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(54) Title: INTEGRATED LATENCY DETECTION AND ECHO CANCELLATION

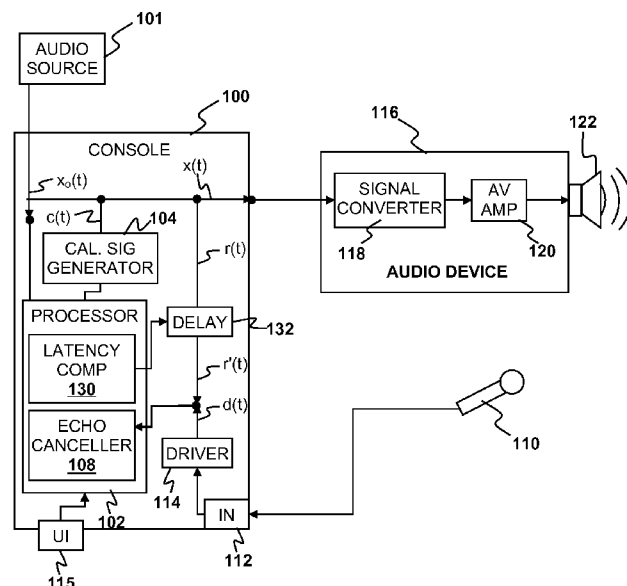


FIG. 1

(57) Abstract: In an audio system having a microphone, a speaker coupled to a source of audio output, and an echo canceller coupled to the speaker and microphone, latency between the source of audio output and the speaker may be compensated in echo cancellation performed by the echo canceller. The echo canceller may use a reference signal derived from a signal from the source of audio output in echo cancellation. The latency may be compensated by measuring the latency between the signal from the source of audio output and the speaker, determining a delay amount from the latency, delaying the reference signal by the delay amount to produce a delayed reference signal, and using the delayed reference signal as the reference signal in the echo canceller.



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INTEGRATED LATENCY DETECTION AND ECHO CANCELLATION

PRIORITY CLAIM

This application claims the benefit of priority of commonly assigned U.S. Patent Application 12/099,030, filed April 07, 2008, the entire disclosures of which are incorporated herein by
5 reference.

FIELD OF THE INVENTION

Embodiments of this invention are related to audio signal processing and more particularly to echo cancellation in audio signal processing.

BACKGROUND OF THE INVENTION

10 Many portable electronics devices, such as interactive video game controllers are capable of handling two-way audio signals. Such a device typically includes a microphone that receives a local speech signal $s(t)$ from a user of the device and a speaker that emits a speaker signal $x(t)$ that is audible to the user. To make the video game controller more compact it is often desirable to place the microphone and speaker relatively close to each other, e.g., within
15 about 10 centimeters of each other. The user, by contrast may be much further from the microphone, e.g., about 3 to 5 meters away. The microphone produces a signal $d(t)$ that includes both the local speech signal $s(t)$ and a speaker echo signal $x_1(t)$. In addition, the microphone may pick up background noise $n(t)$ so that the overall microphone signal $d(t) = s(t) + x_1(t) + n(t)$. Due to the relative proximity of the speaker, the microphone signal $d(t)$
20 may be dominated by the speaker echo signal $x_1(t)$.

Speaker echo is a commonly observed phenomenon in telecommunications applications and echo suppression and echo cancellation are relatively mature technologies. Echo suppressors work by detecting if there is a voice signal going in one direction on a circuit, and then inserting a great deal of loss in the other direction. Usually the echo suppressor at the far-end
25 of the circuit adds this loss when it detects voice coming from the near-end of the circuit. This added loss prevents the speaker signal $x(t)$ from being retransmitted in the local speech signal $d(t)$. Echo cancellation may implemented as part of a software based on a room impulse model that includes the effects of the acoustics of the room in which the microphone and speaker are located. Such echo cancellation often uses the speaker signal $x(t)$ as a
30 reference.

The Acoustic Echo Cancellation (AEC) process works as follows. The received speaker signal $x(t)$ is digitally sampled to form a reference signal $r(t)$. The reference signal $r(t)$ is then

used to produce sound with the speaker. The microphone picks up the resulting direct path sound, and consequent reverberant sound which is converted to the microphone signal $d(t)$. The microphone signal $d(t)$ is digitally sampled and filtered to extract the echo signal $x_1(t)$. The reference signal and echo signal are compared. The reference signal is summed with the
5 echo signal at 180° out of phase. Again, in an ideal system this results in a perfect cancellation. This process continues for every sample.

There are two main issues that echo cancellers must deal with. The first is the changes and additions to the original signal caused by imperfections of the loudspeaker, microphone, reverberant space and physical coupling. The second is the changing nature of those changes.

10 See below. The first problem is dealt with by the room model, which models the acoustic space in the time and frequency domains. AEC (Acoustic Echo Cancellation) algorithms approximate the result of the next sample by comparing the difference between the current and previous samples. Specifically, a sound may be sampled pre-speaker and post microphone, then compared for initial differences in frequencies, and frequencies that are
15 longer than they were in the original sample. This may be visualised by a Fourier Transform. The resulting information may then be used to predict how the next sound will be altered by the acoustic path. The model of the acoustic space (sometimes referred to as the room model) is therefore continually updated.

