282**, the analog audio signal is amplified for each specific speaker element. With reference to **FIG. 12**, power amplifier array **330** amplifies the analog audio signal, received from active cross-over unit **326**. It is noted that communication interface **320** produces amplification commands designated for each of the amplifiers of power amplifier array **330**, according to the received speaker audio control data. The amplified audio signals are provided to each separate speaker element, which in turn produces sound.

[0128] In step **284**, sounds are detected, thereby producing a plurality of acoustic parameters. Microphone **124** (**FIG. 2**) detects sounds, either in the vicinity of a selected computerized speaker or in a predetermined location. These detected sounds can be used for the purpose of feedback by audio management system **102** (**FIG. 1**).

[0129] It will be appreciated by persons skilled in the art that the present invention is not limited to what has been particularly shown and described here in above. Rather the scope of the present invention is defined only by the claims, which follow.

**1**. Audio system for operating a plurality of speakers, the audio system comprising:

an audio management system connected to said speakers via a network;

a user interface connected to said audio management system; and

an audio source cluster connected to said audio management system,

wherein said audio management system operates said speakers by transmitting audio streams and speaker audio control data to said network, and

wherein said audio management system determines said speaker audio control data, according to a plurality of parameters.

**2**. The audio system according to claim 1, wherein a selected ones of said speakers are connected to said network via a plurality of routers.

**3**. The audio system according to claim 1, wherein a selected ones of said speakers are connected to said network via a plurality of wireless transceivers.

**4**. The audio system according to claim 1, wherein the type of said network is selected from the list consisting of:

ring;

star; and

combination of ring and star.

**5**. The audio system according to claim 1, wherein said audio management system transmits said audio streams and said speaker audio control data to each of said speakers, according to an address of a respective one of said speakers.

**6**. The audio system according to claim 1, wherein the type of said parameters is selected from the list consisting of:

automatic;

user defined; and

automatic and user defined.

**7**. The audio system according to claim 1, wherein said audio management system operates according to a digital isochronous protocol selected from the list consisting of:

Universal Serial Bus;

high speed IEEE 1394; and

Short Distance Wireless Blue-tooth format.

**8**. The audio system according to claim 1, wherein said network operates according to a network protocol selected from the list consisting of:

Ethernet; and

ATM.

**9**. The audio system according to claim 1, wherein a user transmits user defined parameters to said audio management system, via said user interface.

**10**. The audio system according to claim 1, wherein said speaker audio control data include at least one audio parameter, selected from the list consisting of:

volume;

balance;

equalization;

harmonic envelop;

echo effects;

audio effects;

phase; and

time-delay.

**11**. The audio system according to claim 1, wherein said audio management system further comprises a digital signal processor, and a central processing unit, said central processing unit connected to said user interface.

**12**. The audio system according to claim 11, wherein said digital signal processor further comprises a real time adaptive analyzer connected to said audio source cluster and to a microphone, and a master band equalizer connected to said real time adaptive analyzer, said central processing unit and to said network via a communication interface, wherein said microphone listens to said speakers, and said real time adaptive analyzer comprises a user defined sound parameter.

**13**. The audio system according to claim 11, wherein said digital signal processor further comprises a real time adaptive analyzer connected to said audio source cluster and to a microphone, a master band equalizer connected to said real time adaptive analyzer, and to said central processing unit, and an audio encoder connected to said network via a communication interface, said audio encoder further connected to said master band equalizer and to said central processing unit, wherein said microphone listens to said speakers, and said audio encoder transmits an encoded speaker element control parameter to said communication interface.

**14**. The audio system according to claim 13, wherein said digital signal processor further comprises an audio decoder connected to said audio source cluster and to said real time adaptive analyzer.

**15**. The audio system according to claim 14, wherein the type of operation of said audio decoder is selected from the list consisting of:

de-multiplexing;

separating said audio streams;

converting said audio streams from a first digital format to a second digital format; and

decompressing said audio streams.

**16**. The audio system according to claim 1, wherein the type of said audio streams is selected from a list consisting of:

Dolby Logic;

Dolby Pro-Logic;

Dolby Digital 5.1 channels;

Digital Theatrical Sound; and

UCPASS multi-channel encoded audio source.

**17**. The audio system according to claim 12, wherein said real time adaptive analyzer transmits parameter sets to said master band equalizer.

**18**. The audio system according to claim 17, wherein the type of said parameter sets is selected from the list consisting of:

quality sharpness of a band-pass filter; and

gain of a filter.

**19**. The audio system according to claim 12, wherein said master band equalizer operates in a mode selected from the list consisting of:

automatic; and

manual.

**20**. The audio system according to claim 1, wherein said user interface comprises a master bass control means and a master treble control means.

**21**. The audio system according to claim 1, wherein a user transmits said parameters to said user interface via an input output device selected from the list consisting of:

video;

audio; and

tactile.

**22**. The audio system according to claim 13 wherein said encoded speaker element control parameter is selected from the list consisting of:

delay;

volume;

equalization; and

cross.

