(12) UK Patent Application (19) GB (11) 2613248 (13) A

(43) Date of A Publication **31.05.2023**

(21) Application No: **2215286.2**

(22) Date of Filing: **24.07.2020**

Date Lodged: **17.10.2022**

(62) Divided from Application No
2011548.1 under section 15(9) of the Patents Act 1977

(51) INT CL:
G10L 19/008 (2013.01) *H04S 5/00* (2006.01)

(56) Documents Cited:
WO 2015/047466 A2 **US 20160249138 A1**
US 20130202119 A1

(58) Field of Search:
Other: **WPI, EPODOC**

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(54) Title of the Invention: **Method and device for processing and providing audio information using bi-phasic separation and re-integration**
Abstract Title: **Processing audio information using a filter which implements a mathematical construct configured for bi-phasic separation of the audio input**

(57) Samples, each having a real and imaginary component, are generated of a waveform corresponding to an audio input. A filter implements a mathematical construct which is configured for bi-phasic separation of the audio input, and which is applied to the samples to generate an output (106A,B) comprising processed renderings of both real components and imaginary components of the samples. The mathematical construct may be a Hilbert transformation, or the filter may be a biquad filter. There may be two audio inputs, for which a first and second set of samples are generated on respective channels, and respective outputs generated and provided to a user. The filter may be implemented as a four-stage filter, where each stage employs a feedback mechanism to eliminate redundancies. The audio output may be converted to an analog form prior to providing the output to the user. User input may determine whether processing is done.

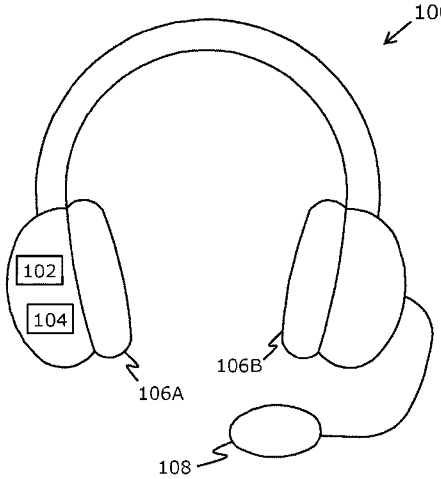

FIG. 1B

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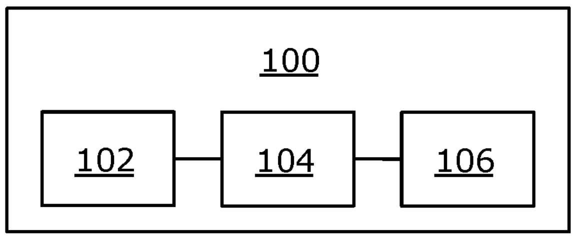

FIG. 1A

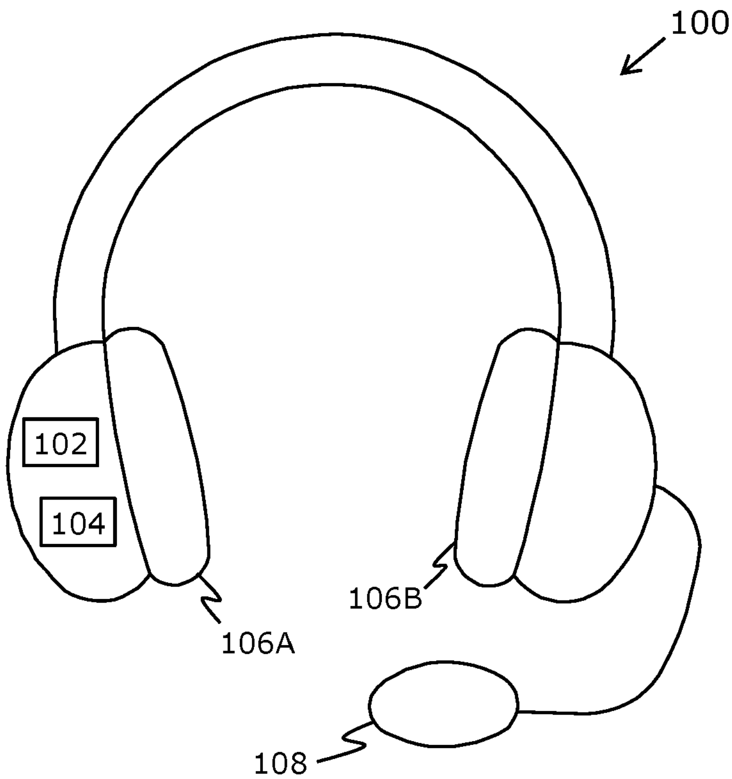

FIG. 1B

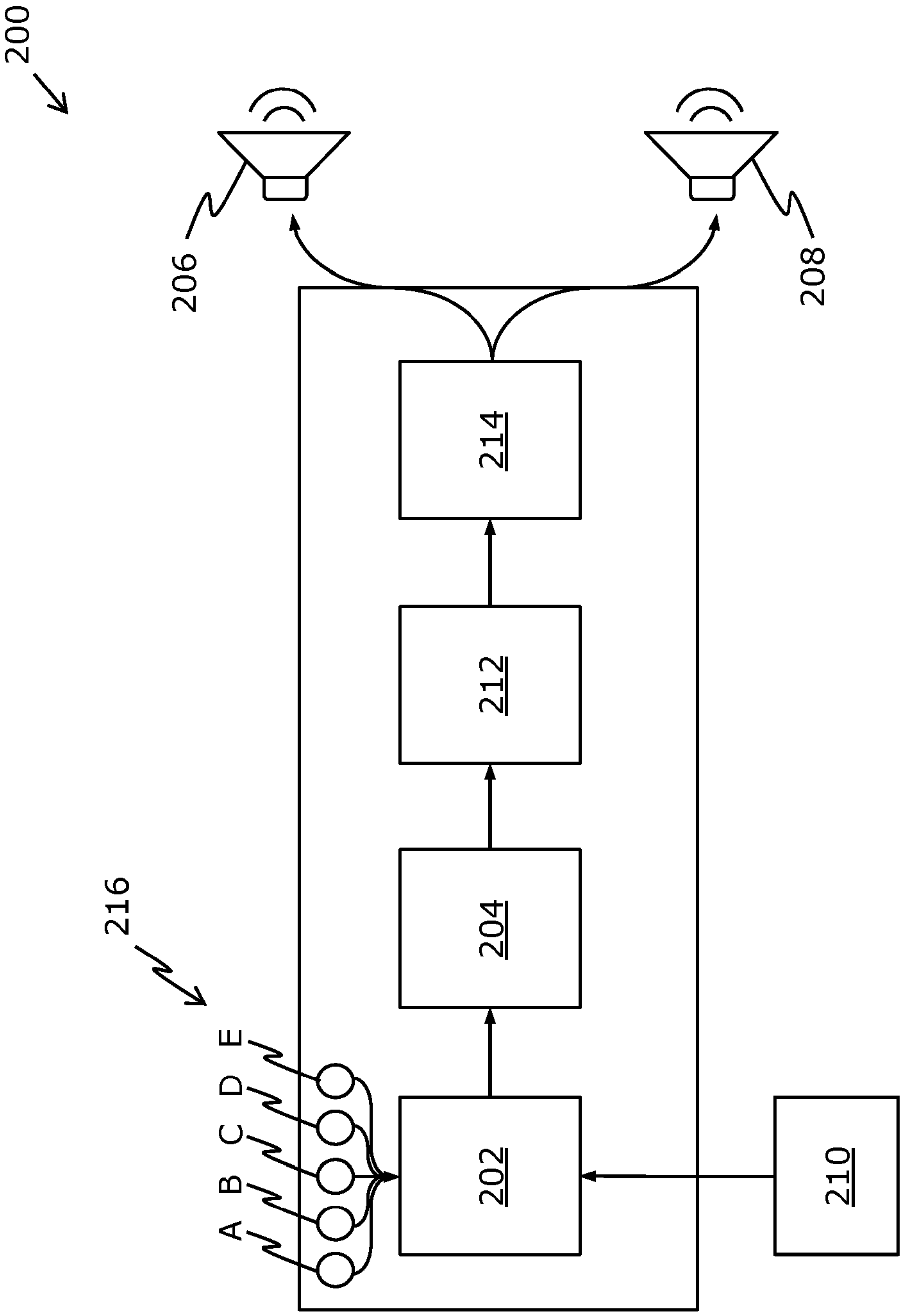

FIG. 2

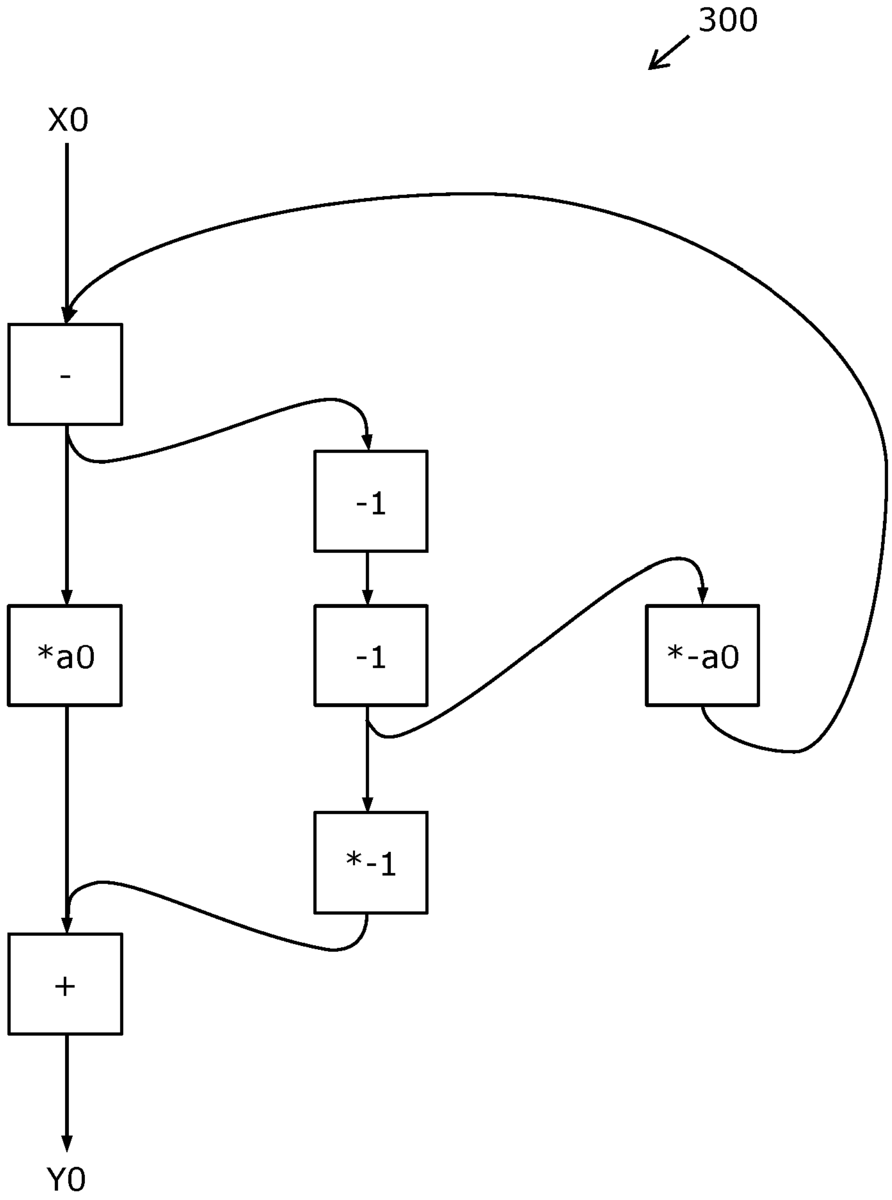

FIG. 3

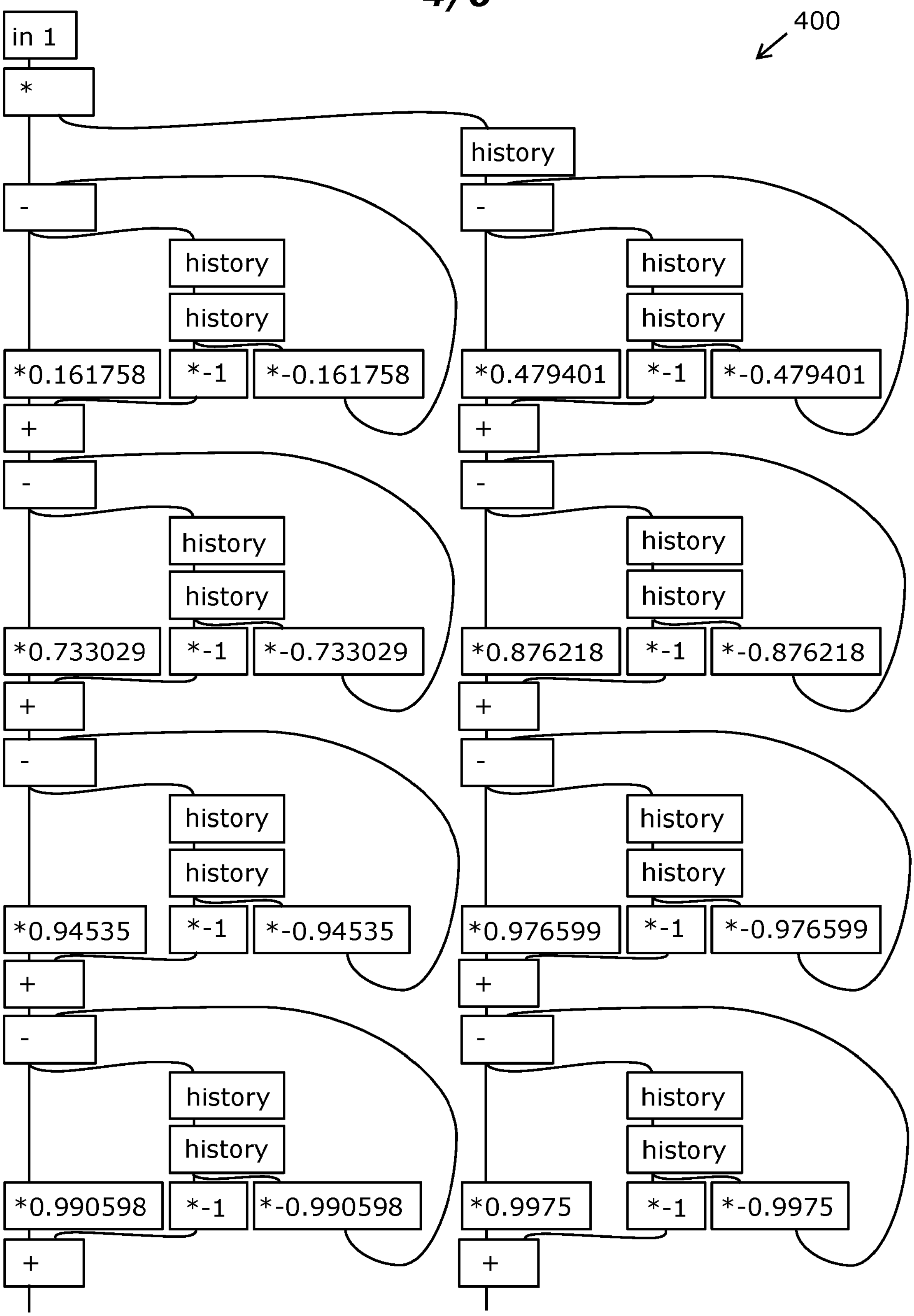
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FIG. 4