The changing nature of the sampled signal is mainly due to changes in the acoustic
20 environment, not the characteristics of loudspeaker, microphone or physical coupling. These are from moving objects in the environment, and movement of the microphone within that environment. For this reason, the cancellation algorithm also has a degree of aggressive adaptation called Non-Linear Processing. This allows the algorithm to make changes to the model of the acoustic path that are suggested, but not yet confirmed by comparison of the two
25 signals.

Echo cancellation algorithms can tolerate a certain amount of latency between the speaker signal $x(t)$ and the microphone signal $d(t)$. However, this latency may sometimes be larger than anticipated.

It is within this context that embodiments of the present invention arise.

30 BRIEF DESCRIPTION OF THE DRAWINGS

The teachings of the present invention can be readily understood by considering the following detailed description in conjunction with the accompanying drawings, in which:

FIG. 1 is a schematic diagram illustrating an audio system according to an embodiment of the present invention.

FIG. 2A is a flow diagram illustrating a method for compensating for latency according to an embodiment of the present invention.

- 5 FIG. 2B is a flow diagram illustrating a method for measuring latency according to an embodiment of the present invention.

FIG. 3 is a schematic diagram of an audio system according to an alternative embodiment of the present invention.

DESCRIPTION OF THE SPECIFIC EMBODIMENTS

- 10 Although the following detailed description contains many specific details for the purposes of illustration, anyone of ordinary skill in the art will appreciate that many variations and alterations to the following details are within the scope of the invention. Accordingly, examples of embodiments of the invention described below are set forth without any loss of generality to, and without imposing limitations upon, the claimed invention.
- 15 FIG. 1 illustrates both the problem and the solution to hardware latency associated with echo cancellation. As seen in FIG. 1 an audio signal processing console **100** may include a processor **102**, calibration signal generator **104**. The processor **102** may be programmed with suitable software **108**, which may implement echo cancellation, e.g., as described above. Specifically, the software may sample a reference signal $r(t)$ derived from a speaker signal
- 20 $x_o(t)$, which may be remotely generated, e.g., by an audio source **101**. Alternatively, the speaker signal $x_o(t)$ may be generated internally within the console device **100**, e.g., in response to software, e.g., video game software, executed by the processor **102**. The echo cancellation software **108** may also sample a microphone signal from a microphone **110**. By way of example, the microphone signal may be coupled to the console **100** through an
- 25 interface **112** such as a Universal Serial Bus (USB) interface. Alternatively, the interface **112** may be a wireless interface, such as a Bluetooth interface. The interface **112** may be coupled to a driver **114** that generates a microphone signal $d(t)$ that is coupled to the processor **102**. The driver **114** may be implemented in hardware or software or some combination of hardware and software.

The sound signal $x_o(t)$ is coupled from an audio output **115** on the console to an external audio device **116**. By way of example the audio device may be a stereo system or television set. The audio device **116** generally includes a signal converter **118** coupled to an amplifier **120** and a speaker **122** coupled to the amplifier. By way of example, and without limitation, the signal converter **118** may be of any conventional type, e.g., High Definition Multimedia interface (HDMI), Sony/Philips Digital Interconnect Format (S/PDIF) or digital to analog interface. HDMI is a licensable compact audio/video connector interface for transmitting uncompressed digital streams. HDMI connects DRM-enforcing digital audio/video sources such as a set-top box, an HD DVD disc player, a Blu-ray Disc player, a personal computer, a video game console, or an AV receiver to a compatible digital audio device and/or video monitor such as a digital television (DTV). S/PDIF specifies a Data Link Layer protocol and choice of Physical Layer specifications for carrying digital audio signals between devices and stereo components. The audio output **115** on the console **102** may include a suitable releasable connector to facilitate coupling to the interface **118** so that the speaker may be removeably coupled to the console **100**.

It is noted that the console **100** and the audio device **116** may be made by different manufacturers. Certain console devices, such as video game system consoles, must often be compatible with many different types of audio devices having signal processing parameters that are unknown a priori. One important parameter is the signal latency introduced by the interface **118**, the amplifier **120** and even the speaker **122**. As used herein this signal latency refers to a time delay between arrival of the signal at the audio output **115** and generation of sound by the speaker **122**. As noted above, the echo cancellation algorithm implemented by the software **108** can typically accommodate a finite latency. Conventional echo cancellation algorithms, such as those based on a room model, typically take latency into account, but use the latency for a different purpose. Conventional echo-cancellers are mostly based on finite impulse response (FIR). The provided latency can be used to simulate the FIR delay tabs. In general, a good FIR in echo-canceller needs some small latency to meet FIR causality. There are a number of different possible echo cancellation schemes that may be used in conjunction with embodiments of the present invention. One possible scheme has been described above. In another possible echo cancellation scheme, an integrated echo and noise-canceller may have two functionally identical filters with orthogonal controls and representations. In such a configuration the two orthogonal filters complement each other in a way that can boost the overall system robustness in a noisy hands free voice

