(12) **EUROPEAN PATENT SPECIFICATION**

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(54) **Method for operating a hearing device and hearing device**

Verfahren zum Betreiben eines Hörgeräts und Hörgerät

Procédé de fonctionnement d'un dispositif auditif et dispositif auditif

EP 2 768 246 B1

**Description**

<u>TECHNICAL FIELD</u>

5       **[0001]**     The present invention relates to a method for operating a hearing device and to a corresponding hearing device. More specifically, the present invention relates to sample-rate conversion in such hearing devices. The invention may e.g. be useful in applications such as e.g. a hearing aid or a listening device, which receives acoustic signals from a person's surroundings, modifies the acoustic signals electronically and transmits the modified acoustic signals into the person's ear or ear canal, or such as e.g. a headset, which receives audio signals electronically and transmits corre-
10      sponding acoustic signals into the person's ear or ear canal.

<u>BACKGROUND ART</u>

**[0002]**     Hearing devices that perform digital signal processing of digital audio signals received from another device are
15      known in the art. Such hearing devices may be designed so that the processing rate, i.e. the rate with which the signal processing requires input samples, equals the input rate, i.e. the rate with which audio signal samples are received. However, various factors, such as production tolerances, variations in the clock frequencies of the transmitting and receiving devices and errors in the transmission, may cause the input rate to at least temporarily deviate from the processing rate. At the same time it is often desired to keep the processing rate constant over time. In such cases, the
20      input signal is preferably pre-processed to ensure that the signal processing receives the required samples at the processing rate.
        **[0003]**     A known and rather simple solution is to provide the hearing device with an input buffer into which the received samples are written on arrival and from which the respective oldest unread sample is read by the signal processing when it requires a new sample. When a buffer under-run occurs, i.e. when the buffer is empty when a read operation is
25      due, then the last sample read is repeated as input to the signal processing. When a buffer over-run occurs, i.e. when the buffer is full when a write operation is due, then the sample to be written is discarded (skipped). This simple form of sample-rate conversion, in the following referred to as skip-and-repeat, has the disadvantage that the skipping and repeating of samples causes clearly audible artefacts in the processed signal. It is further only successfully applicable when the input rate and the processing rate are close to each other.
30      **[0004]**     It is further known to upsample the input signal to provide samples at a resampling rate which is an integer multiple - defined as the upsampling factor - of the input rate. The skip-and-repeat procedure is performed on the upsampled signal, and thus at the resampling rate, to provide a sample rate equalling an integer multiple of the processing rate. The resulting signal is subsequently downsampled to the processing rate by an integer downsampling factor. This upsampled skip-and-repeat method also causes artefacts in the processed signal, but they are less audible than those
35      caused by the simple skip-and repeat method described above. Furthermore, by selecting different up- and downsampling factors this method may be used to perform sample-rate conversion when the input rate and the processing rate are not close to each other.
        **[0005]**     In the present context, the conversion ratio is defined as the ratio of the processing rate to the input rate. The range of conversion-ratio variations with which a device has to cope may differ between different systems and scenarios.
40      If the range of variations is large and/or if variations occur fast, then a more sophisticated control of the skipping and repeating in the upsampled skip-and-repeat method is required to avoid that multiple skip operations or multiple repeat operations occur immediately after each other, which could otherwise significantly reduce the quality of the processed signal. The upsampled skip-and-repeat method is more sensitive to such variations than the simple method because in the upsampled method a single missing input sample may cause several consecutive repetitions of an upsampled signal
45      sample. Where the input signal is received in packets each comprising multiple signal samples, the content of entire packets may be lost due to transmission errors. The loss of packets, and the well known and widely implemented retransmission of lost packets to avoid gaps in the processed audio signal, both worsen these problems.
        **[0006]**     In the above mentioned skip-and-repeat methods, multiple skip operations or multiple repeat operations occurring immediately after each other are mainly caused by the fact that the decision to increase or decrease the conversion
50      ratio is taken on the basis of information regarding a single sample and that the decision is immediately carried out, namely by skipping or repeating the respective sample. Therefore, alternative methods have been developed in the art, which provide a slower working control of the conversion ratio.
        **[0007]**     One such known alternative method comprises performing sample-rate conversion with a piecewise constant conversion ratio, e.g. by upsampling and downsampling as described above, however without skipping and repeating
55      upsampled samples. The received input samples are buffered, and the conversion ratio is increased when the buffer level, i.e. the number of buffered samples, sinks below a predetermined lower threshold, and conversely decreased when the buffer level grows above a predetermined upper threshold. The thresholds are preferably selected so that their difference is smaller than the total buffer size and they thus provide room both below the lower threshold and above the

upper threshold. This extra room is used to allow for overshoot of the control algorithm. The buffer size, the buffer thresholds and the step sizes with which the conversion ratio is respectively increased and decreased are chosen such that they together allow the method to adapt to expected variations in the input rate without risking actual buffer under-runs or over-runs. The method also works when one of the upsampling and the downsampling factor equals unity.

**[0008]** Signal processing generally requires resources, such as logic circuits, memory space, computation time or power, each of which is typically a limited resource in hearing devices - and even more so in battery-powered hearing devices. It is therefore desirable to implement sample-rate conversion in hearing devices using methods that are particularly efficient. In most sample-rate conversion methods, and particular in such methods comprising upsampling and subsequent downsampling of the resampled signal, filtering is required to avoid aliasing artefacts in the processed signal. This filtering adds significantly to the complexity and the resource consumption of the sample-rate conversion, and the design of the filters is therefore of high importance in hearing devices. Upsampling followed by - or integral with - filtering to remove aliased frequencies is in the art generally referred to as interpolation. In the following, the term upsampling should be interpreted to cover both upsampling without such filtering and interpolation.

**[0009]** An efficient known method for sample-rate conversion comprises the use of so-called polyphase filters in the upsampling and in the downsampling steps. Each received input sample is fed in parallel to a set of digital filters, which each represents a specific phase of an interpolation filter to be applied. The outputs of the phase filters in the set are cyclically sampled at the resampling rate, one cycle for each input sample, and the thus resampled samples are cyclically fed to individual digital filters in a second set. Each filter in the second set represents a specific phase of a downsampling filter to be applied, and the outputs of these phase filters are added at the processing rate, thus providing a sample-rate-converted signal at the processing rate. The interpolation filter is typically configured to suppress frequencies above half the input rate, and the downsampling filter is typically configured to suppress frequencies above half the processing rate. The interpolation filter and the downsampling filter thus suppress aliasing artefacts in the processed signal. The sum of the lengths of the phase filters in a set is equal to or slightly larger than the length of the respective interpolation or downsampling filter.

**[0010]** In [1], chapter 10.5.3, "Time-Variant Filter Structures", the authors disclose how sample-rate conversion using polyphase filters may be implemented in a particularly efficient manner. In the disclosed method, the interpolation filter and the downsampling filter are combined in a single filter configured such that it suppresses frequencies above half the minimum of the input rate and the processing rate. The method allows the computation of the sample-rate converted signal to be performed basically by buffering the received input samples and at the processing rate computing a scalar product of a subset of the buffered samples and a subset of filter coefficients of the filter. The actual subsets to be used are selected dynamically from respectively the input buffer and a set of filter coefficients, using respective indices that are dynamically updated for each output sample to be computed.

**[0011]** The above described methods based on polyphase filters, may obviously be used in combination with the above described method wherein a piecewise constant conversion ratio is changed when the buffer level exceeds predetermined buffer thresholds. The conversion ratio may be changed by changing the upsampling factor and/or the downsampling factor. Since, however, the lengths and the coefficients of the phase filters depend on the up- and downsampling factors, the lengths and the coefficients of the phase filters must either be recomputed or be read from a repository of pre-stored filter configurations each time the conversion ratio changes. This obviously requires computation time and/or storage space in addition to the resources required for the conversion itself.