**23**. The audio system according to claim 12, wherein said real time adaptive analyzer maintains a status of last operation.

**24**. The audio system according to claim 12, wherein said user defined sound parameter is selected from the list consisting of:

Concert;

Classical-Hall;

Jazz;

Stadium;

Pop; and

Rock.

**25**. The audio system according to claim 1, wherein said audio source cluster further comprises a plurality of analog audio sources, a plurality of digital audio sources, an analog to digital converter, a sampling rate converter, and a multiplexor,

wherein said analog to digital converter is connected to said analog audio sources and to said multiplexor, said sampling rate converter is connected to said digital audio sources and to said multiplexor, and said multiplexor is connected to said audio management system.

**26**. The audio system according to claim 1, wherein each of said speakers further comprises a communication interface, a local digital signal processor, a first digital to analog converter, a second digital to analog converter, a power management unit, a power amplifier array, a high frequency speaker element, a mid-high frequency speaker element, a mid-low frequency speaker element, and a low frequency speaker element.

**27**. The audio system according to claim 26, wherein said power amplifier array further comprises a high frequency bandwidth amplifier, a mid-high frequency bandwidth amplifier, a mid-low frequency bandwidth amplifier, and a low frequency bandwidth amplifier.

**28**. The audio system according to claim 27, wherein said communication interface is connected to said network and to said local digital signal processor, said local digital signal processor is connected to said first digital to analog converter, said second digital to analog converter and to said power management unit, said power management unit is connected to said power amplifier array, said high frequency bandwidth amplifier is connected to said first digital to analog converter and to said high frequency speaker element, said mid-high frequency bandwidth amplifier is connected to said first digital to analog converter and to said mid-high frequency speaker element, said mid-low frequency bandwidth amplifier is connected to said second digital to analog converter and to said mid-low frequency speaker element, and said low frequency bandwidth amplifier is connected to said second digital to analog converter and to said low frequency speaker element.

**29**. The audio system according to claim 26, wherein said local digital signal processor transmits amplification commands to said first digital to analog converter and to said second digital to analog converter.

**30**. The audio system according to claim 26, wherein said local digital signal processor transmits a characteristic of a respective one of said speakers to said audio management system, via said network.

**31**. The audio system according to claim 30, wherein said characteristic is selected from the list consisting of:

physical structure;

frequency response;

amplification ability;

harmonic distortion values;

protocol support;

programmable aspects; and

cross over configuration.

**32**. The audio system according to claim 1, wherein each of said speakers further comprises a communication interface, a digital to analog converter, an equalizer, a cross-over unit, a power management unit, a power amplifier array, a high frequency speaker element, a mid-high frequency speaker element, a mid-low frequency speaker element, and a low frequency speaker element.

**33**. The audio system according to claim 32, wherein said power amplifier array further comprises a high frequency bandwidth amplifier, a mid-high frequency bandwidth amplifier, a mid-low frequency bandwidth amplifier, and a low frequency bandwidth amplifier.

**34**. The audio system according to claim 33, wherein said communication interface is connected to said network, said digital to analog converter and to said power management unit, said equalizer is connected to said digital to analog converter and to said cross-over unit, said power management unit is connected to said power amplifier array, said high frequency bandwidth amplifier is connected to said cross-over unit and to said high frequency speaker element, said mid-high frequency bandwidth amplifier is connected to said cross-over unit and to said mid-high frequency speaker element, said mid-low frequency bandwidth amplifier is connected to said cross-over unit and to said mid-low frequency speaker element, and said low frequency bandwidth amplifier is connected to said cross-over unit and to said low frequency speaker element.

**35**. The audio system according to claim 32, wherein said audio management system transmits amplification commands to said communication interface via said network.

**36**. The audio system according to claim 35, wherein the type of said amplification commands can be selected from the list consisting of:

on-off;

volume control; and

percentage of maximum gain of said power amplifier array during operation.

**37**. The audio system according to claim 1, wherein each of said speakers further comprises a communication interface, a digital equalizer, a digital to analog converter, an active cross-over unit, a power management unit, a power amplifier array, a high frequency speaker element, a mid-high frequency speaker element, a mid-low frequency speaker element, and a low frequency speaker element.

**38**. The audio system according to claim 37, wherein said power amplifier array further comprises a high frequency bandwidth amplifier, a mid-high frequency bandwidth amplifier, a mid-low frequency bandwidth amplifier, and a low frequency bandwidth amplifier.

**39**. The audio system according to claim 38, wherein said communication interface is connected to said network, said digital equalizer and to said power management unit, said digital to analog converter is connected to said digital equalizer and to said active cross-over unit, said power management unit is connected to said power amplifier array, said high frequency bandwidth amplifier is connected to said active cross-over unit and to said high frequency speaker element, said mid-high frequency bandwidth amplifier is connected to said active cross-over unit and to said mid-high frequency speaker element, said mid-low frequency bandwidth amplifier is connected to said active cross-over unit and to said mid-low frequency speaker element, and said low frequency bandwidth amplifier is connected to said active cross-over unit and to said low frequency speaker element.