communication. Such an echo and noise cancellation scheme is described in detail in commonly assigned US Patent Application Publication 20070274535, the entire contents of which are incorporated herein by reference. In particular, the echo canceller **108** may be an integrated echo-noise-canceller uses two separately controlled sub-systems in parallel, each with orthogonal control mechanism. The echo-noise-canceller includes a front echo canceller and a backup echo canceller. The front echo-canceller uses double-talk detection and takes conservative adaptation approaches to ensure it is robust against local voice, yet provide less echo-suppression and slow adaptation speech to echo-change. The backup echo-canceller uses cross-correlation to measure the similarity between the error signal and echo signal, it takes an aggressive strategy to ensure the filter is quickly updated and provides large echo-suppression, yet it is unstable to local voice/noise, due to possible over adaptation. An integration of these two echo-canceller outputs is based on cross-correlation analysis that measures which echo canceller has a larger differentiation against the echo signal. The integration also checks the filter stability of both echo cancellers. If one filter is overestimated or underestimated it is then complemented by the other filter. Such a system is designed to ensure there is at one filter working correctly at any given time.

An integrated echo-noise canceller may optionally include an echo-residual-noise-estimator that takes a similar approach using two independent sub estimators in parallel, each with orthogonal controlling. A first estimator is based on echo-distance-mismatch that depends on a robust double-talk-detector. The first estimator is relatively accurate yet not stable due to double-talk-detection errors. A second estimator is based on cross-spectrum-analysis. Its estimation is biased but stable and consistent without depending on local voice detection. The integration of these two estimations of the residual echo takes a minimum/maximum approach for far-end -talk only and double-talk respectively.

The audio device **116** may have an (unknown) latency that is significantly larger than that which the echo cancellation algorithm in the software **108** can tolerate. By way of example, the echo cancellation algorithm in the software **108** may tolerate a latency of around 160 milliseconds. If the latency introduced by the interface **118**, amplifier **120**, and speaker **122** is greater than that which can be accommodated by the echo cancellation in the software, the echo canceller will not work. This is particularly a problem for A/V chat. It is theoretically possible to adjust the latency tolerance of the echo cancellation algorithm implemented by the software **108**. However, it is difficult for a conventional echo cancellation algorithm to adjust the amount of latency that is tolerable without resorting to massive computation.

The solution to this problem is to detect the signal latency introduced between the audio output **115** on the console **100** and the speaker **122** and remove the impact of that latency by applying an appropriate delay to the reference signal $r(t)$ that is used by the echo cancellation algorithm in the software **108**. This solution allows a conventional echo canceller to be used in conjunction with any audio device for which the latency can be measured. It is noted that the echo cancellation algorithm in the software **108** need not be changed. To this end the console **100** may include a latency compensator **130**. By way of example, and without loss of generality, the latency compensator **130** may be implemented in software running on the processor **102**. Alternatively, the latency compensator **130** may be implemented in hardware or in some combination of hardware and software. Operation of the latency compensator may be understood with respect to the flow diagram of FIG. 2A, which illustrates a latency compensation method **200**, which may be used in an audio system having a microphone, a speaker coupled to a source of audio output, and an echo canceller coupled to the speaker and microphone. The method compensates for latency between the source of audio output and the speaker in echo cancellation performed by the echo canceller, which uses a reference signal derived from a signal from the source of audio output in echo cancellation.

As indicated at **202** a latency between the signal from the source of audio output and the speaker is measured. There are a number of different ways of measuring latency. One possible way, among others is the latency measurement method **210** shown in FIG. 2B.

Specifically, a calibration signal may be generated at **212**. The calibration signal generator **104** on the console **100** may be used for this purpose. The calibration signal generator **104** may be implemented in software or hardware or some combination of software and hardware. In a preferred embodiment, the calibration signal is a white noise signal. As used herein, the expression "white noise" refers to a random signal characterized by a flat power spectral density. In other words, the signal's power spectral density has equal power in any band, at any centre frequency, having a given bandwidth.

Because white noise is broadband, latency may be detected for every frequency in broadband calibration signal spectrum. Narrow-band noise signals (sometimes called pink noise or sinusoid noise), by contrast are only useful for detecting latency at some particular frequencies. As noted above, the latency of the audio device **116** at any frequency is generally a priori unknown. It is desirable to ensure that this latency is properly taken into account. For example, some AV amplifiers may introduce large latency at low frequencies

for a low frequency effects (LFE) channel, but not at higher frequencies. The low frequency latency might not be detected if the signal from calibration signal generator does not include a sufficient low-frequency component. By using a broad-band spectrum calibration signal, the system can detect latency for any possible AV device **116**.