**[0012]** Furthermore, changing the conversion ratio based on predetermined buffer thresholds as described above causes the latency, i.e. the time delay from an input sample is received until the corresponding output sample is delivered to the signal processing, to vary with time and, at least to some extent, unpredictably. Also, in the known methods and devices, the buffer thresholds are chosen in advance and such that adaptation is possible when the input rate variations are within a range expected at the time of implementing the method or manufacturing the device. If the variations are larger than expected, the sample-rate conversion may not function properly, and if the variations are smaller than expected, the latency will be longer than needed. However, in hearing devices a consistent and short latency is often important, particularly when a hearing-device user is able to see the source of the sound he or she is listening to, e.g. when the user listens to the sound of a video or a TV broadcast.

**[0013]** Also, in hearing devices wherein retransmission of lost packets is implemented, the size of the input buffer and the value of the lower buffer threshold are typically selected to allow for a predetermined number of consecutive retransmissions of the same lost packet before a buffer under-run occurs. This, however, causes the latency to be larger than needed during time periods without retransmission.

**[0014]** US 6,061,410 discloses a digital control arrangement used in connection with sampling rate conversion system having an input signal of one frequency converted to an output signal of another frequency.

**[0015]** US 7,246,057 discloses a receiver system in a communication system supporting packet-based communication, including a receiver, speech decoder and a jitter buffer for handling delay variations in the reception of a speech signal consisting of packets containing frames with encoded speech.

DISCLOSURE OF INVENTION

**[0016]** It is an object of the present invention to provide a method for operating a hearing device, which method overcomes the above problems. It is a further object of the present invention to provide a hearing device, which does not suffer from the above problems.

**[0017]** These and other objects of the invention are achieved by the invention defined in the accompanying independent claims and as explained in the following description. Further objects of the invention are achieved by the embodiments defined in the dependent claims and in the detailed description of the invention.

**[0018]** Controlling the conversion ratio in sample-rate conversion of an input signal buffered in an input buffer in a manner aiming at minimising the difference between estimated average or smoothed input-buffer levels and a target buffer level allows for robust sample rate conversion with precise control of the latency.

**[0019]** Performing the sample-rate conversion by upsampling the input signal and subsequently downsampling the upsampled signal in combination with controlling the conversion ratio by operations on the upsampled signal further allows for efficient and robust sample rate conversion with precise control of the latency at the cost of only minor artefacts in the processed signal. Depending on the implementation, the artefacts may be barely noticeable in speech and other every-day sounds, and the method may therefore advantageously be used in hearing devices.

**[0020]** Performing the sample-rate conversion by computing each sample of the output signal as a scalar product of a subset of the buffered samples and a subset of filter coefficients, wherein said subsets are selected anew for each computed sample from respectively the input buffer and a set of filter coefficients, and wherein the selection of said subset of the buffered samples is made in dependence on the estimated average or smoothed input-buffer level allows for even more efficient sample rate conversion. The achievable low level of generated artefacts in such embodiments is rather surprising, since the Nyquist criterion is in fact violated every time the selection of said subset of the buffered samples is altered due to a change in the estimated average or smoothed input-buffer level, and therefore, aliasing artefacts were expected to be clearly audible in the processed signal.

**[0021]** In the present context, a "hearing device" refers to a device, such as e.g. a hearing aid, a listening device or an active ear-protection device, which is adapted to improve, augment and/or protect the hearing capability of a user by receiving acoustic signals from the user's surroundings, generating corresponding audio signals, optionally modifying the audio signals and providing the received or modified audio signals as audible signals to at least one of the user's ears. A "hearing device" further refers to a device such as an earphone or a headset adapted to receive audio signals electronically, optionally modifying the audio signals and providing the received or modified audio signals as audible signals to at least one of the user's ears. Such audible signals may e.g. be provided in the form of acoustic signals radiated into the user's outer ears, acoustic signals transferred as mechanical vibrations to the user's inner ears through the bone structure of the user's head and/or through parts of the middle ear as well as electric signals transferred directly or indirectly to the cochlear nerve and/or to the auditory cortex of the user.

**[0022]** A hearing device may be configured to be worn in any known way, e.g. as a unit arranged behind the ear with a tube leading air-borne acoustic signals into the ear canal or with a loudspeaker arranged close to or in the ear canal, as a unit entirely or partly arranged in the pinna and/or in the ear canal, as a unit attached to a fixture implanted into the skull bone, as an entirely or partly implanted unit, etc. A hearing device may comprise a single unit or several units communicating electronically with each other.

**[0023]** More generally, a hearing device comprises an input transducer for receiving an acoustic signal from a user's surroundings and providing a corresponding input audio signal and/or a receiver for electronically receiving an input audio signal, a signal processing circuit for processing the input audio signal(s) and an output means for providing an audible signal to the user in dependence on the processed audio signal(s). Some hearing devices may comprise multiple input transducers, e.g. for providing direction-dependent audio signal processing. In some hearing devices, the receiver may be a wireless receiver. In some hearing devices, the receiver may be e.g. an input amplifier for receiving a wired signal. In some hearing devices, an amplifier may constitute the signal processing circuit. In some hearing devices, the output means may comprise an output transducer, such as e.g. a loudspeaker for providing an air-borne acoustic signal or a vibrator for providing a structure-borne or liquid-borne acoustic signal. In some hearing devices, the output means may comprise one or more output electrodes for providing electric signals.

**[0024]** In some hearing devices, the vibrator may be adapted to provide a structure-borne acoustic signal transcutaneously or percutaneously to the skull bone. In some hearing devices, the vibrator may be implanted in the middle ear and/or in the inner ear. In some hearing devices, the vibrator may be adapted to provide a structure-borne acoustic signal to a middle-ear bone and/or to the cochlea. In some hearing devices, the vibrator may be adapted to provide a liquid-borne acoustic signal in the cochlear liquid, e.g. through the oval window. In some hearing devices, the output electrodes may be implanted in the cochlea or on the inside of the skull bone and may be adapted to provide the electric signals to the hair cells of the cochlea, to one or more hearing nerves and/or to the auditory cortex.

**[0025]** A "hearing system" refers to a system comprising one or two hearing devices, and a "binaural hearing system" refers to a system comprising one or two hearing devices and being adapted to cooperatively provide audible signals

to both of the user's ears. Hearing systems or binaural hearing systems may further comprise "auxiliary devices", which communicate with the hearing devices and affect and/or benefit from the function of the hearing devices. Auxiliary devices may be e.g. remote controls, remote microphones, audio gateway devices, mobile phones, public-address systems, car audio systems or music players. Hearing devices, hearing systems or binaural hearing systems may e.g. be used for compensating for a hearing-impaired person's loss of hearing capability, augmenting or protecting a normal-hearing person's hearing capability and/or conveying electronic audio signals to a person.

[0026]   As used herein, the singular forms "a", "an", and "the" are intended to include the plural forms as well (i.e. to have the meaning "at least one"), unless expressly stated otherwise. It will be further understood that the terms "has", "includes", "comprises", "having", "including" and/or "comprising", when used in this specification, specify the presence of stated features, integers, steps, operations, elements and/or components, but do not preclude the presence or addition of one or more other features, integers, steps, operations, elements, components and/or groups thereof. It will be understood that when an element is referred to as being "connected" or "coupled" to another element, it can be directly connected or coupled to the other element, or intervening elements may be present, unless expressly stated otherwise. As used herein, the term "and/or" includes any and all combinations of one or more of the associated listed items. The steps of any method disclosed herein do not have to be performed in the exact order disclosed, unless expressly stated otherwise.

## BRIEF DESCRIPTION OF THE DRAWINGS

[0027]   The invention will be explained in more detail below in connection with preferred embodiments and with reference to the drawings in which:

FIG. 1 shows an embodiment of a hearing device according to the invention,

FIG. 2 shows a further embodiment of a hearing device according to the invention,

FIG. 3 illustrates upsampling in embodiments of the invention,

FIG. 4 illustrates downsampling in embodiments of the invention,

FIG. 5 illustrates sample-rate conversion in embodiments of the invention,

FIG. 6 illustrates determination of input-buffer levels in embodiments of the invention, and

FIG. 7 illustrates adaptive latency control in embodiments of the invention.