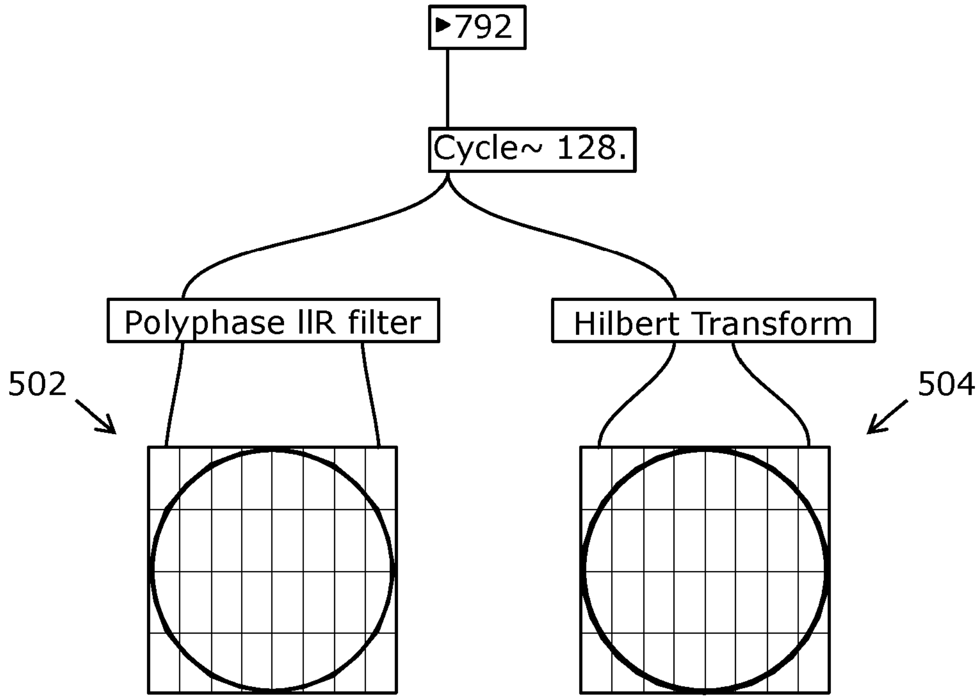
▶792
Cycle~ 128.
Polyphase IIR filter
Hilbert Transform
502
504

FIG. 5A

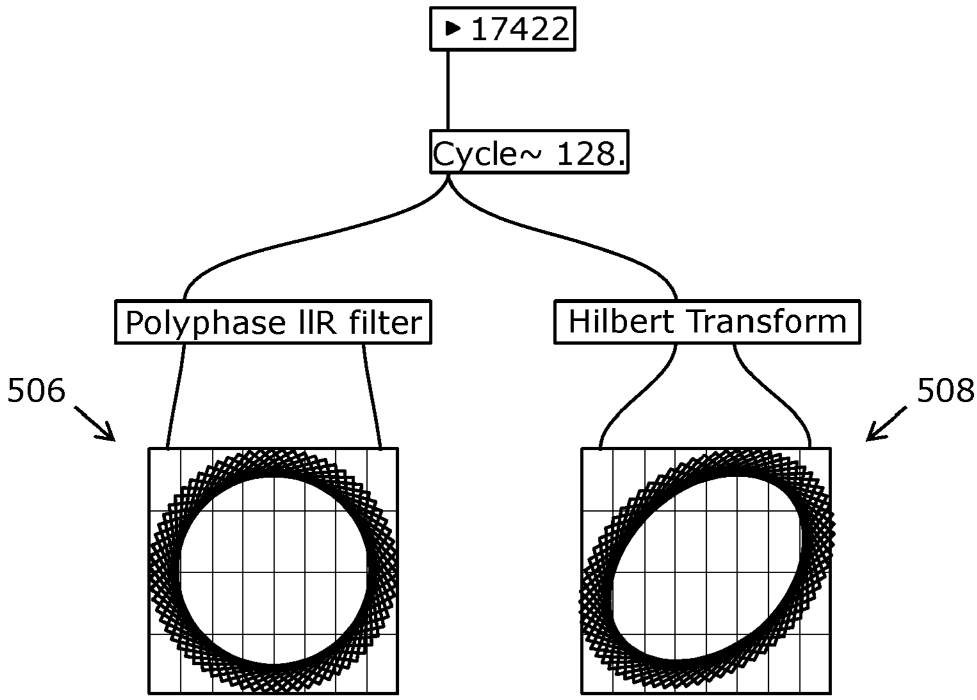
▶ 17422
Cycle~ 128.
Polyphase IIR filter
Hilbert Transform
506
508

FIG. 5B

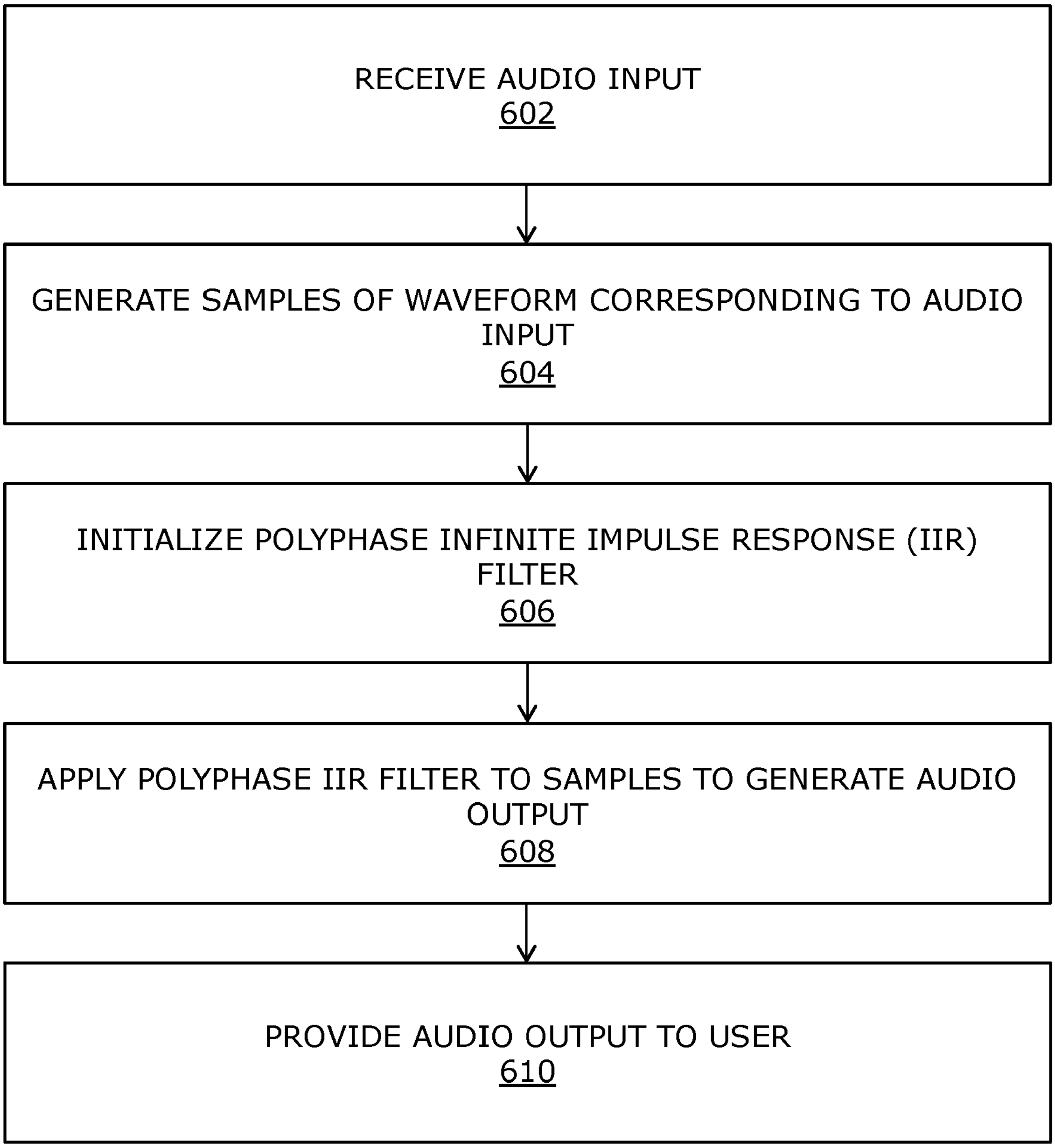
RECEIVE AUDIO INPUT
602
GENERATE SAMPLES OF WAVEFORM CORRESPONDING TO AUDIO INPUT
604
INITIALIZE POLYPHASE INFINITE IMPULSE RESPONSE (IIR) FILTER
606
APPLY POLYPHASE IIR FILTER TO SAMPLES TO GENERATE AUDIO OUTPUT
608
PROVIDE AUDIO OUTPUT TO USER
610

FIG. 6

METHOD AND DEVICE FOR PROCESSING AND PROVIDING AUDIO INFORMATION USING BI-PHASIC SEPARATION AND RE-INTEGRATION

TECHNICAL FIELD

The present disclosure relates generally to audio signal processing; and more specifically, to methods of processing and providing audio information to users. Moreover, the present disclosure also relates to devices for processing and providing audio information to users.

BACKGROUND

Audio signal processing is a continuously developing field, owing to its wide applications in daily life. Over the years, many ideas have been put forth in an ever-increasing attempt to provide realistic audio experiences to users. Notably, continuous advancements have been made to hardware and software of audio signal processing technologies to provide a good listening experience to users.

Typically, during a live performance (for example, a music concert, recording, and the like), a listener experiences audio signals from multiple audio sources that are often scattered in space across a venue of the live performance. Moreover, the listener also experiences reflections of the audio signals that bounce-off of reflective surfaces in the venue. These audio signals from the audio sources and the reflections of the audio signals hit the ears of the listener at different times (depending on distance travelled) and at different angles. These timing differences are known as phase. Such interaction of the listener with different phase components of the audio signals provide a holistic realistic experience for the listener. However, a recording of the audio signals that is recorded using a recording medium (such as a microphone) does not reproduce the same audio experience for the listener as the live performance. Typically, the loss of live experience occurs as phase components of the audio signals and their reflections become locked together in a specific relationship that they were in at time of recording. This causes sounds to lose their live feeling.

Formerly, monoaural sound was provided to listeners, but this had spatial shortcomings. Stereo was invented to address this issue and create better spatial effects for audio signals. Numerous attempts have been made to provide a more spatial or immersive audio experience, in addition to attempts to improve the quality of audio signals before presenting them to the listeners. However, spatial effects created by stereo are merely artificial simulations of real environments and are a psychoacoustic trick on the brain of the listeners that makes it seem as though natural factors are present.

Conventionally, artificial reverberation is used to create a perception of the real environment, whilst frequency equalization and/or dynamic range compression are used to create a perception of a difference in quality and amplitude increase for the audio signals. However, said mechanisms for the processing of audio signals are artificial and often negatively affect the frequency spectrum associated with the original signal. Moreover, current audio technologies often require specialized hardware for artificial processing of the audio signals in advance of being presented to the listener. Furthermore, audio encoding technologies (for example, such as MP3) further amplify problems associated with artificial processing of the audio signals as such technologies ignore one or more phase components altogether whilst encoding. Therefore, realism of the audio signals is lost forever.