- 5 As indicated at **214**, the calibration signal $c(t)$ may be applied to the speaker **122**. By way of example the calibration signal $c(t)$ may be applied to the signal converter **118**, which is coupled between the source of audio output and the speaker **122**. Sound is generated from the calibration signal $c(t)$ the speaker **122** as indicated at **216**. The signal converter **118**, amplifier **120** and speaker **122** may be thought of as a “black box” into which the audio
- 10 output signal $x(t)$ is applied and from which sound emerges. The sound is captured with the microphone **110** and converted to a microphone signal $d(t)$ as indicated at **218**. The calibration signal $c(t)$ is then compared to the microphone signal at **220** to determine the signal latency L due to the signal converter **118**, amplifier **120** and speaker **122**.

- The latency may be determined from the microphone signal and the audio output signal by
- 15 performing a cross-correlation analysis between the original signal and recorded signal, and determining the maximum correlation point. This process, sometimes known as “time-delay-arrival (TDA) analysis”, is widely used in radar, sonar and microphone array applications.

- Referring again to FIG. 2A, a delay amount Δ may be determined from the latency L as indicated at **204**. The delay amount may be determined by comparing the measured latency L
- 20 to a latency tolerance L_{tol} of the echo canceller **108**. By way of example, if the measured latency L is within the tolerance of the echo canceller **108** then the delay amount Δ may be zero or some small amount that is less than the latency tolerance L_{tol} . If the measured latency L is greater than the latency tolerance L_{tol} , the delay amount Δ may be between $L - L_{tol}$ and L .

- The delay amount Δ may be determined experimentally from the measured latency L . AV
- 25 equipment latency usually tends to follow a standard. For example, most HDMI AV equipment has 150 msec latency. AV equipment with a significant amount of optical connection introduces about 32 to 64 msec latency. One may take a conservative approach to setting the delay amount Δ . For example, if the detected latency is 250 msec, the delay amount Δ may be 150 msec. In some embodiments, the console **100** may include a user
- 30 interface **115** coupled to the processor **102** (and therefore to the latency compensator **130**). The latency compensator **130** may be configured (e.g., through suitable programming of the

processor **102**) to adjust the delay amount in response to a signal from the user input **115**. Thus, a user may fine tune the delay amount Δ in response to the perceived amount of echo. For example, if 150 microsecond delay amount Δ does not work well, a user may increase the delay amount Δ to 200 msec. It is noted that a precise estimation of the latency and the
5 delay amount are not required as long as the echo-canceller **108** works sufficiently well.

The reference signal may then be delayed by the delay amount to produce a delayed reference signal $r'(t)$ as indicated at **206**. By way of example, the console may include a delay generator **132** for this purpose. The delay generator **132** may be implemented in hardware or in software or some combination of hardware and software. As indicated at **208**, the delayed
10 reference signal $r'(t)$ may then be used as the reference signal in the echo canceller **108**. By way of example, the echo canceller **108** may compare the delayed reference signal $r'(t)$ with the microphone signal $d(t)$, extract an echo signal $x_1(t)$ from the microphone signal $d(t)$ and subtract the echo signal $x_1(t)$ from the microphone signal $d(t)$.

According to embodiments of the present invention, a signal processing method of the type
15 described above with respect to FIG. 1, and FIG. 2 operating as described above may be implemented as part of a signal processing apparatus **300**, as depicted in FIG. 3. The system **300** may include a processor **301** and a memory **302** (e.g., RAM, DRAM, ROM, and the like). In addition, the signal processing apparatus **300** may have multiple processors **301** if parallel processing is to be implemented. The memory **302** includes data and code
20 configured as described above. Specifically, the memory **302** may program code **304** and signal data **306**. The code **304** may implement the echo canceller **108**, latency compensator **130**, calibration signal generator **132** described above. The signal data **306** may include digital representations of the microphone signal $d(t)$ and/or the speaker signal $x(t)$.

The apparatus **300** may also include well-known support functions **310**, such as input/output
25 (I/O) elements **311**, power supplies (P/S) **312**, a clock (CLK) **313** and cache **314**. The apparatus **300** may optionally include a mass storage device **315** such as a disk drive, CD-ROM drive, tape drive, or the like to store programs and/or data. The controller may also optionally include a display unit **316** and user interface unit **318** to facilitate interaction
30 between the controller **300** and a user. The display unit **316** may be in the form of a cathode ray tube (CRT) or flat panel screen that displays text, numerals, graphical symbols or images. The user interface **318** may include a keyboard, mouse, joystick, light pen or other device. In addition, a speaker **322** and a microphone **324** may be coupled to the processor **302** via the

I/O elements **311**. The processor **301**, memory **302** and other components of the system **300** may exchange signals (e.g., code instructions and data) with each other via a system bus **320** as shown in FIG. 3.