[0028]   The figures are schematic and simplified for clarity, and they just show details, which are essential to the understanding of the invention, while other details are left out. Throughout, like reference numerals and/or names are used for identical or corresponding parts.

[0029]   Further scope of applicability of the present invention will become apparent from the detailed description given hereinafter. However, it should be understood that the detailed description and specific examples, while indicating preferred embodiments of the invention, are given by way of illustration only, since various changes and modifications within the scope of the invention will become apparent to those skilled in the art from this detailed description. Within this description, any such modifications are mentioned in a non-limiting way.

## MODE(S) FOR CARRYING OUT THE INVENTION

[0030]   The first embodiment of a hearing device 100 shown in FIG. 1 comprises a receiver 101, an input buffer 102, a sample processor 103, an estimator 104, a ratio controller 105, an amplifier 106 and an output transducer 107. The hearing device 100 may e.g. be an earphone or a headset and may be intended to be worn in an operating position by a user, such as e.g. at, in, on or close to an ear of the user.

[0031]   A transmitter 108 transmits samples of a digital audio signal to the hearing device 100, e.g. by means of radio signals or other wired or wireless electronic signals. The transmitter may be any kind of device that is capable of transmitting a digital audio signal, such as e.g. a laptop computer, a mobile phone, a wireless microphone, a further hearing device etc. The transmission may be made using any suitable protocol, including continuous data protocols or packet data protocols such as e.g. Bluetooth Advanced Audio Distribution Profile (A2DP), which is a part of the Bluetooth 1.0 standard, or Bluetooth Low Energy (BLE), which is a part of the Bluetooth 4.0 standard. The transmitter 108 is external to the hearing device 100, but in some embodiments the transmitter 108 may instead be comprised by the

hearing device 100.

**[0032]** The receiver 101 receives the samples of the digital audio signal and feeds the received samples as a digital input signal $x(n)$ to the input buffer 102. The receiver 101, optionally assisted by the transmitter 108, may apply any known method and technology, such as e.g. message checksums, forward error correction coding and decoding, re-transmission of lost data packets etc., to ensure that errors in the received samples are corrected and/or to avoid that transmission errors cause gaps or other discontinuities in the audio signal provided in the digital input signal $x(n)$.

**[0033]** The sample processor 103 processes the buffered samples to provide samples of a digital output signal $y(m)$ such that the digital output signal $y(m)$ is a sample-rate converted representation of the digital input signal $x(n)$ with a predetermined target sample rate $F_y$, which is preferably equal to the processing rate of the amplifier 106. The sample processor 103 may use any of the sample-rate conversion methods known in the prior art, in particular any of the methods described further above, provided that the implemented method allows for a variation of the conversion ratio. The sample processor 103 preferably uses one or more of the methods explained below in relation to FIGs. 3, 4 and 5.

**[0034]** The estimator 104 repeatedly determines an average or smoothed input-buffer level $I(t)$ indicating an actual buffer level, i.e. the number N (see FIG. 6) of unprocessed samples in the input buffer 102. The average or smoothed input-buffer level $I(t)$ may be determined using any known averaging or smoothing method, such as e.g. repeatedly counting the number N of unprocessed samples in the input buffer 102 and computing an average over a sliding time window of the count results or low-pass filtering the count results. The counting may be achieved by any known counting method, such as e.g. computing the difference between an input pointer to the next write position and an output pointer to the next read position in the input buffer 102 or by incrementing a counter value when a write operation occurs and decrementing the counter value when a read operation occurs. In the case that samples in the input buffer 102 are skipped in the sample-rate conversion, such samples are regarded as processed and do thus not contribute to the actual buffer level N. The estimator 104 preferably uses one or more of the methods explained below in relation to FIG. 6.

**[0035]** The ratio controller 105 is adapted to control the sample processor 103 in a manner aiming at minimising the difference between the estimated average input-buffer level $I(t)$ and a target buffer level L. The ratio controller 105 may achieve this by means of any known method of minimising said difference, such as e.g. adaptive least-mean-squares (LMS) methods or proportional control methods. The ratio controller 105 preferably comprises or constitutes a propor-tional-derivative (PD) controller or a proportional-integral-derivative (PID) controller and thus preferably achieves its goal by means of respectively proportional-derivative control or proportional-integral-derivative control. The ratio controller 105 may preferably control the sample processor 103 by providing one or more control parameters to the sample processor 103. The control parameters may comprise any parameter that the sample processor 103 may use to determine that, and/or when, a skip or a repeat operation should be performed and/or to determine that the conversion ratio should be changed by other means, such as by changing an upsampling or a downsampling factor. Suitable control parameters may indicate e.g. a target conversion ratio, an amount of change to be effected of the current conversion ratio or a time shift to be effected. The sample processor 103 may adjust the conversion ratio to match as good as possible and within predetermined constraints the indicated target conversion ratio or a conversion ratio resulting from adding the indicated amount of change to the current conversion ratio, or to effect as good as possible and within predetermined constraints the indicated time shift in the processing of buffered input samples $x(n)$.

**[0036]** The amplifier 106 receives the samples of the digital output signal $y(m)$ from the sample processor 103 and provides a corresponding amplified signal to the output transducer 107. The amplifier 106 may be or comprise any known type of amplifier suitable for providing an amplified signal corresponding to the digital output signal $y(m)$, such as e.g. a digital pulse-width-modulator, a digital-to-analog converter followed by an analog amplifier or by an analog pulse-width-modulator, etc. The amplifier 106 may further comprise a signal processor (not shown) adapted to modify the digital output signal $y(m)$ in any known way, e.g. to suppress noise, reverberation, echoes, transients, acoustic feedback etc., to enhance speech perception, spatial cues or the like, to compensate for the user's loss of hearing capability, or to augment or protect the user's hearing capability.

**[0037]** The output transducer 107 converts the amplified signal into an acoustic signal. When the hearing device 1 is worn in the operation position by the user, the acoustic signal may enter the ear or the ear canal of the user, which may thus hear the acoustic signal and thus ideally perceive the original audio signal comprised in digital form by the samples received from the transmitter 108 or an enhanced or improved version thereof. In some embodiments, the output trans-ducer 107 and optionally the amplifier 106 may be replaced by other suitable output means, such as the ones mentioned in the general description of hearing devices in the section "Disclosure of the invention".

**[0038]** In some embodiments, the hearing device 100 may further comprise a latency controller 109 that controls the target buffer level L in dependence on an estimated quality of reception of the digital audio signal. The latency controller 109 may preferably increase the target buffer level L in dependence on the estimated quality of reception decreasing and vice versa. The quality of reception may preferably be estimated on the basis of repeated determinations of the actual buffer level N. The latency controller 109 preferably uses one or more of the methods explained below in relation to FIG. 7.

**[0039]** The hearing device 100 may further comprise a battery or an accumulator 110 that supplies electric power to

the electronic circuits of the hearing device 100. Alternatively, or additionally, the hearing device 100 may receive electric power from an external device, such as the transmitter 108, e.g. via the transmission signal or by other wired or wireless power transmission means.

[0040]    The second embodiment of a hearing device 200 shown in FIG. 2 comprises a receiver 101, an input buffer 102, a sample processor 103, an estimator 104, a ratio controller 105, an amplifier 106, an output transducer 107, an optional latency controller 109 and an optional battery or accumulator 110 having construction, connections and function as the corresponding elements of the first embodiment of FIG. 1, except for deviations explained below. The hearing device 200 may e.g. be a hearing aid, a listening device or an active ear-protection device and may be intended to be worn in an operating position at, in, on or near an ear of a user. The hearing device 200 may receive samples of a digital audio signal from a transmitter 108 as described above in relation to FIG. 1. The hearing device 200 further comprises microphone 201, a preamplifier 202, a digitiser 203, a signal processor 204 connected with the amplifier 106 and the output transducer 107 to form an audio signal path.