Currently available solutions are fixes that modify spectral aspects of the audio signals with extremely subjective results. Moreover, these solutions are computationally intensive and often require considerable processing hardware for their implementation. Such considerable processing hardware is generally available in complex, non-portable devices such as workstations, desktop computers, and the like. Furthermore, these solutions provide high latency (for example, of the order of seconds or milliseconds) of processing owing to their complexity. As an example, use of bi-phasic separation of audio signals to provide different cognitive effects as opposed to presenting original audio signals to listeners is considerably limited, since the bi-phasic separation of audio signals is computationally complex. In particular, mathematical constructs for performing such bi-phasic separation of audio signals are quite complex, thereby requiring considerable processing resources. As the listeners generally listen to audio signals via portable devices (for example, such as headphones, speaker systems, and the like), portability of such currently available solutions is a major issue of concern as the portable devices have limited processing resources (which are a fraction of processing resources of non-portable devices). As an example, the portable devices generally have a few kilobytes of memory and few Megahertz processors, whereas the non-portable devices generally have several gigabytes of memory and several Gigahertz processors. Due to such limitations, the listeners are unable to have an immersive realistic audio experience when using the portable devices.

Therefore, in light of the foregoing discussion, there exists a need to overcome the aforementioned drawbacks associated with presentation of realistic, immersive audio signals to users of portable devices.

SUMMARY

The present disclosure seeks to provide a method of processing and providing audio information to a user. The present disclosure also seeks to provide a device for processing and providing audio information to a user. The present disclosure seeks to provide a solution to the existing problems of suboptimal audio experience when processing audio signals using conventional technologies, and of computational complexity and requirement of elaborate processing resources when processing audio signals using advanced emerging technologies. An aim of the present disclosure is to provide a solution that overcomes at least partially the problems encountered in prior art, and provides a method and a device that provide advanced audio signal processing functionality in a computationally simple manner with minimal processing resources.

In one aspect, an embodiment of the present disclosure provides a method of processing and providing audio information to a user, the method comprising:

- receiving at least one audio input corresponding to the audio information;
- generating samples of at least one waveform corresponding to the at least one audio input, wherein each sample of the at least one waveform has a real component and an imaginary component;
- initializing a polyphase infinite impulse response (IIR) filter with a set of one or more coefficients corresponding to a set of one or more stages of the polyphase IIR filter for both real components of the samples and imaginary components of the samples;
- applying the polyphase IIR filter to the samples to generate at least one audio output, the at least one audio output comprising processed renderings of both the real components of the samples and the imaginary components of the samples; and
- providing the at least one audio output representing the audio information to the user.

In another aspect, an embodiment of the present disclosure provides a device for processing and providing audio information to a user, the device comprising:

- a media handling unit configured to receive at least one audio input corresponding to the audio information from a media source;
- a media processing unit coupled to the media handling unit, the media processing unit being configured to:
  - generate samples of at least one waveform corresponding to the at least one audio input, each sample of the at least one waveform having a real component and an imaginary component;
  - initialize a polyphase infinite impulse response (IIR) filter with a set of one or more coefficients corresponding to a set of one or more stages of the polyphase IIR filter for both real components of samples and the imaginary components of the samples; and
  - apply the polyphase IIR filter to the samples to generate at least one audio output, the at least one audio output comprising processed renderings of both the real components of the samples and the imaginary components of the samples; and
- at least one speaker coupled to the media processing unit, wherein the at least one speaker is configured to provide the at least one audio output representing the audio information to the user.

Embodiments of the present disclosure substantially eliminate or at least partially address the aforementioned problems in the prior art, and enable advanced audio signal processing in a computationally efficient manner using minimal processing resources to provide an immersive and realistic audio listening experience.

Additional aspects, advantages, features and objects of the present disclosure would be made apparent from the drawings and the detailed description of the illustrative embodiments construed in conjunction with the appended claims that follow.

It will be appreciated that features of the present disclosure are susceptible to being combined in various combinations without departing from the scope of the present disclosure as defined by the appended claims.

BRIEF DESCRIPTION OF THE DRAWINGS

The summary above, as well as the following detailed description of illustrative embodiments, is better understood when read in conjunction with the appended drawings. For the purpose of illustrating the present disclosure, exemplary constructions of the disclosure are shown in the drawings. However, the present disclosure is not limited to specific methods and instrumentalities disclosed herein. Moreover, those skilled in the art will understand that the drawings are not to scale. Wherever possible, like elements have been indicated by identical numbers.

Embodiments of the present disclosure will now be described, by way of example only, with reference to the following diagrams wherein:

FIG. 1A illustrates an architecture of a device for processing and providing audio information to a user, in accordance with an embodiment of the present disclosure;

FIG. 1B illustrates an example implementation of the device of FIG. 1A, in accordance with an embodiment of the present disclosure;

FIG. 2 illustrates an environment wherein a device for processing and providing audio information to a user is used, in accordance with an embodiment of the present disclosure;

FIG. 3 illustrates an exemplary node of a polyphase IIR filter, in accordance with an embodiment of the present disclosure;

FIG. 4 illustrates an exemplary implementation of a polyphase IIR filter, in accordance with an embodiment of the present disclosure;

FIG. 5A illustrates representations of phase responses of a low-frequency sine wave input for a polyphase IIR filter and a Hilbert transform, while FIG. 5B illustrates representations of phase responses of a high-frequency sine wave input for the polyphase IIR filter and the Hilbert transform, in accordance with an embodiment of the present disclosure; and

FIG. 6 illustrates steps of a method of processing and providing audio information to a user, in accordance with an embodiment of the present disclosure.

In the accompanying drawings, an underlined number is employed to represent an item over which the underlined number is positioned or an item to which the underlined number is adjacent. A non-underlined number relates to an item identified by a line linking the non-underlined number to the item. When a number is non-underlined and accompanied by an associated arrow, the non-underlined number is used to identify a general item at which the arrow is pointing.

DETAILED DESCRIPTION OF EMBODIMENTS

The following detailed description illustrates embodiments of the present disclosure and ways in which they can be implemented. Although some modes of carrying out the present disclosure have been disclosed, those skilled in the art would recognize that other embodiments for carrying out or practising the present disclosure are also possible.

In one aspect, an embodiment of the present disclosure provides a method of processing and providing audio information to a user, the method comprising:

- receiving at least one audio input corresponding to the audio information;
- generating samples of at least one waveform corresponding to the at least one audio input, wherein each sample of the at least one waveform has a real component and an imaginary component;
- initializing a polyphase infinite impulse response (IIR) filter with a set of one or more coefficients corresponding to a set of one or more stages of the polyphase IIR filter for both real components of the samples and imaginary components of the samples;
- applying the polyphase IIR filter to the samples to generate at least one audio output, the at least one audio output comprising processed renderings of both the real components of the samples and the imaginary components of the samples; and
- providing the at least one audio output representing the audio information to the user.

In another aspect, an embodiment of the present disclosure provides a device for processing and providing audio information to a user, the device comprising:

- a media handling unit configured to receive at least one audio input corresponding to the audio information from a media source;
- a media processing unit coupled to the media handling unit, the media processing unit being configured to:
    - generate samples of at least one waveform corresponding to the at least one audio input, each sample of the at least one waveform having a real component and an imaginary component;
    - initialize a polyphase infinite impulse response (IIR) filter with a set of one or more coefficients corresponding to a set of one or more stages of the polyphase

IIR filter for both real components of the samples and imaginary components of the samples; and

- apply the polyphase IIR filter to the samples to generate at least one audio output, the at least one audio output comprising processed renderings of both the real components of the samples and the imaginary components of the samples; and

- at least one speaker coupled to the media processing unit, wherein the at least one speaker is configured to provide the at least one audio output representing the audio information to the user.

The method described herein adopts a processing efficient methodology of audio signal processing and phase transformation. The inventors have realized that employing a polyphase IIR filter in the method enables elimination of redundant data from calculations thereby considerably simplifying data processing. In other words, data processing is simplified due to the innovative measure of using the polyphase IIR filter. The polyphase IIR filter is a mathematical construct that is effectively used for bi-phasic separation of audio signals since it requires a very small number of multipliers to be implemented, and is inherently stable. Moreover, as the method is computationally efficient, it provides extremely low latency in embedded systems. As an example, the latency provided by the method is of the order of microseconds, as compared to conventional latency that lies in the order of milliseconds or seconds. In other words, the method and device described herein provide real-time or near-real time audio processing capabilities. The computational efficiency and low latency of the described method hugely increase portability of the method, thereby enabling the method to be efficiently implemented even in portable devices that have limited computational resources. As an example, the method can be implemented even in portable devices having a few kilobytes of memory and few Megahertz processors, whilst still providing the low latency. At the aforesaid device, bi-phasic separation of audio signals is efficiently implemented with minimal processing resources and latency, via the polyphase IIR filter. When the at least one audio output is presented to a user, neurological action of perception causes the processed renderings of both the real components of the samples and the imaginary components of the samples to be reintegrated (i.e. "reassembled") into a singular stimulus via the cognitive activity taking place within the user's brain. The device described herein can be easily used by users to have an immersive realistic audio experience.

It will be appreciated that the aforesaid method of processing and providing audio information to the user is implemented on the aforesaid device for processing and providing audio information to the user.

In an embodiment, the device is a portable device. In one example, the device is a device to be worn on a user's head, attached to the user's body and/or carried by the user in hand or in a pocket, thus being a user portable device. Examples of such portable devices include, but are not limited to, headphones, earphones, electronic earbuds, headsets, portable speakers, personal media players, and portable virtual assistant devices. As a specific example, the device (for processing and providing audio information to the user) is a headphone, earphone or a headset. It will be appreciated that when the device is such a user portable device, the device has very limited computational resources.

Optionally, an audio processing chain is implemented on the device as a series of modules of a System-On-Chip (SoC). In other words, internal components (namely, internal modules) of the device are optionally implemented on a single chip. The modules of the SoC generally consume much less power and occupy much lesser space as compared to multi-chip designs with equivalent functionality. General-purpose audio processing chips that can be used to provide same functionality as the internal modules of the device lack required precision and fidelity of audio response for an optimal audio listening experience. Using the SoC on the device increases portability of the audio processing technique described herein, thereby enabling the method of processing and providing audio information to be implemented effectively on both portable devices, as well as non-portable devices. As a specific example, the device is a headphone, earphone or headset. In such a case, using the SoC on the device enables the device to be designed ergonomically. Ergonomically-designed devices can be comfortably operated by users.

In another embodiment, the device is a portable computing device having substantial computational resources. Such portable computing devices are different from the portable devices described earlier, with respect to their computational resources, the portable devices described earlier basically being portable audio devices. Generally, portable computing devices are primarily designed for computational tasks that take up considerable computational resources and therefore have substantial computing resources, whereas the earlier mentioned portable devices are not designed in such a manner. Examples of the portable computing devices include, but are not limited to, a smartphone, a phablet computer, a tablet computer, a laptop computer, and a personal digital assistant.

In yet another embodiment, the device is a non-portable device. Examples of such non-portable devices include, but are not limited to, desktop computers, workstations, non-portable speakers, and non-portable virtual assistant devices.

Throughout the present disclosure, the term "*audio information*" refers to an audio signal (namely, a sound signal). The audio information may simply be understood to be "*sound information*" or "*auditory information*". It will be appreciated that the audio information pertaining to a given audio signal encompasses one or more of: the given audio signal, a waveform of the given audio signal, data associated with a file of the given audio signal. The audio information that is provided to the user may be, for example, associated with a song, a medley of songs, speech, an audiobook, an audio tutorial, an audio podcast, environmental sounds, atonal audio, arhythmic audio (for example, audio used to promote states of consciousness such as meditation, relaxation and sleep), and the like.

The method and the device described herein enable provision of the audio information to the user. The user can therefore, be understood to be a consumer of said audio information. The user is a person who operates the device to receive the audio information, wherein the method of processing and providing the audio information is performed (namely, implemented) at the device to provide the audio information to the user.

In an embodiment, the device is configured to provide the audio information to a single user. In another embodiment, the device is configured to provide the audio information to a plurality of users. Optionally, in this regard, the plurality of users are provided the audio information simultaneously.

Throughout the present disclosure, the term "*device*" refers to an electronic device that is capable of performing tasks pertaining to the aforementioned method. Furthermore, the device is intended to be broadly interpreted to include any electronic device that may be used by the user for receiving the audio information. Notably, the device comprises several specialized components that are configured to perform specific tasks pertaining to processing of the audio information and provision of the audio information to the user. In particular, the device enables provision of the at least one audio output to the user.

Next, there will be described steps of the method of processing and providing the audio information to the user.

The method comprises receiving, at the media handling unit, at least one audio input from the media source. The term "*audio input*" refers to an audio signal that serves as an input for the device. It will be appreciated that the term "*at least one audio input*" refers to "*one audio input*" in some implementations, and "*a plurality of audio inputs*" in other implementations. When a single audio input is used, such as an audio channel, as the input for the device, the single audio input may be understood to be a "*monoaural audio input*". When multiple audio inputs (namely, the plurality of audio inputs), such as multiple audio channels, are used as the inputs for the device, the multiple audio inputs may be understood to be "*stereo audio inputs*".

The "*media handling unit*" is implemented as hardware, software, firmware, or a combination of these. The media handling unit is communicably coupled to the media source. In particular, the media handling unit is communicably coupled to the media source wirelessly and/or in a wired manner. It will be appreciated that the coupling between the media handling unit and the media source enable the media handling unit to receive the at least one audio input from the media source.

In an embodiment, the media handling unit is directly coupled to the media source. In another embodiment, the media handling unit is coupled to the media source via a communication network. Examples of the communication network include, but are not limited to, a radio network (for example, such as a Bluetooth®-based network, a WiFi network, a cellular network, and the like), Internet, an Ethernet-based local area network, and an Ethernet-based personal area network.

The "*media source*" refers to a source of the at least one audio input. In an embodiment, the at least one audio input is generated by the media source. In one alternative or additional embodiment, the at least one audio input is stored at the media source. In such a case, the at least one audio input may be generated elsewhere and is simply stored at the media source. In another case, the audio input is generated by the media source and stored at the media source for later retrieval.