As used herein, the term I/O generally refers to any program, operation or device that
5 transfers data to or from the system **300** and to or from a peripheral device. Every data transfer may be regarded as an output from one device and an input into another. Peripheral devices include input-only devices, such as keyboards and mice, output-only devices, such as printers as well as devices such as a writable CD-ROM that can act as both an input and an output device. The term “peripheral device” includes external devices, such as a mouse,
10 keyboard, printer, monitor, microphone, game controller, camera, external Zip drive or scanner as well as internal devices, such as a CD-ROM drive, CD-R drive or internal modem or other peripheral such as a flash memory reader/writer, hard drive.

The processor **301** may perform digital signal processing on signal data **306** as described above in response to the data **306** and program code instructions of a program **304** stored and
15 retrieved by the memory **302** and executed by the processor module **301**. Code portions of the program **304** may conform to any one of a number of different programming languages such as Assembly, C++, JAVA or a number of other languages. The processor module **301** forms a general-purpose computer that becomes a specific purpose computer when executing programs such as the program code **304**. Although the program code **304** is described herein
20 as being implemented in software and executed upon a general purpose computer, those skilled in the art will realize that the method of task management could alternatively be implemented using hardware such as an application specific integrated circuit (ASIC) or other hardware circuitry. As such, it should be understood that embodiments of the invention can be implemented, in whole or in part, in software, hardware or some combination of both.

25 In one embodiment, among others, the program code **304** may include a set of processor readable instructions that implement a method having features in common with the method **200** of FIG. 2A or the method **210** of FIG. 2B. The program code **304** may generally include echo canceller instructions **305** that, when executed by the processor **301**, use a reference signal derived from a signal from a source of audio output in echo cancellation of a signal
30 from the microphone **324**. The program code **304** may also include latency compensation instructions **307** that, when executed by the processor **301**, measure a latency between the signal from the source of audio output and the speaker **322**, determine a delay amount from

the latency and delay the reference signal by the delay amount, whereby the echo canceller uses the delayed reference signal as the reference signal for echo cancellation of the signal from the microphone. Specifically, the user interface unit **318** may be used in conjunction with the program code **304** to allow a user to manually adjust the delay amount in order to
5 optimize the performance of the echo canceller.

Embodiments of the present invention allow for more robust yet accurate echo cancellation without the burden of massive computation for latency compensation within the echo canceller. Consequently, the echo compensation algorithm need not be modified since the delay is applied to the input reference signal used by the echo canceller. Embodiments of the
10 invention provide a simple yet robust solution to the problem of signal latency due to the unknown signal latency of external audio devices coupled to a console that implements applications that require echo cancellation. Embodiments of the present invention therefore allow for robust echo cancellation in applications such as audio visual chat in online video games.

15 While the above is a complete description of the preferred embodiment of the present invention, it is possible to use various alternatives, modifications and equivalents. Therefore, the scope of the present invention should be determined not with reference to the above description but should, instead, be determined with reference to the appended claims, along with their full scope of equivalents. Any feature described herein, whether preferred or not,
20 may be combined with any other feature described herein, whether preferred or not. In the claims that follow, the indefinite article “A”, or “An” refers to a quantity of one or more of the item following the article, except where expressly stated otherwise. The appended claims are not to be interpreted as including means-plus-function limitations, unless such a limitation is explicitly recited in a given claim using the phrase “means for.

WHAT IS CLAIMED IS:

- 1 1. In an audio system having a microphone, a speaker coupled to a source of audio output,
2 and an echo canceller coupled to the speaker and microphone, a method for compensating
3 for latency between the source of audio output and the speaker in echo cancellation
4 performed by the echo canceller, wherein the echo canceller uses a reference signal
5 derived from a signal from the source of audio output in echo cancellation, the method
6 comprising:
7 a) measuring a latency between the signal from the source of audio output and the
8 speaker;
9 b) determining a delay amount from the latency;
10 c) delaying the reference signal by the delay amount to produce a delayed reference
11 signal; and
12 d) using the delayed reference signal as the reference signal in the echo canceller.
- 1 2. The method of claim 1, wherein a) includes:
2 generating a calibration signal;
3 applying the calibration signal to the speaker;
4 generating sound from the calibration signal with the speaker;
5 capturing the sound with the microphone and converting the sound to a microphone
6 signal; and
7 comparing the calibration signal to the microphone signal.
- 1 3. The method of claim 2, wherein applying the calibration signal to the speaker and
2 generating sound from the calibration signal with the speaker include applying the
3 calibration signal to a signal converter coupled between the source of audio output and
4 the speaker.
- 1 4. The method of claim 1 wherein the calibration signal is a white noise signal.
- 1 5. The method of claim 1 wherein the delay amount is determined by comparing the latency
2 determined in b) to a latency tolerance of the echo canceller.