[0041]    The microphone 201 is arranged such that it may receive an acoustic input signal from the user's surroundings when the hearing device 200 is worn in the operating position by the user and thus provide a corresponding microphone signal to the preamplifier 202. The preamplifier 202 amplifies the microphone signal and provides the amplified micro-phone signal to the digitiser 203. The digitiser 203 digitises the amplified microphone signal and provides a digitised main audio signal to the signal processor 204, which may modify the main audio signal in accordance with the purpose of the hearing device 200, e.g. to improve, augment and/or protect the hearing capability of the individual. The signal processor 204 provides the modified audio signal to the amplifier 106 instead of the digital output signal $y(m)$.

[0042]    The signal processor 204 is further connected to receive the digital output signal $y(m)$ from the sample processor 103 and provides the modified audio signal to the amplifier 106 in dependence on a combination of the digital output signal $y(m)$ and the digitised main audio signal. If, for instance, the transmitter 108 is a mobile phone, the signal processor 204 may switch between providing a modified or unmodified version of the digital output signal $y(m)$ and a modified version of the digitised main audio signal in the modified audio signal to the amplifier 106, depending on whether the mobile phone is respectively engaged in a call or not. The signal processor 204 may, alternatively or additionally, combine the audio signals in the two input signals in any other known way, such as e.g. by adding them.

[0043]    FIG. 3 illustrates upsampling of a digital input signal $x(n)$ using a polyphase filter as it may be comprised by embodiments of the invention, such as the ones shown in FIGs. 1 and 2. The sample processor 103 and/or the input buffer 102 may thus comprise the elements shown in FIG. 3. The use of polyphase filters for upsampling is well known in the art, and so are its advantages with respect to efficiency and ease of implementation (see e.g. [1], chapter 10.5.2). When a sample of the digital audio signal arrives from the transmitter 108, the receiver 101 may perform error correction, request retransmission of lost samples or packets of samples and/or discard duplicate samples or packets of samples in order to provide a digital input signal $x(n)$ with as few errors and discontinuities as possible.

[0044]    The digital input signal $x(n)$ has a sample rate $F_x$ that may vary over time. The sample rate $F_x$ may be directly reflected in the arrival times for the individual samples of the digital audio signal at the receiver 101 if the samples are transmitted continuously. If, however, the transmission from the transmitter 108 to the hearing device 100, 200 is packet-based, the exact instantaneous sample rate $F_x$ is undefined. In the latter case, or in the case that the samples arrive irregularly due to transmission errors or other causes, references in the following to the sample rate $F_x$ should be interpreted to include average values over an appropriate number of samples, e.g. over one or more packets.

[0045]    The individual samples of the digital input signal $x(n)$ are written in parallel to the inputs of a number U of upsampling phase filters $P_u$, $u = 0..U-1$, which each represents a specific phase of an interpolation filter Hu. The number U of upsampling phase filters $P_u$ equals the upsampling factor. The interpolation filter Hu is a conventional finite-impulse-response (FIR) filter of length J, and its coefficients hu(j) are computed such that the interpolation filter Hu would suppress frequencies above half the sample rate $F_x$ and pass frequencies below unaltered if applied to a digital signal. If the filter length J is initially not an integer multiple of U, then the interpolation filter Hu is padded with zero coefficients to achieve this. The coefficients $p_u(i)$ of the upsampling phase filters $P_u$, which are FIR filters of length J/U, are computed from $p_u(i) = hu(u+i \cdot U)$ for $u = 0..U-1$ and $i = 0..J/U-1$. The outputs of the upsampling phase filters $P_u$ are cyclically resampled at the resampling rate $F_z = U \cdot F_x$ in order of increasing u to provide an upsampled signal $z(r)$.

[0046]    FIG. 4 illustrates downsampling of a digital signal, such as the upsampled signal $z(r)$ provided in FIG. 3, using a polyphase filter as it may be comprised by embodiments of the invention, such as the ones shown in FIGs. 1 and 2. The sample processor 103 may thus comprise the elements shown in FIG. 4. The use of polyphase filters for downsam-pling is well known in the art, and so are its advantages with respect to efficiency and ease of implementation (see e.g. [1], chapter 10.5.2).

[0047]    The individual samples of the upsampled signal $z(r)$ are written cyclically to the inputs of a number D of down-sampling phase filters $Q_d$, $d = 0..D-1$, in order of decreasing d. The number D of downsampling phase filters $Q_d$ equals the downsampling factor, and each downsampling phase filter $Q_d$ represents a specific phase of a downsampling filter $H_D$. The downsampling filter $H_D$ is a conventional FIR filter of length K. If the filter length K is initially not an integer multiple of D, then the downsampling filter $H_D$ is padded with zero coefficients to achieve this. The coefficients $q_d(i)$ of

the downsampling phase filters $Q_d$, which are FIR filters of length $K/D$, are computed from $q_d(i) = h_D(d+i \cdot D)$ for $d = 0..D-1$ and $i = 0..K/D-1$. The outputs of all the downsampling phase filters $Q_d$ are summed and resampled at the target sample rate $F_y = F_z/D = F_x \cdot U/D$ to provide a sample-rate converted signal that preferably constitutes the digital output signal $y(m)$ of the sample processor 103. The coefficients $h_D(k)$ of the downsampling filter $H_D$ are computed such that the downsampling filter Hu would suppress frequencies above half the sample rate $F_y$ of the digital output signal $y(m)$ and pass frequencies below unaltered if applied to a digital signal.

[0048] In embodiments of the invention, such as the ones shown in FIGs. 1 and 2, sample-rate conversion may be achieved by combining the elements shown in FIGs. 3 and 4. In such embodiments, the sample processor 103 may control the conversion ratio U/D by performing skip and/or repeat operations on the upsampled samples $z(r)$ and/or by changing the upsampling and/or downsampling factor U, D. Preferably, the sample processor 103 works in one mode, wherein it performs skip and/or repeat operations, as long as this suffices to minimise the difference between the estimated average or smoothed input-buffer level $I(t)$ and the target buffer level L, and works in another mode, wherein it only changes the upsampling and/or the downsampling factor U, D, when said difference is too large or changing too fast to allow minimising by skip and/or repeat operations without either risking multiple skip or multiple repeat operations on consecutive samples $z(r)$ or risking a too large deviation from the target buffer level L, e.g. a buffer over-run or a buffer under-run. The ratio controller 105 controls the working mode of the sample processor 103 in dependence on said difference and/or on the speed of variation of said difference, e.g. by comparing one or both of these to predetermined thresholds. The upsampling and downsampling described above and possible implementations are discussed in detail in [1], chapter 10.5.2.

[0049] Alternatively, and preferably, sample-rate conversion based on a time-variant filter structure as disclosed in [1], chapter 10.5.3 may be used and may thus be comprised by embodiments of the invention, preferably by the sample processor 103 and/or the input buffer 102 described above. Such sample-rate conversion is illustrated in FIG. 5 and may preferably be based on the following equation:

$$EQ1: \quad y(m) = \sum_{i=0}^{\frac{J}{U}-1} g\left(i - \left\lfloor \frac{m}{U} \right\rfloor U\right) \cdot x\left(\left\lfloor \frac{mD}{U} \right\rfloor - i\right)$$

wherein U and D are up- and downsampling factors as described above, and the operator $\lfloor R \rfloor$ provides the integer portion of the number R such that the remainder is always positive. EQ1 corresponds to equation (10.5.8) from [1], however with slightly deviating letter symbols. The meanings of the remaining letter symbols are explained below. EQ1 is preferably combined with equation EQ2, which corresponds to equation (10.5.7) from [1]:

$$EQ2: \quad g(a,b) = h(aU - (bD)_U)$$

wherein the function $g(a,b)$ defines a time-variant filter and $(bD)_U$ provides bD modulo U, i.e. the positive remainder of bD after division with U. EQ1 may thus be rewritten as:

$$EQ3: \quad y(m) = \sum_{i=0}^{\frac{J}{U}-1} h(iU - (mD)_U) \cdot x\left(\left\lfloor \frac{mD}{U} \right\rfloor - i\right)$$

[0050] Thus, up- and downsampling as well as the accompanying filtering to prevent aliasing may be accomplished by buffering samples of the digital input signal $x(n)$ and for each sample to provide of the digital output signal $y(m)$, preferably at the target sampling rate $F_y$, computing the sample $y(m)$ as a scalar product of a dynamically selected subset of the buffered samples $x(n)$ and a dynamically selected subset of filter coefficients $h(j)$ of an FIR filter H of length J. The filter coefficients $h(j)$ of the filter H are preferably computed such that the filter H would suppress frequencies above the lower one of the cut-off frequencies of respectively the interpolation filter Hu and the downsampling filter $H_D$ described further above. The filter H would thus suppress frequencies above half the minimum of the input sample rate $F_x$ and the target sampling rate $F_y$ and pass frequencies below unaltered if applied to a digital signal. If J is initially not an integer multiple of the upsampling factor U, then the filter H is padded with zero coefficients to achieve this. The filter coefficients $h(j)$ are preferably stored in a coefficient table 501, which may be comprised by the sample processor 103.