Examples of audio generation devices possibly connected to the media source for generating audio include, but are not limited to, a musical instrument (comprising a pickup or other audio recording device), a microphone and a computing device. In an embodiment where the audio generating device is a computing device, the computing device may comprise the media source. In an embodiment, the audio generating device may be a computing device whereat a software application executes to generate the at least one audio input. The software application may, for example, be a wellness-oriented software application that generates atonal, arhythmic audio effects such as sounds of wind, rain, whale song, ambient sound scapes, and the like).

Optionally, the media source is at least one of: an internal memory of the device, a memory of an external device, a data repository, a cloud-based memory.

In an embodiment, when the media source is the internal memory of the device, the media handling unit is coupled to the internal memory of the device in a wired manner. As an example, the internal memory of the device may be an 8 gigabytes random access memory (RAM).

In another embodiment, when the media source is the memory of the external device, the media handling unit is coupled to said memory either wirelessly or in a wired manner. In some implementations, the external device is a portable device (for example, such as a mobile phone, a laptop computer, a tablet computer, a portable audio player (such as a Walkman®, an iPod®, and the like), and the like), whereas in other implementations, the external device is a non-portable device (for example, such as a desktop computer, a workstation, a mixing console, a Digital Audio Workstation (DAW), a channel insert, and the like). Optionally, the device is coupled to the external device in a manner that communicable coupling is established between the media handling unit of the device and the memory of the external device. As an example, the device may be implemented as a headphone, earphone or headset whereas the external device may be implemented as a smartphone. In such an example, the smartphone may include a receptacle for receiving an audio jack of the headphone earphone or headset to establish coupling therebetween. The at least one audio input may, for example, be songs stored on the memory of the smartphone.

In yet another embodiment, when the media source is the data repository, the media handling unit is coupled to the data repository either wirelessly or in a wired manner. The data repository may, for example, be implemented as a database arrangement.

In still another embodiment, when the media source is the cloud-based memory, the media handling unit is wirelessly coupled to the cloud-based memory. As an example, the media handling unit may be coupled to the cloud-based memory of an audio streaming service, via the Internet.

In yet another embodiment, when the media source is or connected to the microphone (or other audio generating device), the media handling unit is coupled to the microphone either wirelessly or in a wired manner. Optionally, the device further comprises the microphone. In other words, the microphone is a part of the device. Alternatively, optionally, the microphone is an external microphone that is coupled to the media handling unit. As an example, the media source may be a microphone that is used to record a song played at a live concert, the microphone being wirelessly coupled (for example, via a Bluetooth®-based network) to the media handling unit. In such an example, the song is an audio input.

The method comprises generating, at the media processing unit, samples of the at least one waveform corresponding to the at least one audio input. In other words, the media processing unit is configured to generate the samples of the at least one waveform. Throughout the present disclosure, the term "*media processing unit*" refers to hardware, software, firmware, or a combination of these that is configured to perform specialized audio signal processing steps on the at least one audio input to eventually generate the at least one audio output. In simple terms, the media processing unit is a computational element having processing capabilities or is a set of instructions implemented in conjunction with one or more processors of the device. As an example, the media processing unit may be implemented by way of a digital signal processor (DSP), which is a specialized chip that is optimized for operational needs of digital signal processing.

The media processing unit is coupled to the media handling unit either wirelessly and/or in a wired manner. By way of said coupling, the received at least one audio input is obtained by the media processing unit from the media handling unit, for further processing.

A "*waveform*" corresponding to an "*audio input*" refers to a graphical representation of the audio input, wherein the graphical representation is a function of time. In other words, the at least one waveform is a continuous-time signal representative of the at least one audio input. Moreover, a "*sample*" of a given waveform is a value of the given waveform at a given point in time. In other words, the sample of the given waveform is an amplitude of the given waveform at the given point in time. It will be appreciated that the at least one audio input is received in an analog form. By sampling the at least one waveform corresponding to the at least one audio input, the at least one audio input is digitized (namely, converted into a digital form). Then, the at least one audio input (which is digitized) can be efficiently processed to produce the at least one audio output. In particular, multiple samples of the given waveform are generated to accurately represent the given waveform as a discrete-time signal.

In an embodiment, when generating the samples of the at least one waveform, the media processing unit is configured to acquire a value of the at least one waveform, after every 'T' seconds. Herein, the 'T' seconds form a sampling interval at which the at least one waveform is sampled. In an embodiment, a sampling rate of the at least one waveform is equal to 1/T samples per second, wherein 'T' is the sampling interval. The "*sampling rate*" of the at least one waveform may also be understood to be a "*sampling frequency*" of the at least one waveform.

In an embodiment, when generating the samples of the at least one waveform, the media processing unit is configured to select a value of 'T' in a manner that the sampling rate is equal to or greater than twice of a maximum frequency component of the at least one audio input. In other words, when generating the samples of the at least one waveform, the media processing unit selects the value of 'T' so as to satisfy Nyquist sampling theorem. It will be appreciated that when the Nyquist sampling theorem is satisfied at the time of sampling, all requisite information from the at least one audio input can be accurately captured within the generated samples.

As an example, the samples of the at least one waveform corresponding to the at least one audio input may be generated at a sampling rate of 60 samples per second, when the maximum frequency component of the at least one audio input is equal to 30 Hertz. In such a case, the sampling interval T may be equal to $0.01\overline{6}$ second.

Notably, each sample of the at least one waveform has a real component and an imaginary component. Therefore, when the at least one media processing unit generates the samples of the at least one waveform, the media processing unit performs not only sampling of the at least one waveform, but also performs bi-phasic separation of the samples of the at least one waveform via the polyphase IIR filter. In other words, generating the samples of the at least one waveform involves sampling the at least one waveform at a required sampling rate to obtain raw samples, and performing bi-phasic separation (via the polyphase IIR filter) of each raw sample of the at least one waveform to yield the real component and the imaginary component of each sample.

In an embodiment, the real component of a given sample is a cosine value of the given sample, whereas the imaginary component of the given sample is a sine value of the given sample. The real component and the imaginary component are two different phase dimensions of the given sample. It will be appreciated that the imaginary component of the given sample is complementary to the real component of the given sample.

In an embodiment, the media processing unit is configured to employ the polyphase IIR filter to decompose a given sample into its real component and imaginary component. The separation of the given sample into its real component and imaginary component is performed exclusively within the polyphase IIR filter. Notably, raw (namely, unprocessed) samples of the at least one waveform are inputs to the polyphase IIR filter. The polyphase IIR filter separates these raw samples into their real and imaginary components. For a stream of samples of the at least one waveform, the polyphase IIR filter generates two streams - a stream of real components of the samples, a stream of imaginary components of the samples. The samples of the at least one waveform are complex samples that are separated into their constituent real components and imaginary components, by the polyphase IIR filter.

It will be appreciated that an audio signal such as a given audio input has both time and spectral components. In the given audio input, the time component and the spectral component are locked together into a complex waveform associated with the given audio input. When generating the samples of the at least one waveform, the media processing unit unlocks the real component and the imaginary component of each sample. Said unlocking of the real component and the imaginary component of each sample of the at least one waveform is known as "*bi-phasic separation of the samples of the at least one waveform*". According to embodiments of the teachings herein, bi-phasic separation is performed by the polyphase IIR filter.

In an embodiment, the method further comprises placing the samples of the at least one waveform into an input buffer (namely, an input array). In an embodiment, the media processing unit is configured to place the samples of the at least one waveform into the input buffer, upon generating the samples. It will be appreciated that the input buffer provides a mechanism through which the media processing unit can systematically fetch the samples of the at least one waveform for further processing, as required. In an embodiment, separate input buffers are employed for the real components of the samples and the imaginary components of the samples.

Next, the method comprises initializing, via the media processing unit, the polyphase IIR filter with the set of one or more coefficients corresponding to the set of one or more stages of the polyphase IIR filter for both the real components of the samples and the imaginary components of the samples. The one or more coefficients are applied to the polyphase IIR filter as constant values. The polyphase IIR filter is implemented on the media processing unit of the device. Pursuant to embodiments of the present disclosure, the polyphase IIR filter is implemented to perform transformation operations on the samples of the at least one waveform. In particular, said transformation operations pertain to bi-phasic separation of the samples into their real components and their imaginary components.

Notably, in the present method, the polyphase IIR filter is optimized to perform operations similar to that of bi-phasic separation of waveform(s) using a Hilbert transform or a biquad filter. The polyphase IIR filter performs bi-phasic separation using a much more concise methodology of signal filtering and phase transformation, as compared to both the Hilbert transform and the biquad filter.

For sake of reference only, there will now be described briefly, bi-phasic separation for a given waveform using both the Hilbert transform and the biquad filter.

The Hilbert transform can be used to perform bi-phasic separation of samples of a given waveform into corresponding real components and imaginary components. A signal u(t) corresponding to the given waveform is convoluted with a function h(t)= 1/ (ϖt) to generate a Hilbert transform of the signal u(t). Mathematically, the Hilbert transform for the signal u(t) is given by:

$$\mathrm{H(u)(t)} = \frac{1}{\pi}\,\mathrm{p.v.}\int_{-\infty}^{+\infty} \frac{\mathrm{u}(\tau)}{\mathrm{t}-\tau}\,\mathrm{d}\tau$$

wherein u(t) is the signal, p.v. is Cauchy principal value, and τ is an instant of time such that the Hilbert transform is not possible as an ordinary improper integral at τ = t.

In practice, the Hilbert transform generates the signal H (u)(t) for the signal u(t) by shifting phase (namely, phase transformation) of the signal u(t) by $\pm \varpi/2$. As an example, for a sine input signal, a phase transformed signal for the input signal may be a cosine signal, wherein the cosine signal forms a real component for the input signal, while the sine signal forms an imaginary component for the input signal. Applying a Hilbert transform on samples of the given waveform enables identification of a real component and an imaginary component for each sample of the waveform.

The biquad filter can also be used to perform bi-phasic separation of samples of a given waveform. In such a case, the biquad filter is constructed as two pairs of two-pole, two-zero biquad filters in a cascaded series. A typical biquad filter uses a two-sample feedback mechanism. These two samples are used immediately and are also stored to be used again later. The two stored samples comprise historical input samples X_{-1} and X_{-2} and corresponding historical output samples Y_{-1} and Y_{-2}. The input sample X_{-2} and the output sample Y_{-2} are stored in a first input node and a first output node, respectively. Likewise, the input sample X_{-1} and the output sample Y_{-1} are stored in a second input node and a second output node, respectively. For a newly arrived input sample X0, the biquad filter produces output sample Y0 which is stored in a third input node and a third output node, to be used again. The output sample Y0 is produced based on the input samples (X0, X_{-1}, X_{-2}), the output samples (Y_{-1}, Y_{-2}) and a set of one or more coefficients (a0, a1, a2, b1 and b2) corresponding to one or more stages of the biquad filter. Mathematically, the generation of the output sample Y0 using the set of one or more coefficients is represented as:

$$Y[n] = (a0*x[n])+(a1*x[n-1])+(a2*x[n-2])+(b1*y[n-1])+(b2*y[n-2]);$$

$$\text{Hence, } Y0 = (a0*X0) + (a1*X_{-1}) + (a2*X_{-2}) + (b1*Y_{-1}) + (b2*Y_{-2});$$

wherein a0, a1 and a2 are coefficients for the biquad filter for the input samples X0, X_{-1} and X_{-2}, respectively; and b1 and b2 are coefficients for the biquad filter for the output samples Y_{-1} and Y_{-2}, respectively.

When the samples of the given waveform are provided as input samples to the biquad filter, the biquad filter performs transforms on each of the samples to achieve bi-phasic separation of each of the samples into a real component and an imaginary component. The biquad filter is initialized with the aforesaid set of 5 coefficients (namely, a0, a1, a2, b1 and b2) corresponding to each stage (for example, four stages) for the real components of the samples and the imaginary components of the samples. More specifically, the set of 5 coefficients for the real component is different from the set of 5 coefficients for the imaginary component. The biquad filter initialized with the set of 5 coefficients corresponding to the set of one or more stages for imaginary component shifts an input sample by 90 degrees. Furthermore, the biquad filter initialized with the set of 5 coefficients corresponding to the set of one or more stages for real component offsets the input sample in time to preserve phase integrity, for example, between different channels from where an output sample will be generated.

There will now be discussed implementation details of the polyphase IIR filter.

In an embodiment, the polyphase IIR filter is implemented as a four-stage filter. Optionally, in this regard, the four stages of the polyphase IIR filter are serially cascaded one after the other. An output of a given stage serves as an input of a next stage. In an embodiment, at each stage of the four-stage filter, separate nodes are employed for processing real and imaginary components of a given sample. Therefore, optionally, a four-stage polyphase IIR filter comprises eight nodes, wherein a first set of four nodes and a second set of four nodes are arranged parallelly for processing the real components and the imaginary components of the given sample, respectively, and wherein nodes of a given set are serially cascaded one after the other. One such exemplary node has been elucidated in conjunction with FIG. 3. Further, one such exemplary four-stage implementation of the polyphase IIR filter has been elucidated in conjunction with FIG. 4.

In the polyphase IIR filter, historical samples are used to provide a feedback loop. In an embodiment, at each stage, the polyphase IIR filter employs two historical input samples and two historical output samples. Optionally, in this regard, the media processing unit is configured to store the two historical input samples and two historical output samples in memory elements. Said memory elements are associated with at least one of: the polyphase IIR filter, the media processing unit. It will be appreciated that the two historical input samples and two historical output samples are employed in the polyphase IIR filter to mimic a two-sample delay that is inherent in a typical biquad filter.