**40**. The audio system according to claim 38, wherein said active cross-over unit changes a band frequency limitation of a frequency band of said audio streams according to predetermined cross-over parameters.

**41**. The audio system according to claim 40, wherein type of said band frequency limitation is selected from the list consisting of:

physical structure of a respective one of said speakers;

position of a respective one of said speakers; and

user predetermined sound parameters.

**42**. The audio system according to claim 38, wherein said active cross-over unit performs an out-of-phase manipulation and an in-phase manipulation on said audio streams.

**43**. Method for operating an audio system, the audio system including an audio management system, a user interface, an audio source cluster and a plurality of speakers, wherein the speakers are connected to the audio management system via a network, and the audio management system is connected to the user interface and to the audio source cluster, the method comprising the steps of:

determining network audio control data according to automatic predetermined parameters and user defined parameters;

conforming audio streams according to said network audio control parameters;

encoding said conformed audio streams; and

determining speaker audio control data according to said automatic predetermined parameters and said user defined parameters.

**44**. The method according to claim 43, further comprising a preliminary step of decoding a multiplexed form of digital audio streams.

**45**. The method according to claim 44, further comprising a preliminary step of multiplexing said digital audio streams.

**46**. The method according to claim 45, further comprising a preliminary step of converting said digital audio streams to digital audio streams with a predetermined sampling rate, when said digital audio streams are received from a digital audio source.

**47**. The method according to claim 45, further comprising a preliminary step of converting an analog audio signal to said digital audio streams, when said analog signal is received from an analog audio source.

**48**. The method according to claim 43, comprising a further step of transmitting said encoded audio streams and said speaker audio control data to a selected one of said speakers.

**49**. Method for operating an audio system, the audio system including an audio management system, a user interface, an audio source cluster and a plurality of speakers, each of the speakers including a speaker element, wherein the speakers are connected to the audio management system via a network, and the audio management system is connected to the user interface and to the audio source cluster, the method comprising the steps of:

amplifying an analog audio signal;

providing said amplified analog audio signal to each of said speaker elements;

detecting sounds from each of said speaker elements; and

producing an acoustic parameter respective of said sounds.

**50**. The method according to claim 49, further comprising a preliminary step of converting a digital audio signal to said analog audio signal.

**51**. The method according to claim 50, further comprising a preliminary step of conforming said digital audio signal according to speaker audio control data.

**52**. The method according to claim 51, further comprising a preliminary step of receiving said speaker audio control data.

**53**. The method according to claim 51, further comprising a preliminary step of decoding an encoded digital audio signal.

**54**. The method according to claim 53, further comprising a preliminary step of receiving said encoded digital audio signal.

**55**. The method according to claim 49, further comprising a step of transmitting said acoustic parameters to a control unit.

**56**. Method for operating an audio system, the audio system including an audio management system, a user interface, an audio source cluster and a plurality of speakers, each of the speakers including a speaker element, wherein the speakers are connected to the audio management system via a network, and the audio management system is connected to the user interface and to the audio source cluster, the method comprising the steps of:

performing a cross-over function on an analog audio signal;

amplifying said analog audio signal for each of said speaker elements;

detecting sounds from each of said speaker elements; and

producing an acoustic parameter respective of said sounds.

**57**. The method according to claim 56, further comprising a preliminary step of converting a digital audio signal to said analog audio signal.

**58**. The method according to claim 57, further comprising a preliminary step of conforming said digital audio signal according to speaker audio control data.

**59**. The method according to claim 58, further comprising a preliminary step of receiving said speaker audio control data.

**60**. The method according to claim 58, further comprising a preliminary step of decoding an encoded digital audio signal.

**61**. The method according to claim 60, further comprising a preliminary step of receiving said encoded digital audio signal.

**62**. Method for operating an audio system, the audio system including an audio management system, a user interface, an audio source cluster and a plurality of speakers, each of the speakers including a speaker element, wherein the speakers are connected to the audio management system via a network, and the audio management system is connected to the user interface and to the audio source cluster, the method comprising the steps of:

performing a cross-over function on an equalized analog audio signal;

amplifying said equalized analog audio signal for each of said speaker elements;

detecting sounds from each of said speaker elements; and

producing an acoustic parameter respective of said sounds.

**63**. The method according to claim 62, further comprising a preliminary step of receiving speaker audio control data.

**64**. The method according to claim 62, further comprising a preliminary step of equalizing an analog audio signal.

**65**. The method according to claim 64, further comprising a preliminary step of converting a digital audio signal to said analog audio signal.

**66**. The method according to claim 65, further comprising a preliminary step of decoding an encoded digital audio signal.

**67**. The method according to claim 66, further comprising a preliminary step of receiving said encoded digital audio signal.

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