[0051] The high efficiency of this sample-rate conversion is partly due to the fact that only J/U multiplications need be

performed for each output sample y(m), namely the J/U products between respective elements of the two subsets. Since each of the indices i and m is preferably traversed in steps of unity, the product iU may be computed by successively adding U, and the term $(mD)_U$ may be computed by successively adding D and subtracting U whenever the sum exceeds U. Similarly, the integer portion of the fraction mD/U may be computed by successively adding D, subtracting U when the sum exceeds U, and increasing a counter for each made subtraction; the value of the counter yields said integer portion. Thus, the actual subsets to be used in the scalar product may be selected dynamically from respectively an input buffer 102 and the predetermined set of filter coefficients h(j) in the coefficient table 501, using respective indices that are dynamically updated using only addition, subtraction and comparison operations. The computations may be made in an arithmetic unit 502, which may be comprised by the sample processor 103. Further details and possible implementations are discussed in detail in [1], chapter 10.5.3. In particular, preferred embodiments include those disclosed in [1], chapter 10.5.3, such as e.g. the block processing algorithm visualised in [1], Figure 10.16, the filter structure disclosed in [1], Figure 10.17 and the embodiment with U polyphase filters disclosed in the last part of the chapter.

[0052]    Prior to the buffering of received samples, error correction etc. as described above for FIG. 3, may be performed, e.g. by the receiver 101, and the observations as to the meaning of the sample rate $F_x$ of the digital input signal x(n) are also the same as for FIG. 3.

[0053]    The sample processor 103 may control the conversion ratio by changing the upsampling and/or downsampling factor U, D. Since, however, the upsampled signal z(r) is not directly available in EQ1 or EQ3, nor in the implementations disclosed in [1], performing skip or repeat operations on the upsampled signal z(r) is not possible without further. However, EQ1 and EQ3 as well as the implementations disclosed in [1] may be modified to allow an operation similar to skipping and repeating samples of the upsampled signal z(r). This is achieved by modifying the computation of the index of the digital input signal x(n) in EQ1 and EQ3 such that it changes in dependence on the difference between the estimated average or smoothed input-buffer level (l(t)) and the target buffer level (L) as shown in:

$$EQ4: \quad y(m) = \sum_{i=0}^{\frac{J}{U}-1} h(iU - (mD)_U) \cdot x\left(\left\lfloor \frac{mD}{U} \right\rfloor + S(m) - i\right)$$

which corresponds to EQ3, except that an integer primary sample index modifier S(m) is added to the index to x(n). The primary sample index modifier S(m) is preferably updated for each output sample y(m) to compute. Increasing or decreasing the primary sample index modifier S(m) before computation of an output sample y(m) causes the time-variant filter to operate on a subset of the buffered samples x(n) that is shifted one sample respectively forwards or backwards in time. The effect hereof is equal to the effect of respectively a skip or a repeat operation on the digital input signal x(n) as described further above in the respect that it temporarily modifies the conversion ratio. Since, however, the entire subset is shifted in time in EQ4, the digital output signal y(m) has less pronounced artefacts than a corresponding digital output signal y(m) obtained by sample-rate conversion using simple skip or repeat operations on the digital input signal x(n); the level of artefacts produced is in fact comparable to the level produced when using simple skip or repeat operations on an upsampled signal z(r).

[0054]    The ratio controller 105 may thus control the conversion ratio in the sample processor 103 by causing a change of the primary sample index modifier S(m). The ratio controller 105 may provide the primary sample index modifier S(m) as a control parameter to the sample processor 103. Alternatively, the ratio controller 105 may provide a secondary sample index modifier s(m) as a control parameter to the sample processor 103, and the sample processor 103 may explicitly or implicitly compute the primary sample index modifier S(m) in dependence on the secondary sample index modifier s(m), e.g. according to:

$$EQ5: \quad y(m) = \sum_{i=0}^{\frac{J}{U}-1} h(iU - (mD)_U) \cdot x\left(\left\lfloor \frac{mD + \sum s(m)}{U} \right\rfloor - i\right)$$

where $\sum s(m)$ provides the accumulated sum of the secondary sample index modifier s(m) over time. Note that EQ5 may be transformed into EQ4 by expressing the primary sample index modifier S(m) as a function of the secondary sample index modifier s(m), m, D and U. The secondary sample index modifier s(m) may be an integer function, but is preferably a fractional function in order to allow for a more even distribution in time of shifts in the subset of the buffered samples x(n). The value of the secondary sample index modifier s(m) for a specific index m thus indicates a desired time shift in the sample-rate conversion in the form of a fraction s(m)/U of an input sample interval $1/F_x$, whereas the primary sample

index modifier S(m) more or less corresponds to an integral over time of the secondary sample index modifier s(m) and thus indicates a sum of all preceding desired time shifts. Both for the primary sample index modifier S(m) and for the secondary sample index modifier s(m), the index to x(n) may be computed merely by adding, subtracting and comparing values as described further above.

**[0055]** The ratio controller 105 may compute or update the primary or the secondary sample index modifier S(m), s(m) once for each output sample y(m) or at larger intervals depending on the required speed of controlling the conversion ratio. In some embodiments, the ratio controller 105 may compute or update the primary or the secondary sample index modifier S(m), s(m) during computation of an output sample y(m), such that the primary or the secondary sample index modifier S(m), s(m) becomes in effect a function of both indices m and i. In this case, less than the entire subset is shifted in time for the particular output sample y(m), and the artefacts produced are typically more pronounced. Nevertheless, such embodiments may be preferable in some situations.

**[0056]** The person skilled in the art should be readily able to apply the above described modifications according to EQ4 or EQ5 to the implementations disclosed in [1], chapter 10.5.3, such as e.g. the block processing algorithm visualised in [1], Figure 10.16, the filter structure disclosed in [1], Figure 10.17 and the embodiment with U polyphase filters disclosed in the last portion of the chapter. In the filter structure disclosed in [1], Figure 10.17, the modifications may comprise increasing the length of the shift register and providing a multiplexer before each hold-and-sample device that allows the hold-and-sample device to select its input from any one of a number of consecutive memory cells in the shift register. In the case that the sample-rate conversion is implemented in software, the required modifications are rather straight forward.

**[0057]** In some embodiments, the sample processor 103 works in one mode, wherein it performs the above described time shift operations, as long as this suffices to minimise the difference between the estimated average or smoothed input-buffer level l(t) and the target buffer level L, and works in another mode, wherein it only changes the upsampling and/or the downsampling factor U, D, when said difference is too large or changing too fast to allow minimising by the above described time shift operations without risking a too large deviation from the target buffer level L, e.g. a buffer over-run or a buffer under-run. The ratio controller 105 controls the working mode of the sample processor 103 in dependence on said difference and/or on the speed of variation of said difference, e.g. by comparing one or both of these to predetermined thresholds.