For illustration purposes only, let us consider that the two historical input samples are represented as X_{-1} and X_{-2}, whereas the two historical output samples are represented as Y_{-}

$_1$ and Y_{-2}. For a new input sample X0, the polyphase IIR filter produces a new output sample Y0 by processing real and imaginary components of the input sample X0 through the one or more stages of the polyphase IIR filter, wherein at each stage of the polyphase IIR filter, a given sample is multiplied with its corresponding coefficient. In other words, the output sample Y0 is produced based on the input samples (X0, X_{-1}, X_{-2}), the output samples (Y_{-1}, Y_{-2}) and a set of one or more coefficients (a0, a1, a2, b1 and b2) corresponding to one or more stages (optionally, four stages) of the polyphase IIR filter.

Mathematically, the output sample Y0 is generated as:

$$Y[n] = (a0*x[n])+(a1*x[n-1])+(a2*x[n-2])+(b1*y[n-1])+(b2*y[n-2])$$

$$\text{Hence, } Y0 = (a0*X0) + (a1*X_{-1}) + (a2*X_{-2}) + (b1*Y_{-1}) + (b2*Y_{-2})$$

wherein a0, a1 and a2 are coefficients for the polyphase IIR filter for the input samples X0, X_{-1} and X_{-2}, respectively; and b1 and b2 are coefficients for the polyphase IIR filter for the output samples Y_{-1} and Y_{-2}, respectively.

It will be appreciated that the set of one or more coefficients corresponding to the set of one or more stages of the polyphase IIR filter for both the real components of the samples and the imaginary components of the samples are selected in a manner that computations of the polyphase IIR filter are considerably reduced in comparison to similar computations for the Biquad filter. When the polyphase IIR filter is initialized, the set of one or more coefficients are applied to the one or more stages of the polyphase IIR filter as constant values for both the real components of the samples and the imaginary components of the samples.

In an embodiment, when the polyphase IIR filter is implemented as the four-stage filter, the polyphase IIR filter is initialized separately with a set of 5 coefficients for the real components of the samples and a set of 5 coefficients for the imaginary components of the samples.

In an embodiment, the polyphase IIR filter is initialized with the set of 5 coefficients (notably, a0, a1, a2, b1, and b2) corresponding to the real components of the samples in the four-stage implementation of the polyphase IIR filter as presented in Table 1. Preferably, values of the set of 5 coefficients are in the range of a minimum value to a maximum value for the four stages as presented in Table 1.

Table 1

Stage	a0		a1		a2		b1		b2	
	Min	Max	Min	Max	Min	Max	Min	Max	Min	Max
Stage 1	0.13587672	0.18763928	0	0	-1.16	-0.84	0	0	-0.18763928	-0.13587672
Stage 2	0.61574436	0.85031364	0	0	-1.16	-0.84	0	0	-0.85031364	-0.61574436
Stage 3	0.794094	1.096606	0	0	-1.16	-0.84	0	0	-1.096606	-0.794094
Stage 4	0.83210232	1.14909368	0	0	-1.16	-0.84	0	0	-1.14909368	-0.83210232

In a specific embodiment, the polyphase IIR filter may be initialized with the set of 5 coefficients corresponding to the real components of the samples in the four-stage implementation of the polyphase IIR filter as presented in Table 1A.

Table 1A

Stage	a0	a1	a2	b1	b2
Stage 1	0.161758	0	-1	0	-0.161758
Stage 2	0.733029	0	-1	0	-0.733029
Stage 3	0.94535	0	-1	0	-0.94535
Stage 4	0.990598	0	-1	0	-0.990598

In an embodiment, the polyphase IIR filter is initialized with the set of 5 coefficients (notably, a0, a1, a2, b1, and b2) corresponding to the imaginary components of the samples in the four-stage implementation of the polyphase IIR filter as presented in Table 2. Preferably, values of the set of 5 coefficients are in the range of a minimum value to a maximum value for the four stages as presented in Table 2.

Table 2

Stage	a0		a1		a2		b1		b2	
	Min	Max	Min	Max	Min	Max	Min	Max	Min	Max
Stage 1	0.40269684	0.55610516	0	0	-1.16	-0.84	0	0	-0.55610516	-0.40269684
Stage 2	0.73602312	1.01641288	0	0	-1.16	-0.84	0	0	-1.01641288	-0.73602312
Stage 3	0.82034316	1.13285484	0	0	-1.16	-0.84	0	0	-1.13285484	-0.82034316

Stage 4	0.8379	1.1571	0	0	-1.16	-0.84	0	0	-1.1571	-0.8379

In a specific embodiment, the polyphase IIR filter may be initialized with the set of 5 coefficients corresponding to the imaginary components of the samples in the four-stage implementation of the polyphase IIR filter as presented in Table 2A.

Table 2A

Stage	a0	a1	a2	b1	b2
Stage 1	0.479401	0	-1	0	-0.479401
Stage 2	0.876218	0	-1	0	-0.876218
Stage 3	0.976599	0	-1	0	-0.976599
Stage 4	0.9975	0	-1	0	-0.9975

In an embodiment, when the polyphase IIR filter is initialized with the set of 5 coefficients presented in Table 1 (and more optionally, Table 1A) and the set of 5 coefficients corresponding presented in Table 2 (and more optionally, Table 2A) in the four-stage implementation of the polyphase IIR filter, an optimized mathematical equation for generating the output sample Y[n] is:

$$Y[n] = (a0*x[n]) + (a2*x[n-2]) - (a0*y[n-2])$$

Hence, in such a case, $Y0 = (a0*X0) + (a2*X_{-2}) - (a0*Y_{-2})$

In other words, $Y[n] = (a0*x[n]) + (a2*x[n-2]) + (b2*y[n-2])$, which can be further reduced based upon commutative and associative properties yielding that $Y[n]=a0*(x[n]-y[n-2])-x[n-2]$, when $a2=-1$

It will be appreciated that the polyphase IIR filter described herein, when optionally initialized with the coefficients specified in Tables 1 and 2 corresponding to the one or more stages of the polyphase IIR filter for both the real and imaginary components of the samples, enables reduction of 2/5ths of calculations as compared to a similar implementation of the biquad filter with the same coefficients. In other words, when the polyphase IIR filter is optionally initialized with the coefficients specified in Tables 1 and 2 (and more optionally, initialized with the coefficients specified in Tables 1A and 2A), the polyphase IIR filter eliminates 40 percent of redundancies as compared to a similar implementation of the biquad filter with the same coefficients.

In an embodiment, each stage of the polyphase IIR filter employs the feedback mechanism to eliminate redundancies. Optionally, in this regard, the feedback mechanism is implemented by way of double use feedback history loops. In an embodiment, the feedback history loops loop through the new input sample, the two historical input samples, and the two historical output samples in a manner that redundancies in the polyphase IIR filter are effectively eliminated. Said feedback history loops enable elimination of zero-sum multiplications at each stage of the polyphase IIR filter, thereby enabling the polyphase IIR filter to cut half of historical sample delays from each stage. It will be appreciated that zero-sum multiplications (namely, a multiplication where a product of a given sample component and its coefficient equals zero, said product being a constituent summing element of an overall sum) are redundancies in any given filter. By eliminating the zero-sum multiplications, the polyphase IIR filter provides a technical effect of considerably simplified processing over typical biquad filters which are unable to remove such multiplications. In particular, the feedback mechanism of a given node of the polyphase IIR filter loops through memory elements of the given node in a manner that the number of mathematical operations (for example, multiplications, additions, subtractions, and the like) in the computation of output samples are reduced to a maximum possible extent, whilst achieving the same transformation effects as the typical biquad filters. Effectively, the polyphase IIR filter reduces an overall number of historical sample delays from each stage to half their maximum number in similar typical biquad filters.

Moreover, it will be appreciated that as the redundancies in the polyphase IIR filter are eliminated, the processing latency of the device (and particularly, of the media processing unit) is reduced. With a decrease in the number of required computations, the latency between receiving the at least one audio input and generating the at least one audio output is considerably reduced. Optionally, said latency is of the order of microseconds. Therefore, processing of the at least one audio input to yield the at least one audio output occurs in real time or near-real time. Beneficially, bi-phasic separation of audio signals using the polyphase IIR filter enables even portable devices having limited processing capabilities to provide improved immersive cognitive effects to users of such devices. Simplification of complex computations and reduction of latency, as described above, hugely increase portability of bi-phasic separation and reassembly of audio signals.

In an experimental observation, it was verified that using the set of the one or more coefficients provided in Table 1 and Table 2 (and more optionally, provided in Tables 1A and 2A) corresponding to the set of one or more stages of the polyphase IIR filter, enable the polyphase IIR filter to accurately represent phase dynamics similar to those of the Hilbert transform. This result was verified by comparing two phase responses of a sine wave input for both the polyphase IIR filter and the Hilbert transform using a digital oscilloscope, across a range of frequencies. At a given frequency, a first phase response is generated by employing the polyphase IIR filter and a second phase response is generated by employing the Hilbert transform.

At a low frequency, the first and second phase responses for the sine wave input are circular in shape for both the polyphase IIR filter and the Hilbert transform, respectively, indicating that a phase transform of the sine wave input is uniformly 90 degrees for both the polyphase IIR filter and the Hilbert transform. Furthermore, it was observed that the circular shape of the second phase response begins to distort at higher frequencies, thereby indicating that at the higher frequencies, the phase transformation of the sine wave input is not 90 degrees uniformly using the Hilbert transform. However, the first phase response generated by employing the polyphase IIR filter initialized with the set of one or more coefficients corresponding to the set of one or more stages retains its pure circular shape on the digital oscilloscope even at the higher frequencies (and even at the Nyquist frequency) for the sine wave input. This indicates an improvement in cognitive effects and audio quality when processing audio inputs of very high frequencies using the method described herein. The comparison between the first phase response and the second phase response across a frequency spectrum indicates an improvement in quality of the phase transformation of the sine wave input using the polyphase IIR filter over using the Hilbert transform.

Beneficially, the aforementioned set of one or more coefficients corresponding to the set of one or more stages of the polyphase IIR filter for the real components and the imaginary components of the samples of the at least one waveform enable swift transformation of said samples across a very broad frequency range. Moreover, optionally, the polyphase IIR filter when operated based on the set of one or more coefficients enables the bi-phasic display of an oscilloscope to represent the samples of the at least one waveform as a pure circular form (namely, shape) for any frequency of the samples of the at least one waveform lying a range of at least 0 hertz to Nyquist frequency. In other words, the polyphase IIR filter when operated based on the set of one or more coefficients performs phase transformation on the samples of the at least one waveform.

It will be appreciated that when the polyphase IIR filter is initialized in the aforesaid manner, the polyphase IIR filter is optimized to enable elimination of redundant data from calculations. This, in turn, considerably simplifies data processing at the media processing unit. As the polyphase IIR filter enables computationally efficient data processing, there is reduced consumption of processing power and energy at the device during implementation of the method.

The method comprises applying the polyphase IIR filter to the samples to generate at least one audio output, the at least one audio output comprising a processed rendering of both the real components of the samples and the imaginary components of the samples. In particular, the media processing unit is configured to apply the polyphase IIR filter to the samples to generate the at least one audio output.

It will be appreciated that the term "*at least one audio output*" refers to "*one audio output*" in some implementations, and "*a plurality of audio outputs*" in other implementations. As an example, a single audio output may be generated using the polyphase IIR filter, the single audio output comprising 2 constituents - a processed rendering of the real components of the samples, and a processed rendering of the real components of the samples.

In an embodiment, when applying the polyphase IIR filter to the samples, the media processing unit is configured to:

- fetch a required number of samples from the input buffer and/or memory elements;
- transform the samples through the one or more stages of the polyphase IIR filter, by applying a corresponding set of one or more coefficients to the samples at each stage of the polyphase IIR filter; and
- obtain output samples from the polyphase IIR filter to generate the at least one audio output, wherein real components of the output samples form the processed rendering of the real components of the samples and imaginary components of the output samples form the processed rendering of the imaginary components of the samples.

Optionally, in this regard, the output samples are generated by the polyphase IIR filter as a result of transformation operations performed on the samples as the samples are processed through the one or more stages of the polyphase IIR filter.

In an embodiment, transformation of the samples to which the polyphase IIR filter is applied is performed over the one or more stages of the polyphase IIR filter. In this regard, a transformed sample of a given stage of the polyphase IIR filter is provided as an input of a next stage of the polyphase IIR filter. The transformed sample of the given stage can be understood to be an intermediate output sample of the given stage. In an embodiment, transformed samples of a last stage of the polyphase IIR filter constitute the output samples of the polyphase IIR filter.

In an embodiment, the method further comprises placing the output samples into an output buffer (namely, an output array). In an embodiment, the media processing unit is configured to place the output samples into the output buffer, upon generating the output samples using the polyphase IIR filter. In an embodiment, separate output buffers are employed for the processed real components of the samples and the processed imaginary components of the samples.

In an embodiment, the output samples constitute the at least one audio output. In other words, the at least one audio output is generated in the form of discrete output samples.

The method comprises providing, via the at least one speaker, the at least one audio output to the user. In other words, the at least one speaker is configured to provide the at least one audio output to the user.

Herein, the term "*speaker*" refers to an electroacoustic transducer which converts the at least one audio output into a corresponding sound. The at least one sound is heard by the user of the device. When the user hears the at least one sound, he/she is provided with a cognitive experience that is different from a cognitive experience which would have been provided by provision of the at least one audio input (without processing). In other words, the at least one audio input is processed by the device to produce the at least one audio output such that when the at least one audio output is provided to the user, the user is provided with an immersive, rich cognitive experience.

It will be appreciated that the at least one speaker is coupled to the media processing unit, either directly, or via one or more components of the device. At a given speaker, an electrical current corresponding to the at least one audio output is passed through a coil of a wire, thereby inducing a magnetic field. This magnetic field interacts with a permanent magnet connected to a cone of the given speaker. The cone moves back and forth to produce air pressure waves that the user hears as sound. The term "*speaker*" may commonly be referred to as "*loudspeaker*".