- 1 6. The method of claim 1 wherein d) includes comparing the delayed reference signal with
2 the microphone signal, extracting an echo signal from the microphone signal and
3 subtracting the echo signal from the microphone signal.
- 1 7. An audio system operable with a source of audio output, a speaker and a microphone,
2 comprising:
3 a microphone input configured to receive an audio signal from the microphone;
4 a speaker output configured to be coupled to the source of audio output;
5 an echo canceller coupled to the speaker output and the microphone input, wherein the
6 echo canceller is configured to use a reference signal derived from a signal from the
7 source of audio output in echo cancellation,
8 a latency compensator configured to measure a latency between the signal from the
9 source of audio output and the speaker, determine a delay amount from the latency and
10 delay the reference signal by the delay amount, whereby the echo canceller uses the
11 delayed reference signal as the reference signal for echo cancellation of a signal from the
12 microphone.
- 1 8. The audio system of claim 7, further comprising a calibration signal generator coupled to
2 the latency compensator and the speaker output wherein the calibration signal generator is
3 configured to generate a calibration signal and apply the calibration signal to the speaker
4 output.
- 5
1 9. The audio system of claim 8 wherein the calibration signal is a white noise signal.
- 1 10. The audio system of claim 8 wherein the speaker output is configured to receive a
2 removable connection to the speaker.
- 1 11. The audio system of claim 8 wherein the removable connection is connection for a HDMI
2 or S/PDIF interface.
- 1 12. The audio system of claim 7, further comprising a user interface coupled to the latency
2 compensator.
- 1 13. The audio system of claim 12 wherein the latency compensator is configured to adjust the
2 delay amount in response to a signal from the user input.

1 14. An audio system comprising:
2 a microphone input configured to receive an audio signal from the microphone;
3 a speaker output configured to be coupled to a source of audio output;
4 a computer processor coupled to the speaker output and the microphone input; and
5 computer readable instructions executable by the processor, wherein the computer
6 readable instructions include
7 echo canceller instructions that, when executed by the processor, use a reference signal
8 derived from a signal from the source of audio output in echo cancellation of a signal
9 from the microphone; and
10 latency compensation instructions that, when executed by the processor, measure a
11 latency between the signal from the source of audio output and the speaker, determine a
12 delay amount from the latency and delay the reference signal by the delay amount,
13 whereby the echo canceller uses the delayed reference signal as the reference signal for
14 echo cancellation of the signal from the microphone.

1 15. The audio system of claim 14 wherein the latency compensation instructions include
2 instructions include instructions that, when executed by the processor, generate a
3 calibration signal, apply the calibration signal to a signal converter coupled between the
4 source of audio output and the speaker, capture sound and compare the calibration signal
5 applied to the signal converter to a microphone signal containing sound generated from
6 the calibration signal with the signal converter and the speaker.

1 16. The audio system of claim 15 wherein the calibration signal is a white noise signal.

1 17. The audio system of claim 14, further comprising a user interface coupled to the
2 processor.

1 18. The audio system of claim 17 wherein the compensation instructions are configured to
2 adjust the delay amount in response to a signal from the user input.

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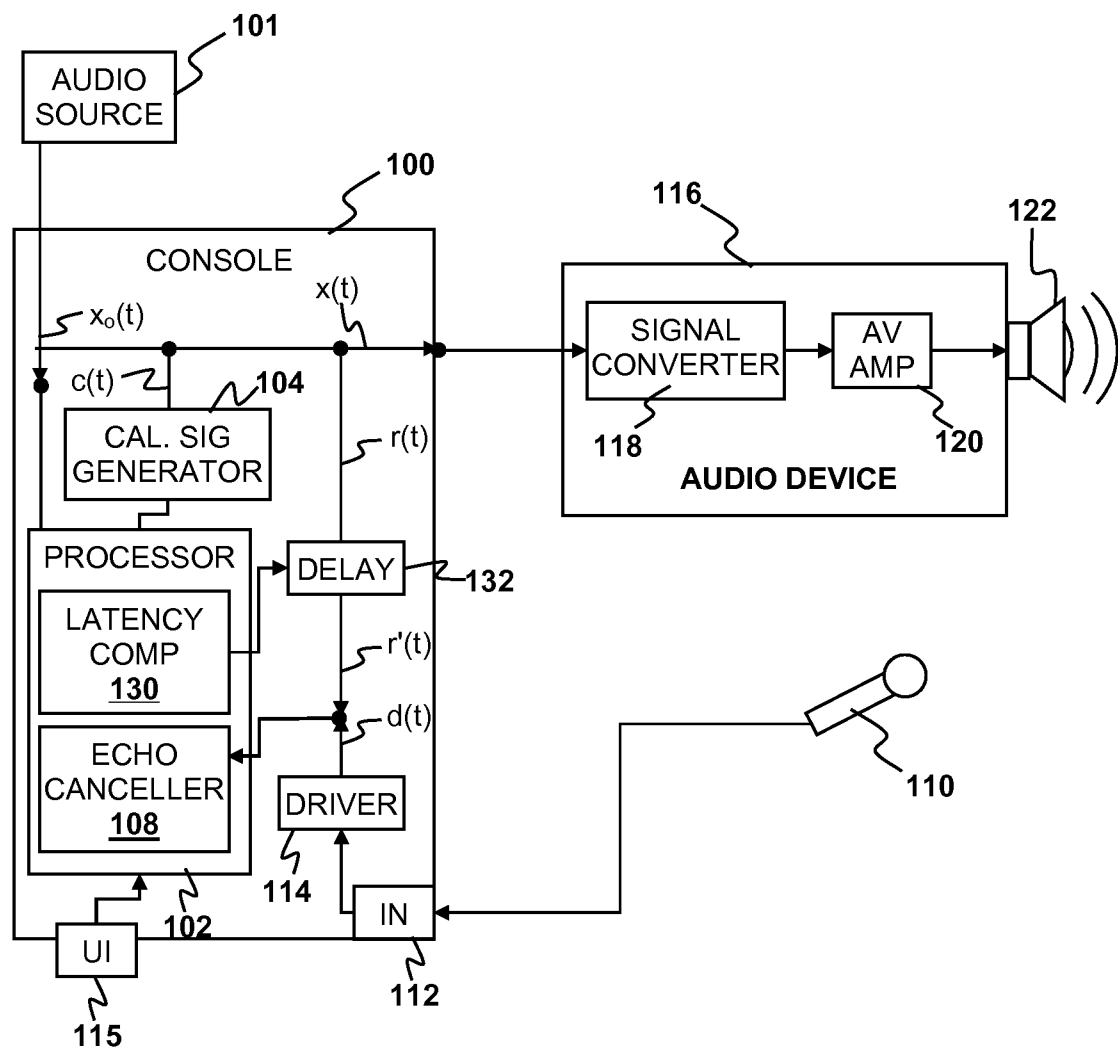