**[0058]** In any of the above described embodiments of sample-rate conversion, the ratio controller 105 may provide one or more control parameters, such as e.g. the primary or the secondary sample index modifier S(m), s(m) or any other suitable control parameter, e.g. a target conversion ratio or a desired shift in the current conversion ratio. The ratio controller 105 preferably computes or updates the one or more control parameters repeatedly, e.g. once for every output sample y(m) or any other intervals, such as e.g. larger or smaller intervals, depending on the required speed of controlling the conversion ratio. The ratio controller 105 may preferably compute or update the one or more control parameters by means of the well known proportional-derivative control algorithm or the well known proportional-integral-derivative control algorithm.

**[0059]** To this end, the ratio controller 105 may preferably compute the difference e(k) between the average or smoothed input-buffer level l(t) received from the estimator 104 and the target buffer level L, which may e.g. be a fixed value stored in the ratio controller 105 or a varying value received from the latency controller 109, and use this difference e(k) as an error input to an error-minimising control algorithm that computes consecutive values of a control parameter w(k). In a proportional-integral-derivative control, the control parameter w(k) is iteratively computed according to:

$$EQ6: \quad w(k) = w(k-1) + K_p \cdot e(k) + K_d \cdot \frac{de}{dk} + K_i \cdot \int e\, dk$$

wherein $K_p$, $K_d$ and $K_i$ are predetermined constants, which may be determined in advance by experimentation or analytically from knowledge about the expected behaviour of the digital input signal x(n). In a proportional- derivative control $K_i$ is set to zero. In embodiments providing the secondary sample index modifier s(m) as a control parameter, EQ6 may be rewritten as:

$$EQ7: \quad s(m) = s(m-1) + K_p \cdot e(k) + K_d \cdot \frac{de}{dm} + K_i \cdot \int e\, dk$$

**[0060]** In addition or alternatively, the ratio controller 105 may use other known types of control algorithms, such as e.g. lead, lag or lead-lag compensators.

**[0061]** In any of the above described embodiments of sample-rate conversion, the estimator 104 may preferably provide the average or smoothed input-buffer level l(t) as illustrated in FIG. 6. The curve 601 illustrates an example of

the variation over time t of the actual buffer level N. The input buffer 102 may have a lower threshold 602 and an upper threshold 603 which may equal respectively zero and the buffer size, or alternatively lie away from these extremes in order to allow overshoot of the control algorithm applied by the ratio controller 105. In the shown example, the receiver 101 feeds packets of samples x(n) into the input buffer 102, which causes the curve 601 to increase by the packet size, e.g. during the write operation 604. The packets arrive irregularly due to transmission errors. The sample processor 103 reads a number N of buffered samples x(n) at regular intervals, which causes the curve 601 to decrease by the number of samples read, e.g. during the read operation 605.

[0062] The time line 606 is divided into three time periods, wherein vertical dotted lines illustrate different methods for determining an actual input-buffer level N. In the first period 607, the actual buffer level N is sampled every time a read operation 605 by the sample processor 103 occurs. In the second period 608, the actual buffer level N is sampled every time a write operation 604 by the receiver 101 occurs. In the third period 609, the actual buffer level N is sampled at regular time intervals which are preferably substantially shorter than the intervals between successive write operations 604 and substantially shorter than the intervals between successive read operations 605. The estimator may use any of the illustrated sample methods, or any combination thereof, to provide input data to a smoothing or averaging algorithm for providing the average or smoothed input-buffer level l(t).

[0063] The estimator 104 may preferably compute an average over a sliding time window of the sample results or, alternatively or additionally, low-pass filter the sample results to obtain the average or smoothed input-buffer level l(t). Due to the different sampling times relative to upwards and downwards shifts in the actual buffer level N, the illustrated sample methods typically cause the smoothing or averaging algorithm to provide different results, and the estimator 104 may therefore preferably subtract from the output of the average or low-pass filter a constant value that reflects a typical estimation error for the implemented sampling method before providing the average or smoothed input-buffer level l(t) to the ratio controller 105. The width of the time window, the characteristics of the low-pass filter and/or typical estimation errors may be determined in advance by experimentation or analytically from knowledge of the packet sizes, of the number of samples read in batch by the sample processor 105, of expected variations in the arrival times of the digital input signal x(n), and of the characteristics of the hardware and/or software involved in the sample-rate conversion.

[0064] In any of the above described embodiments of sample-rate conversion, the latency controller 109 may preferably provide the target buffer level L adaptively as illustrated in FIG. 7. As in FIG. 6, the curve 601 illustrates an example of the variation over time t of the actual buffer level N, and the input buffer 102 may have a lower threshold 602 and an upper threshold 603 as described above. Again, the sample processor 103 reads a number of buffered samples x(n) at regular intervals, e.g. during the read operation 605, and the receiver 101 feeds packets of samples x(n) into the input buffer 102, e.g. during the write operation 604. In the shown example, the packets arrive irregularly due to a decrease over time of the quality of reception of the digital audio signal. Thus, in the leftmost part of the figure, the difference between the local minimum 701 and the local maximum 702 of the actual buffer level N is relatively small, whereas in the rightmost part, the difference between the local minimum 703 and the local maximum 704 of the actual buffer level N is relatively large. The average or smoothed input-buffer level l(t) provided by the estimator 104 is also shown.

[0065] The latency controller 109 may preferably control the target buffer level L in dependence on the estimated quality of reception of the digital audio signal in order to achieve a short latency when the quality of reception is high and a long latency when the quality of reception is low. The latency is correlated with the target buffer level L because the ratio controller 105 works to minimise the difference e(k) between the estimated average or smoothed input-buffer level l(t) and the target buffer level L, and the latency controller 109 may thus preferably decrease the target buffer level L when the estimated quality of reception is high and increase the target buffer level L when the estimated quality of reception is low.

[0066] In some embodiments, the latency controller 109 may achieve control of the target buffer level L in dependence on the estimated quality of reception of the digital audio signal in an indirect manner, e.g. by determining local minima 701, 703 in the actual buffer level N and controlling the target buffer level L in dependence on the determined local minima 701, 703. The latency controller 109 may e.g. increase the target buffer level L whenever a determined local minimum 701, 703 is below a predetermined threshold 705 and iteratively decrease the target buffer level L as long as the determined local minima 701, 703 are above the predetermined threshold 705. The predetermined threshold 705 may equal the lower threshold 602 or may be higher in order to allow overshoot of the latency control algorithm without risking buffer under-run. The latency controller 109 may e.g. immediately increase the target buffer level L by an amount equal to the amount which a determined local minimum 701, 703 is below the predetermined threshold 705 and subsequently decrease the target buffer level L, e.g. exponentially, towards a minimum target buffer level $L_{min}$, e.g. with a time constant between 1 and 3 s, between 2 and 6 s or between 3 and 10 s. Alternatively, the latency controller 109 may track the local minima 701, 703 and employ the above described increases and decreases of the target buffer level L in dependence on a low-pass filtered or otherwise smoothed minimum track value.

[0067] Preferably, the latency controller 109 may further determine local maxima 702, 704 in the actual buffer level N and control the target buffer level L in dependence on both the determined local minima 701, 703 and the determined local maxima 702, 704. For instance, the latency controller 109 may increase the time constant for decreasing the target

buffer level L in dependence on the difference between the determined local maxima 702, 704 and the determined local minima 701, 703 increasing. Alternatively, or additionally, the latency controller 109 may increase the target buffer level L in dependence on the difference between the determined local maxima 702, 704 and the determined local minima 701, 703 increasing. The latter could e.g. be accomplished by adding a safety margin value to the target buffer level L before providing it to the ratio controller 105, wherein the safety margin value depends on said difference. The time constant and/or the safety margin value may e.g. be proportional to said difference or otherwise increase when said difference increases and vice versa. The latency controller 109 may determine the difference between local maxima 702, 704 and local minima 701, 703 by tracking the local minima 701, 703 and the local maxima 702, 704 and determine said difference in dependence on low-pass filtered or otherwise smoothed minimum and maximum track values. Instead of using the difference between the determined local maxima 702, 704 and the determined local minima 701, 703 in the above computations, the latency controller 109 may use the difference between the average or smoothed input-buffer level l(t) provided by the estimator 104 and the determined local minima 701, 703, since this difference is highly correlated with the former difference when the ratio controller 105 works as described further above.