It will also be appreciated that in some implementations, the at least one speaker is integrated into the device, whereas in other implementations, the at least one speaker is a peripheral element that can be detachably coupled to the media processing unit via an input/output port. In an example, when the device is implemented as a headphone, two speakers are integrated (namely, built-in) into a body of the headphone. In another example, when the device is implemented as a laptop computer, the at least one speaker is a peripheral element that can be coupled to the media processing unit via an audio jack of the laptop computer.

Notably, the cognitive experience provided by providing the at least one audio output is improved over cognitive effects provided by conventional audio technologies. When the at least one audio output is provided to the user, the immersive cognitive experience provided to the user is related to the at least one audio input, but is far more improved as compared to the cognitive effect that would be provided if the at least one audio input were provided to the user. In particular, separated processed renderings of the real components of the samples and the imaginary components of the samples, when provided to the user, enable the user to experience improved cognitive effects owing to different levels of phase information (namely, phase components) of the at least one audio output. This ensures the improved cognitive effect and an immersive audio experience for the user. In the provided at least one audio output, as the processed renderings of the real components and the imaginary components are not locked together, the user can perceive these different components effectively to experience a live feeling of hearing the at least one audio output.

It will be appreciated that the processed renderings of both the real components of the samples and the imaginary components of the samples are reassembled when these processed renderings reach the user's brain independently via separate sensory mechanisms (such as the user's ears). These processed renderings remain separated until the neurological action of perception causes them to the reintegrated (i.e. "reassembled") into a singular stimulus via the cognitive activity taking place within the user's brain. In other words, reassembly of processed bi-phasic components of the at least one audio input occurs within the user's brain.

In an embodiment, the processed rendering of the real components of the samples and the processed rendering of the imaginary components of the samples are provided to different ears of the user. In such a case, the user's brain is asked to resolve the phase difference between these processed renderings. Such activity of the user's brain leads the user to perceive to an illusion of a multi-directional source of the at least one audio output thereby giving a live feeling to the user. Given the out-of-phase nature of the aforesaid processed renderings, the user's brain would integrate their sounds in a manner that produces a much richer cognitive experience than if the user heard a) the original audio input, (b) only one of the real and imaginary components, or (c) a signal resulting from a reverse synthesis of the real and imaginary components.

In an example, complex samples of a single audio input may be fed into the polyphase IIR filter one sample at a time. For each sample, the polyphase IIR filter performs bi-phasic separation to obtain a real component and an imaginary component. These real and imaginary components of the complex samples are processed by the polyphase IIR filter, and processed renderings of the real components are presented to the user via one output channel and an inverse of the processed renderings of the imaginary components are presented to the user via another output channel.

In an embodiment, the at least one audio input comprises a first audio input and a second audio input, and the at least one waveform comprises a first waveform and a second waveform corresponding to the first audio input and the second audio input, respectively. Herein, the terms "*first waveform*" and "*second waveform*" are used merely to distinguish between two distinct waveforms corresponding to two distinct audio inputs. Here, the "*first audio input*" and the "*second audio input*" are different audio inputs of two separate audio channels. The first audio input and the second audio input may be understood to be "*stereo audio inputs*".

In an embodiment, the media handling unit is configured to receive the first audio input and the second audio input at a first input channel and a second input channel, respectively. Optionally, in this regard, the first audio input and the second audio input are processed using the polyphase IIR filter, via different processing channels.

In an embodiment, the media handling unit is configured to receive the first audio input and the second audio input from a same media source. In another embodiment, the media handling unit is configured to receive the first audio input and the second audio input from different media sources. As an example, the first audio input may be recorded via a first microphone, whereas the second audio input may be recorded via a second microphone, the first and second microphones being placed at different locations in a live concert venue.

In an embodiment, the method comprises:

- generating a first set of samples of the first waveform on a first channel and a second set of samples of the second waveform on a second channel;
- initializing the polyphase IIR filter with a first set of one or more coefficients and a second set of one or more coefficients for processing the first set of samples and the second set of samples, respectively;
- applying the polyphase IIR filter to the first set of samples and the second set of samples to generate a first audio output and a second audio output on a first output channel and a second output channel, respectively; and
- providing the first audio output and the second audio output to the user.

In an embodiment, the media processing unit is configured to:

- generate the first set of samples of the first waveform on the first channel and the second set of samples of the second waveform on the second channel;
- initialize the polyphase IIR filter with a first set of one or more coefficients and a second set of one or more coefficients for processing the first set of samples and the second set of samples, respectively; and
- apply the polyphase IIR filter to the first set of samples and the second set of samples to generate a first audio output and a second audio output on a first output channel and a second output channel, respectively.

It will be appreciated that the first channel and the second channel are different processing channels used by the media processing unit for processing the first audio input and the second audio input.

In an embodiment, the polyphase IIR filter comprises a first polyphase IIR filter entity for processing the first audio input and a second polyphase IIR filter entity for processing the second audio input. In an embodiment, the first and second polyphase IIR filter entities are configured to work parallelly. In an embodiment, at a given polyphase IIR filter entity, real components and imaginary components of samples of a given set are processed separately.

Optionally, each sample of the first set of samples has a real component and an imaginary component. Similarly, optionally, each sample of the second set of samples has a real component and an imaginary component.

In an embodiment, the first set of one or more coefficients for processing the first set of samples is the same as the second set of one or more coefficients for processing the second set of samples. In another embodiment, the first set of one or more coefficients for processing the first set of samples is different from the second set of one or more coefficients for processing the second set of samples.

Optionally, coefficients of the first set for processing the real components of the first set of samples are different from coefficients of the first set for processing the imaginary components of the first set of samples. Similarly, optionally, coefficients of the second set for processing the real components of the second set of samples are different from coefficients of the second set for processing the imaginary components of the second set of samples.

In an embodiment, when applying the polyphase IIR filter to a given set of samples to generate a given audio output, the media processing unit is configured to:

- fetch a required number of samples of the given set from the input buffer and/or memory elements;
- transform the samples through the one or more stages of the polyphase IIR filter, by applying corresponding coefficients of the given set to the samples at each stage of the polyphase IIR filter; and
- obtain given output samples from the polyphase IIR filter to generate the given audio output.

In an embodiment, the first audio output comprises processed renderings of real components of the first set of samples and imaginary components of the first set of samples, whereas the second audio output comprises processed renderings of real components of the second set of samples and imaginary components of the second set of samples.

Optionally, real components of first output samples form the processed rendering of the real components of the first set of samples and imaginary components of the first output samples form the processed rendering of the imaginary components of the first set of samples. Similarly, optionally, real components of second output samples form the processed rendering of the real components of the second set of samples and imaginary components of the second output samples form the processed rendering of the imaginary components of the second set of samples. It will be appreciated that optionally, the first output samples are generated by the polyphase IIR filter upon processing the first set of samples, whereas the second output samples are generated by the polyphase IIR filter upon processing the second set of samples.

In another example, stereo audio inputs S1 and S2 (corresponding to left and right input channels, respectively) may be fed into two separate, parallel, cascaded polyphase IIR filters for processing. Upon processing, there are generated the following outputs - processed real components of the audio input S1, processed imaginary components of the audio input S1, processed real components of the audio input S2, processed imaginary components of the audio input S2. Herein, a processed rendering to be provided to a left ear of the user (via a left output channel) may be obtained by subtracting processed imaginary components of the audio input S2 from the processed real components of the audio input S1. Moreover, a processed rendering to be provided to a right ear of the user (via a right output channel) may be obtained by subtracting the processed imaginary components of the audio input S1 from the processed real components of the audio input S2.

In an embodiment, the at least one speaker comprises a first speaker and a second speaker configured to provide the first audio output and the second audio output, respectively, to the user. As an example, the device may be implemented as a headphone having the first and second speakers, wherein a given speaker provides a given audio output to its corresponding ear. The first and second speakers may be integrated with the headphone.

In an embodiment, the first audio output and the second audio output are provided to a first ear of the user and a second ear of the user, respectively. In other words, sound of the first audio output is rendered in the first ear of the user, whereas sound of the second audio output is rendered in the second ear of the user.

In an embodiment, the first audio output and the second audio output are provided simultaneously. It will be appreciated that the first audio output and the second audio output, when provided, provide a realistic and immersive listening experience to the user that emulates a live audio listening experience.

In an embodiment, when providing a given audio output to the user, the media processing unit is configured to mix the processed renderings of real components of samples and imaginary components of samples. In this regard, the media processing unit may automatically select a percentage of mixing, or may determine a percentage of mixing based on the user's input. As an example, the first audio output provided to the user may be 100 percent processed rendering of the real components of the first set of samples, 90 percent the processed rendering of the real components and 10 percent the processed rendering of the imaginary components of the first set of samples, 80 percent the processed rendering of the real components and 20 percent the processed rendering of the imaginary components, and so on till 100 percent the processed rendering of the imaginary components. In interpreting these numbers, it should be appreciated that the recitation of ranges herein is merely intended to serve as a shorthand method of referring individually to each separate value falling within the range. Thus, unless otherwise expressly indicated, each individual intervening value is incorporated into the specification as if it were individually recited herein. In addition, numeric values set forth herein should be construed in light of the number of reported significant digits, and by applying ordinary rounding techniques.

In an embodiment, the method further comprises converting, prior to providing to the user, the at least one audio output from a digital form to an analog form. In an embodiment, the device further comprises a digital-to-analog converter configured to convert, prior to provision to the user, the at least one audio output from the digital form to the analog form. The digital-to-analog converter is a specialized module of the device that, in operation, converts digital signals into analog signals. The digital-to-analog converter may be implemented as a dedicated integrated circuit (IC), as a set of discrete electronic components, or as a constituent module of a system on a chip (SoC) that integrates all or most components of the device.

In an embodiment, the digital-to-analog converter is coupled to the media processing unit in a manner that an output of the media processing unit serves as an input for the digital-to-analog converter. Notably, the output samples (of the at least one audio output) generated by the media processing unit is in digital form. The output samples serve as the input for the digital-to-analog converter. The digital-to-analog converter processes (the output samples of) the at least one audio output in a manner that an output of the digital-to-analog converter is the at least one audio output in an analog form. In other words, the digital-to-analog converter converts a form of the at least one audio output from the digital form to the analog form. Said conversion of the at least one audio output from the digital form to the analog form occurs prior to provision of the at least one audio output to the user. According to an embodiment, the digital-to-analog converter is coupled to the media processing unit either directly, or via one or more components of the device. According to another embodiment, the digital-to-analog converter is integrated with the media processing unit. In this case, the at least one audio output generated by the media processing unit is in analog form.

In an embodiment, the digital-to-analog converter is coupled to the at least one speaker in a manner that the output of the digital-to-analog converter serves as an input for the at least one speaker. The output of the digital-to-analog converter is the at least one audio output in the analog form, said at least one audio output serving as the input for the at least one speaker. It will be appreciated that the digital-to-analog converter is coupled to the at least one speaker either directly, or via one or more components of the device.

In an embodiment, the method further comprises modifying, prior to providing to the user, a magnitude of the at least one audio output. In an embodiment, the device further comprises an amplifier configured to modify, prior to provision to the user, the magnitude of the at least one audio output. The term "*amplifier*" refers to an electronic equipment that can modify (namely, adjust) magnitude of a given signal passing therethrough. In this regard, the amplifier may either increase or decrease the magnitude of the given signal, as required. Herein, gain of the amplifier refers to a ratio of a magnitude of an output of the amplifier to a magnitude of an input of the amplifier. When the gain of the amplifier is less than one, the magnitude of the given signal passing through the amplifier is decreased. Alternatively, when the gain of the amplifier is greater than one, the magnitude of the given signal passing through the amplifier is increased. In simpler terms, "*magnitude*" of a given audio signal can be understood to be "*volume*" of the given audio signal. The amplifier may be implemented as a dedicated integrated circuit (IC), as a set of discrete electronic components, or as a constituent module of a system on a chip (SoC) that integrates all or most components of the device.

It will be appreciated that when the at least one audio input is processed using the aforesaid method to yield the at least one audio output, a previously unheard phase dimension is added to the at least one audio output, effectively doubling the number of partials and phase variables in any given moment. In other words, the number of phase components in the at least one audio input is increased (as the at least one audio output includes processed renderings of both real components and imaginary components of the samples). This in turn, increases a magnitude (namely, an amplitude or a volume) of the at least one audio output vis-à-vis a magnitude of the at least one audio input by 3 decibels (dB). As an example, let us consider an amplitude of a given audio input to be 1. Then, an amplitude of both a real component of a given audio output and an imaginary component of the given audio output is 1. In such a case, the magnitude of the given audio output would be calculated as follows:

Magnitude = sqrt (r^2 + i^2) = sqrt (1^2 + 1^2) = sqrt (2) = 1.414

In terms of decibels, Decibels (dB) = 20*log10(1.414) ~+3dB, there is a 3 decibel increase in perceived magnitude of the given audio output in relation to the magnitude of the given audio input.

In an embodiment, when modifying the magnitude of the at least one audio output, the media processing unit is configured to reduce the magnitude of the at least one audio output to prevent distortion of the at least one audio output during provision. As described above, the magnitude of the at least one audio output is higher than the magnitude of the at least one audio input. When a given audio output is generated using a given audio input whose magnitude equals a maximum allowable level, a magnitude of the given audio output would be increased to such an extent beyond the maximum allowable level that clipping distortion would occur if the given audio output were provided to the user. This clipping distortion would be audible to the user (and therefore, be perceivable to the user) and would considerably diminish the listening experience of the user.