FIG. 1

2/3

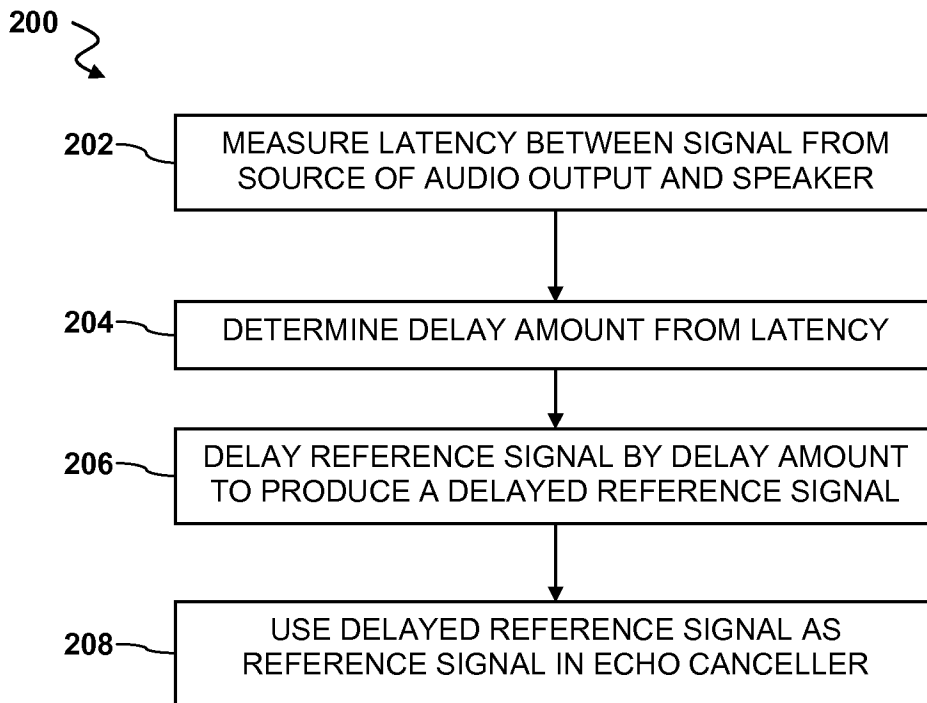


FIG. 2A

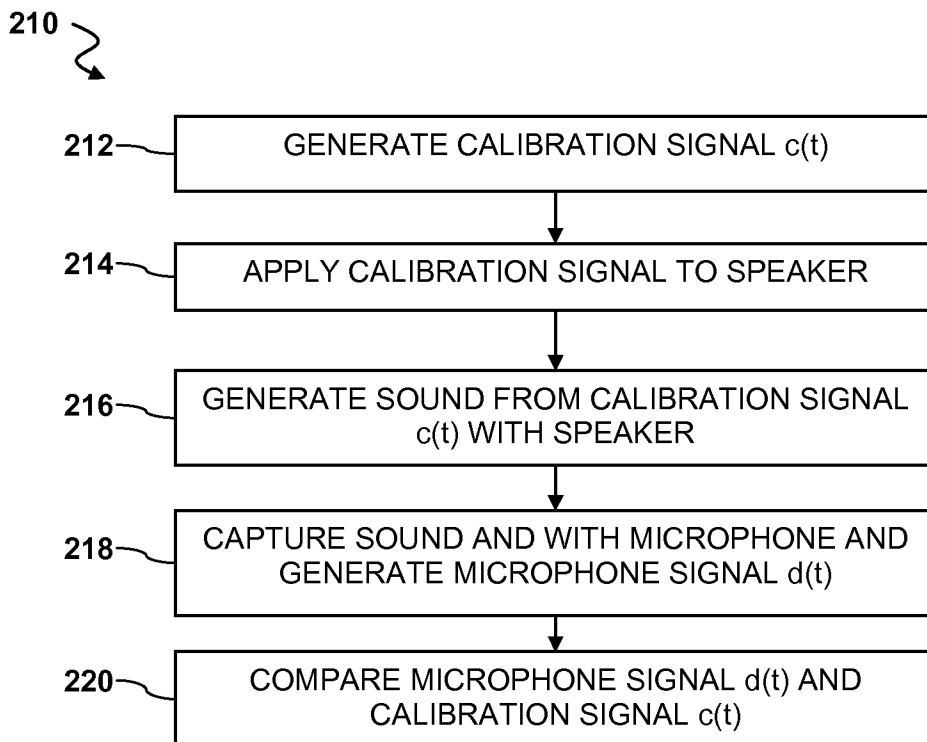