[0068] The latency controller 109 may use any known methods and/or means to track or otherwise determine the development over time of the local minima 701, 703 and/or the local maxima 702, 704 and may control the target buffer level L in dependence on the determined development. The latency controller 109 may receive a quality signal from the receiver 101 indicating e.g. the frequency of transmission errors, lost packets and/or packet retransmissions etc., and the latency controller 109 may estimate the quality of reception of the digital audio signal based on the quality signal alone or on the quality signal in combination with the methods described above using the actual buffer level N. In any embodiment, the latency controller 109 may further low-pass filter the target buffer level L with an appropriate cut-off frequency before providing it to the ratio controller 105 in order to avoid abrupt changes of the conversion ratio.

[0069] In any embodiment, the hearing device 100, 200 is preferably implemented using mainly digital circuits operating in the discrete time domain, but some parts hereof may alternatively be implemented as analog circuits operating in the continuous time domain. Digital functional blocks of the hearing device 100, 200, such e.g. the functional blocks shown in the drawings, may be implemented in any suitable combination of hardware, firmware and software and/or in any suitable combination of hardware units. Furthermore, any single hardware unit may execute the operations of several functional blocks in parallel or in interleaved sequence and/or in any suitable combination thereof.

[0070] In any embodiment, the input buffer 102 may preferably be implemented as a ring buffer or as a first-in-first-out (FIFO) buffer.

[0071] The hearing device 100, 200 may be part of a binaural hearing system, and the hearing device 100, 200 may e.g. receive the digital audio signal from the other hearing device in the binaural hearing system. The binaural hearing system may comprise one or more auxiliary devices.

[0072] Embodiments of the invention may be used in any type of device, and most advantageously in battery-driven and/or portable devices.

[0073] Some preferred embodiments have been described in the foregoing, but it should be stressed that the invention is not limited to these, but may be embodied in other ways within the subject-matter defined in the following claims. For example, the features of the described embodiments may be combined arbitrarily, e.g. in order to adapt the system, the devices and/or the method according to the invention to specific requirements.

[0074] It is further intended that the structural features of the system and/or devices described above, in the detailed description of 'mode(s) for carrying out the invention' and in the claims can be combined with the methods, when appropriately substituted by a corresponding process. Embodiments of the methods have the same advantages as the corresponding systems and/or devices.

[0075] Any reference numerals and names in the claims are intended to be non-limiting for their scope.

REFERENCES

[0076]

[1] Digital Signal Processing; John G. Proakis and Dimitris K Manolakis; Prentice Hall; 4th edition (April 7, 2006); ISBN 978-0131873742

**Claims**

1. A method for operating a hearing device (100, 200), the method comprising:

   • receiving samples of a digital audio signal;
   • feeding received samples as a digital input signal (x(n)) to an input buffer (102);

• processing the buffered samples (x(n)) to provide samples of a digital output signal (y(m)) such that the digital output signal (y(m)) is a sample-rate converted representation of the digital input signal (x(n)) with a predetermined target sample rate ($F_y$);

• estimating an average or smoothed input-buffer level (l(t)) indicating the number (N) of unprocessed samples (x(n)) in the input buffer (102); and

• controlling the processing of the buffered samples (x(n)) in a manner aiming at minimising the difference (e(k)) between the estimated average or smoothed input-buffer level (l(t)) and a target buffer level (L);

• comparing a parameter representative of a difference between the estimated average or smoothed input buffer level and the target buffer level (L), or a speed of variation in the difference between the average input buffer level and the target level (L), to a predetermined threshold

• controlling a conversion ratio between the input rate and the sample rate according to one of a first mode and a second mode, which is selected according to a result of the comparison between the parameter and the predetermined threshold, wherein:

• in the first mode, the conversion ratio is controlled by performing skip and/or repeat operation in order to minimise the difference between the average input buffer level and the target buffer level,

• in the second mode, the conversion ratio is controlled by changing an upsampling factor and/or downsampling factor;

• the first mode is selected for controlling the conversion ratio when the parameter is below the predetermined threshold, and

• the second mode is selected for controlling the conversion rate when the parameter exceeds the predetermined threshold thus indicating that the difference between the average input buffer level and the target buffer level is too large or changing too fast allow minimizing the difference by skip and/or repeat operations.

2.  A method according to claim 1, wherein said processing comprises upsampling the buffered samples (x(n)) to provide an upsampled signal (z(r)) and downsampling the upsampled signal (z(r)), and wherein controlling said processing comprises performing operations on the upsampled signal (z(r)).

3.  A method according to claim 2, wherein said upsampling and/or said downsampling comprises the use of a polyphase filter ($P_u$, $Q_d$).

4.  A method according to any preceding claim, wherein said processing comprises computing each sample of the output signal (y(m)) as a scalar product of a subset of the buffered samples (x(n)) and a subset of filter coefficients (h(j)), wherein said subsets are selected anew for each computed sample (y(m)) from respectively the input buffer (102) and a set of predetermined coefficients (h(j)), and wherein the selection of said subset of the buffered samples (x(n)) is made in dependence on the estimated average or smoothed input-buffer level (l(t)).

5.  A method according to any preceding claim, wherein controlling the processing of the buffered samples (x(n)) comprises proportional-derivative control or proportional-integral-derivative control.

6.  A method according to any preceding claim and further comprising increasing the target buffer level (L) in dependence on an estimated quality of reception of the digital audio signal decreasing and vice versa.

7.  A method according to claim 6, wherein said quality of reception is estimated on the basis of repeated determinations of the number (N) of unprocessed samples (x(n)) in the input buffer (102).

8.  A hearing device (100, 200) comprising a receiver (101), an input buffer (102) and a sample processor (103), the receiver (101) being adapted to receive samples of a digital audio signal and feed received samples as a digital input signal (x(n)) to the input buffer (102), the sample processor (103) being adapted to process the buffered samples (x(n)) to provide samples of a digital output signal (y(m)) such that the digital output signal (y(m)) is a sample-rate converted representation of the digital input signal (x(n)) with a predetermined target sample rate ($F_y$), **characterised in that** the hearing device (100, 200) further comprises:

• an estimator (104) adapted to estimate an average or smoothed input-buffer level (l(t)) indicating the number (N) of unprocessed samples (x(n)) in the input buffer (102);

• a ratio controller (105) adapted to control the sample processor (103) in a manner aiming at minimising the difference (e(k)) between the estimated average or smoothed input-buffer level (l(t)) and a target buffer level (L);

• a comparator configured to compare a parameter representative of a difference between the estimated average or smoothed input buffer level and the target buffer level (L), or a speed of variation in the difference between

the average input buffer level and the target level (L), to a predetermined threshold; and
• a ratio controller configured to control a conversion ratio between the input rate and the sample rate according to one of a first mode and a second mode, which is selected according to a result of the comparison between the parameter and the predetermined threshold,
wherein in the first mode, the conversion ratio is controlled by performing skip and/or repeat operation in order to minimise the difference between the average input buffer level and the target buffer level and in the second mode, the conversion ratio is controlled by changing an upsampling factor and/or downsampling factor;
wherein the first mode is selected for controlling the conversion ratio when the parameter is below the predetermined threshold, and the second mode is selected for controlling the conversion rate when the parameter exceeds the predetermined threshold thus indicating that the difference between the average input buffer level and the target buffer level is too large or changing too fast allow minimizing the difference by skip and/or repeat operations.

9.  A hearing device according to claim 8, wherein the sample processor (103) is further adapted to upsample the buffered samples (x(n)) to provide an upsampled signal (z(r)) and to downsample the upsampled signal (z(r)), and wherein the ratio controller (105) is further adapted to cause the sample processor (103) to perform operations on the upsampled signal (z(r)).