Therefore, optionally, the magnitude of the at least one audio output is reduced by a gain factor to prevent distortion of the at least one audio output during provision. In other words, the magnitude of the at least one audio output is multiplied by the gain factor to control a gain thereof. In particular, the at least one audio output is attenuated by the gain factor. Here, the gain factor may be understood to be a gain associated with the amplifier. Such a manner of controlling gain of the amplifier provides enough headroom for enabling addition and mixing of the separated real components and the imaginary components of the samples in the at least one audio output, without causing clipping or distortion of the at least one audio output.

In an embodiment, the gain factor is equal to 1/(√2). In such a case, the gain factor is equal to 0.70710678118655. In another embodiment, the gain factor is equal to 1/2. In such a case, the gain factor is equal to 0.5, which is an experimental value for real:imaginary combinatorics with non-linear audio signal sources. Optionally, the gain factor lies in a range of 0.5 to 1. For example, the gain factor may be equal to 0.5, 0.55. 0.6, 0.65, 0.7, 0.75, 0.8, 0.85, 0.9, 0.95, 1 or any other intermediate value. It will be appreciated that the aforesaid values and range of the gain factor are exemplary only, and other values or ranges of the gain factor may be employed as required. Optionally, the gain factor is selected based on audio signal type (for example, such as mono or stereo), selected mode of operation (for example, such as audio signal restoration with spatialization, audio signal restoration only, and the like), or similar criteria. As an example, when no gain reduction is required, the gain factor may be selected to be equal to 1.

In an embodiment, each output sample in the output buffer is multiplied by the gain factor.

Optionally, when the at least one audio input comprises the first audio input and the second audio input, the method further comprises employing an attenuation multiplier to pad the at least one audio input by -3dB. This compensates for √2 magnitude increase in partial density afforded by addition of a resynthesized imaginary stream to a pre-existing real stream. The attenuation multiplier may be employed upon receiving the at least one audio input.

In an embodiment, the method further comprises:

- obtaining an input from the user, the input pertaining to an audio signal that the user wishes to be provided;
- determining, based on the input, whether or not to process the at least one audio input to generate the at least one audio output; and
- providing either the at least one audio input or the at least one audio output to the user.

In an embodiment, the device further comprises an input element coupled to the media handling unit, wherein the media handling unit is further configured to:

- obtain an input from the user via the input element, the input pertaining to the audio signal that the user wishes to be provided;

- determine, based on the input, whether or not to process the at least one audio input to generate the at least one audio output; and
- enable provision of either the at least one audio input or the at least one audio output to the user.

Optionally, in this regard, the media handling unit acts as a logic processing controller that receives the input from the input element and takes necessary actions according to the input.

Throughout the present disclosure, the term "*input element*" refers to a component that is used by the user to provide his/her input to the device. In other words, input element is a means by which the user provides his/her input to interact with the device. Examples of a given input element include, but are not limited to, a push button, a slider, a switch, a touch panel, and a scrolling wheel. It will be appreciated that the input element may be coupled to the media handling unit electrically (namely, via electrical connections), mechanically (namely, via physical connections), communicably, or a combination of these.

As an example, the input element may be implemented as a push button, wherein the user may press the push button to indicate his/her wish of hearing the at least one audio output and may release the push button to indicate his/her wish of hearing the at least one audio input.

In an embodiment, the input device is configured to generate a control signal indicative of the input, and to send the control signal to the media handling unit. In another embodiment, the media handling unit is configured to generate a control signal indicative of the input, upon detecting the input at the input element.

In an embodiment, the media handling unit is configured to process the input of the user (and more optionally, the control signal) to determine which audio signal the user wishes to be provided. This enables the media handling unit to determine whether or not to process the at least one audio input to generate the at least one audio output. When the input is indicative of the user's wish to be presented the at least one audio output, it is determined that the at least one audio input needs be processed to generate the at least one audio output, and the at least one audio output is provided to the user. In other words, the user is only provided processed (according to the aforesaid method) audio signal(s). Alternatively, when the input is indicative of the user's wish to be presented the at least one audio input, it is determined that the at least one audio input need not be processed to generate the at least one audio output, and the at least one audio input is directly provided to the user. In other words, the user is only provided unprocessed audio signal(s).

It will be appreciated that such an input element optionally provides a "*bypass*" functionality in the device, wherein the user may employ said input element to provide an input for bypassing processing of the at least one audio input, if he/she desires. When the user provides the input for selecting the at least one audio input, processing of the at least one audio input using the polyphase IIR filter is bypassed and the at least one audio input is provided to the user in its unprocessed form. As an example, the user may provide a toggled input via the input element, wherein based on the toggled input, the at least one audio input is provided at a first time to the user, while the audio output is provided at a second time to the user. In such a case, user can compare the at least one audio input and the at least one audio output to ascertain and perceive different cognitive effects before and after processing of the at least one audio input using the device.

In an embodiment, the device comprises a plurality of input elements to obtain a plurality of inputs from the user. Optionally, in this regard, the plurality of input elements are provided on an input module of the device. In an embodiment, the plurality of inputs pertain to a manner of operating the device. In other words, the plurality of inputs are provided by the user, via the plurality of input elements, to exercise various functionalities of the device. As an example, 5 input elements may be provided on the input module of the device, the 5 input elements enabling the user to provide his/her input pertaining to '*playing/pausing a given audio signal*', '*increasing volume of a given audio signal*', '*decreasing volume of a given audio signal*', '*turning the device on/off*' and '*bypass audio input processing*' (described above).

As an example, the device may be implemented as a headphone or a portable speaker system having a plurality of push buttons to obtain the plurality of inputs from the user. The plurality of push buttons may be arranged ergonomically on the device.

Optionally, the plurality of input elements are individually coupled to the media handling unit. More optionally, the input module is coupled to the media handling unit. Optionally, the media handling unit acts as a logic processing controller that receives a given input from a given input element and takes necessary actions according to the given input.

DETAILED DESCRIPTION OF THE DRAWINGS

Referring to FIG. 1A, illustrated is an architecture of a device **100** for processing and providing audio information to a user, in accordance with an embodiment of the present disclosure. The device **100** comprises a media handling unit **102**, a media processing unit **104** coupled to the media handling unit **102**, and at least one speaker (depicted as a speaker **106**) coupled to the media processing unit **104**. The media handling unit **102** is configured to receive at least one audio input corresponding to the audio information from a media source (not shown). The media processing unit **104** is configured to: generate samples of at least one waveform corresponding to the at least one audio input, each sample of the at least one waveform having a real component and an imaginary component; initialize a polyphase infinite impulse response (IIR) filter with a set of one or more coefficients corresponding to a set of one or more stages of the polyphase IIR filter for both real components of the samples and imaginary components of the samples; and apply the polyphase IIR filter to the samples to generate at least one audio output, the at least one audio output comprising a processed rendering of both the real components of the samples and the imaginary components of the samples. The at least one speaker **106** is configured to provide the at least one audio output representing the audio information to the user.

Referring to FIG. 1B, illustrated is an example implementation of the device **100** of FIG. 1A, in accordance with an embodiment of the present disclosure. The device **100** is a headphone comprising the media handling unit **102**, the media processing unit **104**, and speakers **106A** and **106B**. The device **100** optionally also comprises a microphone **108**. When the device **100** comprises the microphone **108**, the device **100** may be understood to be a headset. The microphone **108** may be detachable.

It may be understood by a person skilled in the art that the FIGs. 1A and 1B include a simplified architecture of the device **100** and an example implementation of the device **100**, respectively, for sake of clarity, which should not unduly limit the scope of the claims herein. The person skilled in the art will recognize many variations, alternatives, and modifications of embodiments of the present disclosure.

Referring to FIG. 2, illustrated is an environment wherein a device **200** for processing and providing audio information to a user is used, in accordance with an embodiment of the present disclosure. The device **200** comprises a media handling unit **202**, a media processing unit **204** coupled to the media handling unit **202**, and at least one speaker (depicted as a first speaker **206** and a second speaker **208**) coupled to the media processing unit **204**. The media handling unit **202** is configured to receive at least one audio input from a media source **210**. When the at least one audio input comprises a first audio input and a second audio input, the media processing unit **204** employs the polyphase IIR filter to generate at least one audio output, wherein the at least one audio output comprises a first audio output and a second audio output. In such a case, the first speaker **206** and the second speaker **208** provide the first audio output and the second audio output, respectively, to the user.

As shown, the device **200** further comprises a digital-to-analog converter **212** configured to convert, prior to provision to the user, the at least one audio output from a digital form to an analog form. Moreover, the device **200** further comprises an amplifier **214** configured to modify, prior to provision to the user, a magnitude of the at least one audio output. The speakers **206** and **208** are coupled to the media processing unit **204** via the digital-to-analog converter **212** and the amplifier **214**.

The device **200** is shown to further comprise an input element **E** coupled to the media handling unit **202**, wherein the media handling unit **202** is further configured to:

- obtain an input from the user via the input element **E**, the input pertaining to an audio signal that the user wishes to be provided;
- determine, based on the input, whether or not to process the at least one audio input to generate the at least one audio output; and
- enable provision of either the at least one audio input or the at least one audio output to the user.

In this regard, the input element **E** is shown to be one input element among a plurality of input elements (depicted as input elements **A**, **B**, **C**, **D**, and **E**) of an input module **216** of the device **200**. In other words, the input element **E** enables the user to bypass processing of the at least one audio input, thereby providing a '*bypass audio input processing*' functionality to the user. The other input elements **A**, **B**, **C** and **D** of the input module **216** may, for example, enable the user to provide his/her input pertaining to '*playing/pausing a given audio signal*', '*increasing volume of a given audio signal*', '*decreasing volume of a given audio signal*', and '*turning the device **200** on/off* functionalities, respectively. Alternatively, the input element **E** is implemented as a single input element on the device **200**.

It may be understood by a person skilled in the art that the FIG. 2 includes a simplified illustration of the device **200** for sake of clarity, which should not unduly limit the scope of the claims herein. The person skilled in the art will recognize many variations, alternatives, and modifications of embodiments of the present disclosure.

Referring to FIG. 3, illustrated is an exemplary node **300** of a polyphase IIR filter, in accordance with an embodiment of the present disclosure. The node **300** corresponds to a stage of the polyphase IIR filter. At the node **300**, a new input sample X0 is received as shown. A new output sample Y0 is generated at the node **300** using the new input sample X0, historical input samples X_{-1} and X_{-2}, historical output samples Y_{-1} and Y_{-2}.

Given coefficients a0, a1, a2, b1, b2 corresponding to the samples X0, X_{-1}, X_{-2}, Y_{-1}, Y_{-2}, respectively, the new output sample Y0 is generated as:

$$Y0 = (a0*X0) + (a1*X_{-1}) + (a2*X_{-2}) + (b1*Y_{-1}) + (b2*Y_{-2})$$

Considering for example, values of the coefficients a0, a1, a2, b1, b2 as 0.161758, 0, -1, 0, -0.161758, respectively, an optimized equation for generating the new output sample Y0 is:

$$Y0 = (a0*X0) + (a2*X_{-2}) - (a0*Y_{-2})$$

As shown, the node **300** employs a feedback mechanism to eliminate redundancies, the feedback mechanism being implemented by way of double use feedback history loops. Notably, the feedbacks history loops loop through the new input sample X0, the two historical input samples X_{-1} and X_{-2}, and the two historical output samples Y_{-1} and Y_{-2} in a manner that redundancies in the polyphase IIR filter are effectively eliminated.

It may be understood by a person skilled in the art that the FIG. 3 includes a simplified illustration of the node **300** for sake of clarity, which should not unduly limit the scope of the claims herein. The person skilled in the art will recognize many variations, alternatives, and modifications of embodiments of the present disclosure.

Referring to FIG. 4, illustrated is an exemplary implementation of a polyphase IIR filter **400**, in accordance with an embodiment of the present disclosure. As shown, the polyphase IIR filter **400** is implemented as a four-stage filter comprising eight nodes. Two sets of four nodes each are arranged parallelly to separately process real components and imaginary components of a given sample. In each set, the four nodes are serially cascaded one after the other. The real components of the given sample are, for example, processed in the left set of four serially cascades nodes. The imaginary components of the given sample are, for example, processed in the right set of four serially cascades nodes. The polyphase IIR filter **400** is shown to be initialized with coefficients presented in Tables 1A and 2A (in the description of embodiments section).

It may be understood by a person skilled in the art that the FIG. 4 includes a simplified illustration of the polyphase IIR filter **400** for sake of clarity, which should not unduly limit the scope of the claims herein. The person skilled in the art will recognize many variations, alternatives, and modifications of embodiments of the present disclosure.

Referring to FIGs. 5A and 5B, FIG. 5A illustrates representations of phase responses **502** and **504** of a low-frequency sine wave input for a polyphase IIR filter and a Hilbert transform, while FIG. 5B illustrates representations of phase responses **506** and **508** of a high-frequency sine wave input for the polyphase IIR filter and the Hilbert transform, in accordance with an embodiment of the present disclosure.

In FIG. 5A, both the phase response **502** for the polyphase IIR filter and the phase response **504** for the Hilbert transform are circular in shape, indicating that the phase transforms **502** and **504** of the low-frequency sine wave input are uniformly 90 degrees for both the polyphase IIR filter and the Hilbert transform.

In FIG. 5B, the phase response **506** for the polyphase IIR filter is circular in shape, whereas the phase response **508** for the Hilbert transform is elliptical in shape. This indicates that for the high-frequency sine wave input, the phase response **508** for the Hilbert transform begins to distort from a pure circular form. However, the phase response **506** generated by the polyphase IIR filter retains its pure circular shape (even at the Nyquist frequency) for the high-frequency sine wave input.

The comparison between the phase responses **502**, **504**, **506** and **508** across a frequency spectrum indicates an improvement in quality of the phase transformation of sine wave input using the polyphase IIR filter over using the Hilbert transform.