FIG. 2B

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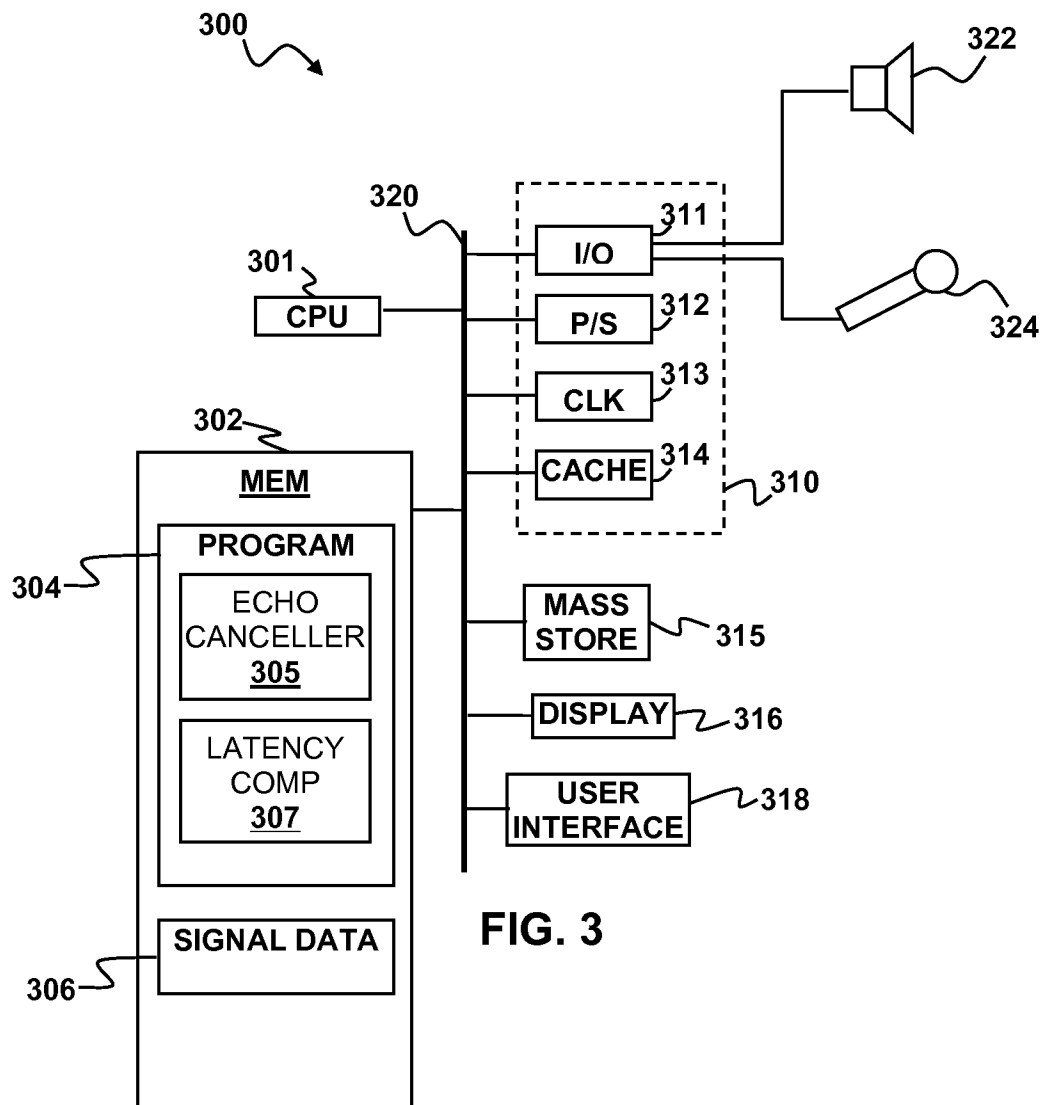


FIG. 3