Referring to FIG. 6, illustrated are steps of a method of processing and providing audio information to a user, in accordance with an embodiment of the present disclosure. At step **602**, at least one audio input is received. At step **604**, samples of at least one waveform corresponding to the at least one audio input are generated. Each sample of the at least one waveform has a real component and an imaginary component. At step **606**, a polyphase infinite impulse response (IIR) filter is initialized with a set of one or more coefficients corresponding to a set of one or more stages of the polyphase IIR filter for both the real components of the samples and the imaginary components of the samples. At step **608**, the polyphase IIR filter is applied to the samples to generate at least one audio output. The at least one audio output comprises a processed rendering of both the real components of the samples and the imaginary components of the samples. At step **610**, the at least one audio output is provided to the user.

The steps **602** to **610** are only illustrative and other alternatives can also be provided where one or more steps are added, one or more steps are removed, or one or more steps are provided in a different sequence without departing from the scope of the claims herein.

Modifications to embodiments of the present disclosure described in the foregoing are possible without departing from the scope of the present disclosure as defined by the accompanying claims. Expressions such as "including", "comprising", "incorporating", "have", "is" used to describe and claim the present disclosure are intended to be construed in a non-exclusive manner, namely allowing for items, components or elements not explicitly described also to be present. Reference to the singular is also to be construed to relate to the plural.

CLAIMS

What is claimed is:

1. A method of processing and providing audio information to a user, the method comprising:

- receiving at least one audio input corresponding to the audio information;
- generating samples of at least one waveform corresponding to the at least one audio input, wherein each sample of the at least one waveform has a real component and an imaginary component;
- initializing a polyphase infinite impulse response (IIR) filter with a set of one or more coefficients corresponding to a set of one or more stages of the polyphase IIR filter for both real components of the samples and imaginary components of the samples;
- applying the polyphase IIR filter to the samples to generate at least one audio output, the at least one audio output comprising processed renderings of both the real components of the samples and the imaginary components of the samples; and
- providing the at least one audio output representing the audio information to the user.

2. A method of claim 1, wherein the at least one audio input comprises a first audio input and a second audio input, and wherein the at least one waveform comprises a first waveform and a second waveform corresponding to the first audio input and the second audio input, respectively.

3. A method of claim 2, wherein the method comprises:

- generating a first set of samples of the first waveform on a first channel and a second set of samples of the second waveform on a second channel;
- initializing the polyphase IIR filter with a first set of one or more coefficients and a second set of one or more coefficients for processing the first set of samples and the second set of samples, respectively;
- applying the polyphase IIR filter to the first set of samples and the second set of samples to generate a first audio output and a second audio output on a first output channel and a second output channel, respectively; and
- providing the first audio output and the second audio output to the user.

4. A method of claim 3, wherein the first audio output comprises processed renderings of real components of the first set of samples and imaginary components of the first set of samples, whereas the second audio output comprises processed renderings of real components of the second set of samples and imaginary components of the second set of samples.

5. A method of any of claims 1-4, wherein the polyphase IIR filter is implemented as a four-stage filter.

6. A method of any of claims 1-5, wherein each stage of the polyphase IIR filter employs a feedback mechanism to eliminate redundancies.

7. A method of any of claims 1-6, further comprising converting, prior to providing to the user, the at least one audio output from a digital form to an analog form.

8. A method of any of claims 1-7, further comprising modifying, prior to providing to the user, a magnitude of the at least one audio output.

9. A method of any of claims 1-8, further comprising:

- obtaining an input from the user, the input pertaining to an audio signal that the user wishes to be provided;
- determining, based on the input, whether or not to process the at least one audio input to generate the at least one audio output; and
- providing either the at least one audio input or the at least one audio output to the user.

10. A device for processing and providing audio information to a user, the device comprising:

- a media handling unit configured to receive at least one audio input corresponding to the audio information from a media source;
- a media processing unit coupled to the media handling unit, the media processing unit being configured to:
    - generate samples of at least one waveform corresponding to the at least one audio input, each sample of the at least one waveform having a real component and an imaginary component;

- initialize a polyphase infinite impulse response (IIR) filter with a set of one or more coefficients corresponding to a set of one or more stages of the polyphase IIR filter for both the real components of the samples and the imaginary components of the samples; and
- apply the polyphase IIR filter to the samples to generate at least one audio output, the at least one audio output comprising processed renderings of both the real components of the samples and the imaginary components of the samples; and

- at least one speaker coupled to the media processing unit, wherein the at least one speaker is configured to provide the at least one audio output representing the audio information to the user.

11. A device of claim 10, wherein the at least one audio input comprises a first audio input and a second audio input, and wherein the at least one waveform comprises a first waveform and a second waveform corresponding to the first audio input and the second audio input, respectively.

12. A device of claim 11, wherein the media processing unit is configured to:

- generate a first set of samples of the first waveform on a first channel and a second set of samples of the second waveform on a second channel;
- initialize the polyphase IIR filter with a first set of one or more coefficients and a second set of one or more coefficients for processing the first set of samples and the second set of samples, respectively; and
- apply the polyphase IIR filter to the first set of samples and the second set of samples to generate a first audio output and a second audio output on a first output channel and a second output channel, respectively.

13. A device of claim 12, wherein the first audio output comprises processed renderings of real components of the first set of samples and imaginary components of the first set of samples, whereas the second audio output comprises processed renderings of real components of the second set of samples and imaginary components of the second set of samples.

14. A device of claim 12 or 13, wherein the at least one speaker comprises a first speaker and a second speaker configured to provide the first audio output and the second audio output, respectively, to the user.

15. A device of any of claims 10-14, wherein the polyphase IIR filter is implemented as a four-stage filter.

16. A device of any of claims 10-15, wherein each stage of the polyphase IIR filter employs a feedback mechanism to eliminate redundancies.

17. A device of any of claims 10-16, further comprising a digital-to-analog converter configured to convert, prior to provision to the user, the at least one audio output from a digital form to an analog form.

18. A device of any of claims 10-17, further comprising an amplifier configured to modify, prior to provision to the user, a magnitude of the at least one audio output.

19. A device of any of claims 10-18, further comprising an input element coupled to the media handling unit, wherein the media handling unit is further configured to:

- obtain an input from the user via the input element, the input pertaining to an audio signal that the user wishes to be provided;
- determine, based on the input, whether or not to process the at least one audio input to generate the at least one audio output; and
- enable provision of either the at least one audio input or the at least one audio output to the user.

20. A device of any of claims 10-19, wherein the media source is at least one of: an internal memory of the device, a memory of an external device, a data repository, a cloud-based memory, a microphone.

CLAIMS

What is claimed is:

1. A method of processing and providing audio information to a user, the method comprising:

- receiving at least one audio input corresponding to the audio information;
- generating samples of at least one waveform corresponding to the at least one audio input, wherein each sample of the at least one waveform has a real component and an imaginary component;
- initializing a filter which implements a mathematical construct with a set of one or more coefficients corresponding to a set of one or more stages of the mathematical construct for both real components of the samples and imaginary components of the samples, wherein the mathematical construct is configured for bi-phasic separation of the audio input;
- applying the mathematical construct to the samples to generate at least one audio output, the at least one audio output comprising processed renderings of both the real components of the samples and the imaginary components of the samples; and
- providing the at least one audio output representing the audio information to the user.

2. The method of claim 1, wherein the mathematical construct is a Hilbert transformation.

3. The method of claim 1, wherein the filter is a biquad filter, and wherein the biquad filter comprises at least two pairs of two-pole, and at least two-zero biquad filters in a cascaded series.

4. The method of claim 1, wherein the at least one audio input comprises a first audio input and a second audio input, and wherein the at least one waveform comprises a first waveform and a second waveform corresponding to the first audio input and the second audio input, respectively.

5. The method of claim 4, wherein the method comprises:

- generating a first set of samples of the first waveform on a first channel and a second set of samples of the second waveform on a second channel;
- initializing the mathematical construct with a first set of one or more coefficients and a second set of one or more coefficients for processing the first set of samples and the second set of samples, respectively;
- applying the mathematical construct to the first set of samples and the second set of samples to generate a first audio output and a second audio output on a first output channel and a second output channel, respectively; and
- providing the first audio output and the second audio output to the user.

6. The method of claim 5, wherein the first audio output comprises processed renderings of real components of the first set of samples and imaginary components of the first set of samples, whereas the second audio output comprises processed renderings of real components of the second set of samples and imaginary components of the second set of samples.

7. The method of any of claims 1 to 6, wherein the filter is implemented as a four-stage filter.

8. The method of any of claims 1 to 7, wherein each stage of the filter employs a feedback mechanism to eliminate redundancies.

9. The method of any of claims 1 to 8, further comprising converting, prior to providing to the user, the at least one audio output from a digital form to an analog form.

10. The method of any of claims 1 to 9, further comprising modifying, prior to providing to the user, a magnitude of the at least one audio output.

11. The method of any of claims 1 to 10, further comprising:

- obtaining an input from the user, the input pertaining to an audio signal that the user wishes to be provided;
- determining, based on the input, whether or not to process the at least one audio input to generate the at least one audio output; and
- providing either the at least one audio input or the at least one audio output to the user.

12. A device for processing and providing audio information to a user, the device comprising:

- a media handling unit configured to receive at least one audio input corresponding to the audio information from a media source;
- a media processing unit coupled to the media handling unit, the media processing unit being configured to:
- generate samples of at least one waveform corresponding to the at least one audio input, each sample of the at least one waveform having a real component and an imaginary component;
- initialize a filter which implements a mathematical construct with a set of one or more coefficients corresponding to a set of one or more stages of the mathematical construct for both the real components of the samples and the imaginary components of the samples; and
- apply the mathematical construct to the samples to generate at least one audio output, the at least one audio output comprising processed renderings of both the real components of the samples and the imaginary components of the samples; and

- at least one speaker coupled to the media processing unit, wherein the at least one speaker is configured to provide the at least one audio output representing the audio information to the user.

13. The device of claim 12, wherein the mathematical construct is a Hilbert transformation.

14. The device of claim 12, wherein the filter is a biquad filter, and wherein the biquad filter comprises at least two pairs of two-pole, and at least two-zero biquad filters in a cascaded series.

15. The device of claim 12, wherein the at least one audio input comprises a first audio input and a second audio input, and wherein the at least one waveform comprises a first waveform and a second waveform corresponding to the first audio input and the second audio input, respectively.

16. The device of claim 15, wherein the media processing unit is configured to:

- generate a first set of samples of the first waveform on a first channel and a second set of samples of the second waveform on a second channel;
- initialize the mathematical construct with a first set of one or more coefficients and a second set of one or more coefficients for processing the first set of samples and the second set of samples, respectively; and
- apply the mathematical construct to the first set of samples and the second set of samples to generate a first audio output and a second audio output on a first output channel and a second output channel, respectively.

17. The device of claim 16, wherein the first audio output comprises processed renderings of real components of the first set of samples and imaginary components of the first set of samples, whereas the second audio output comprises processed renderings of real components of the second set of samples and imaginary components of the second set of samples.

18. The device of claim 16 or 17, wherein the at least one speaker comprises a first speaker and a second speaker configured to provide the first audio output and the second audio output, respectively, to the user.

19. The device of any of claims 12 to 18, wherein the filter is implemented as a four-stage filter.

20. The device of any of claims 12 to 19, wherein each stage of the filter employs a feedback mechanism to eliminate redundancies.

21. The device of any of claims 12 to 20, further comprising a digital-to-analog converter configured to convert, prior to provision to the user, the at least one audio output from a digital form to an analog form.

22. The device of any of claims 12 to 21, further comprising an amplifier configured to modify, prior to provision to the user, a magnitude of the at least one audio output.

23. The device of any of claims 12 to 22, further comprising an input element coupled to the media handling unit, wherein the media handling unit is further configured to:

- obtain an input from the user via the input element, the input pertaining to an audio signal that the user wishes to be provided;
- determine, based on the input, whether or not to process the at least one audio input to generate the at least one audio output; and
- enable provision of either the at least one audio input or the at least one audio output to the user.

24. The device of any one of claims 12 to 23, wherein the media source is at least one of: an internal memory of the device, a memory of an external device, a data repository, a cloud based memory, a microphone.

Intellectual Property Office

Application No:	GB2215286.2	**Examiner:**	Dr Maurice Blount
Claims searched:	1-24	**Date of search:**	22 March 2023

Patents Act 1977: Search Report under Section 17

Documents considered to be relevant:

Category	Relevant to claims	Identity of document and passage or figure of particular relevance
X	1-24	WO 2015/047466 A2 (INNERSENSE) See figure 2 and paragraphs [0005], [0006],[0048] and [0051] in particular.
X	1, 2, 4-6, 9-13, 21, 23 and 24	US 2013/0202119 A1 (THEIDE) See figure 2 and paragraphs [0045]-[0050] in particular.
X	1, 2, 4-6, 9-13, 21, 23 and 24	US 2016/0249138 A1 (VAN DER WERF) See paragraphs [0003]-[0008] in particular.

Categories:

X	Document indicating lack of novelty or inventive step	A	Document indicating technological background and/or state of the art.
Y	Document indicating lack of inventive step if combined with one or more other documents of same category.	P	Document published on or after the declared priority date but before the filing date of this invention.
&	Member of the same patent family	E	Patent document published on or after, but with priority date earlier than, the filing date of this application.

Field of Search:

Search of GB, EP, WO & US patent documents classified in the following areas of the UKC^{X} :

Worldwide search of patent documents classified in the following areas of the IPC

The following online and other databases have been used in the preparation of this search report

WPI, EPODOC

International Classification:

Subclass	Subgroup	Valid From
G10L	0019/008	01/01/2013
H04S	0005/00	01/01/2006