10. A hearing device according to claim 9, wherein the sample processor (103) comprises a polyphase filter ($P_u$, $Q_d$) for said upsampling and/or downsampling.

11. A hearing device according to any of claims 8-10, wherein the sample processor (103) is further adapted to compute each sample of the output signal (y(m)) as a scalar product of a subset of the buffered samples (x(n)) and a subset of filter coefficients (h(j)), to select said subsets anew for each computed sample (y(m)) from respectively the input buffer (102) and a set of predetermined coefficients (h(j)), and to select said subset of the buffered samples (x(n)) in dependence on the estimated average or smoothed input-buffer level (l(t)).

12. A hearing device according to claim 11, wherein the sample processor (103) is further adapted to shift said subset of the buffered samples (x(n)) in time in dependence on the estimated average or smoothed input-buffer level (l(t)).

13. A hearing device according to any of claims 8-12, wherein the ratio controller (105) comprises a proportional-derivative controller or a proportional-integral-derivative controller.

14. A hearing device according to any of claim 8-13 and further comprising a latency controller (109) adapted to increase the target buffer level (L) in dependence on an estimated quality of reception of the digital audio signal decreasing and vice versa.

15. A hearing device according to any of claims 8-14, wherein the latency controller (109) is further adapted to estimate said quality of reception on the basis of repeated determinations of the number (N) of unprocessed samples (x(n)) in the input buffer (102).

**Patentansprüche**

1.  Verfahren zum Betrieb eines Hörgeräts (100, 200), wobei das Verfahren umfasst

ein Empfangen von Abtastungen eines Digitalaudiosignals,
ein Einspeisen von empfangenen Abtastungen als ein Digitaleingangssignal (x(n)) in einen Eingangspuffer (102),
ein Verarbeiten der gepufferten Abtastungen (x(n)), um Abtastungen eines Digitalausgangssignals (y(m)) derart bereitzustellen, dass das Digitalausgangssignal (y(m)) eine abtastratenkonvertierte Repräsentation des Digitaleingangssignals (x(n)) mit einer vorbestimmten Sollabtastrate ($F_y$) ist,
ein Schätzen eines durchschnittlichen oder geglätteten Eingangspufferpegels (l(t)), der die Anzahl (N) von unverarbeiteten Abtastungen (x(n)) in dem Eingangspuffer (102) indiziert, und
ein Steuern des Verarbeitens der gepufferten Abtastungen (x(n)) auf eine Weise, die auf ein Minimieren der Differenz (e(k)) zwischen dem geschätzten durchschnittlichen oder geglätteten Eingangspufferpegel (l(t)) und einem Sollpufferpegel (L) abzielt,
ein Vergleichen eines Parameters, der eine Differenz zwischen dem geschätzten durchschnittlichen oder geglätteten Eingangspufferpegel und dem Sollpufferpegel (L) oder eine Variationsgeschwindigkeit der Differenz

zwischen dem durchschnittlichen Eingangspufferpegel und dem Sollpegel (L) repräsentiert, mit einer vorbestimmten Schwelle,

ein Steuern eines Umwandlungsverhältnisses zwischen der Eingangsrate und der Abtastrate gemäß einer aus einer ersten Betriebsart und einer zweiten Betriebsart, die gemäß einem Ergebnis des Vergleichs zwischen dem Parameter und der vorbestimmten Schwelle ausgewählt ist, wobei

bei der ersten Betriebsart das Umwandlungsverhältnis durch Durchführen eines Weglass- und/oder Wiederholbetriebs zur Minimierung der Differenz zwischen dem durchschnittlichen Eingangspufferpegel und dem Sollpufferpegel gesteuert wird,

bei der zweiten Betriebsart das Umwandlungsverhältnis durch Ändern eines Hochabtastfaktors und/oder eines Runterabtastfaktors gesteuert wird,

die erste Betriebsart für ein Steuern des Umwandlungsverhältnisses ausgewählt wird, wenn der Parameter unterhalb der vorbestimmten Schwelle ist, und

die zweite Betriebsart zur Steuerung des Umwandlungsverhältnisses ausgewählt wird, wenn der Parameter die vorbestimmte Schwelle überschreitet und dadurch indiziert, dass die Differenz zwischen dem durchschnittlichen Eingangspufferpegel und dem Sollpufferpegel zu groß oder eine Änderung zu schnell ist für eine Minimierung der Differenz durch Weglass- und/oder Wiederholbetriebe.

2. Verfahren nach Anspruch 1, wobei das Verarbeiten ein Hochabtasten der gepufferten Abtastungen (x(n)) zur Bereitstellung eines hochabgetasteten Signals (z(r)) und ein Runterabtasten des hochabgetasteten Signals (z(r)) umfasst, und wobei das Steuern des Verarbeitens ein Durchführen von Betrieben bezüglich des hochabgetasteten Signals (z(r)) umfasst.

3. Verfahren nach Anspruch 2, wobei das Hochabtasten und/oder das Runterabtasten die Verwendung eines Polyphasenfilters ($P_u$, $Q_d$) umfasst.

4. Verfahren nach einem der vorstehenden Ansprüche, wobei das Verarbeiten ein Berechnen jeder Abtastung des Ausgangssignals (y(m)) als ein Skalarprodukt eines Untersatzes der gepufferten Abtastungen (x(n)) und eines Untersatzes von Filterkoeffizienten (h(j)) umfasst, wobei die Untersätze neu für jede berechnete Abtastung (y(m)) aus jeweils dem Eingangspuffer (102) und einem Satz von vorbestimmten Koeffizienten (h(j)) ausgewählt werden, und wobei die Auswahl des Untersatzes der gepufferten Abtastungen (x(n)) abhängig von dem geschätzten durchschnittlichen oder geglätteten Eingangspufferpegel (l(t)) ist.

5. Verfahren nach einem der vorstehenden Ansprüche, wobei das Steuern des Verarbeitens der gepufferten Abtastungen (x(n)) eine Proportionalableitungssteuerung oder eine Proportionalintegralableitungssteuerung umfasst.

6. Verfahren nach einem der vorstehenden Ansprüche, ferner mit einem Erhöhen des Sollpufferpegels (L) in Abhängigkeit von einer Verringerung einer geschätzten Qualität eines Empfangs des Digitalaudiosignals und andersherum.

7. Verfahren nach Anspruch 6, wobei die Qualität des Empfangs basierend auf wiederholten Bestimmungen der Anzahl (N) von unverarbeiteten Abtastungen (x(n)) in dem Eingangspuffer (102) geschätzt wird.

8. Hörgerät (100, 200) mit einem Empfänger (101), einem Eingangspuffer (102), und einem Abtastprozessor (103), wobei der Empfänger (101) dazu eingerichtet ist, um Abtastungen eines Digitalaudiosignals zu empfangen und empfangene Abtastungen als ein Digitaleingangssignal (x(n)) zu dem Eingangspuffer (102) einzuspeisen, wobei der Abtastprozessor (103) dazu eingerichtet ist, um die gepufferten Abtastungen (x(n)) zu verarbeiten, um Abtastungen eines Digitalausgangssignals (y(m)) derart bereitzustellen, dass das Digitalausgangssignal (y(m)) eine abtastratenkonvertierte Repräsentation des Digitaleingangssignals (x(n)) mit einer vorbestimmten Sollabtastrate ($F_y$) ist,

**dadurch gekennzeichnet, dass** das Hörgerät (100, 200) ferner aufweist

ein Schätzelement (104), das dazu eingerichtet ist, um einen durchschnittlichen oder geglätteten Eingangspufferpegel (l(t)) zu schätzen, der die Anzahl (N) von unverarbeiteten Abtastungen (x(n)) in dem Eingangspuffer (102) indiziert,

ein Verhältnissteuerelement (105), das dazu eingerichtet ist, um den Abtastprozessor (103) auf eine Weise zu steuern, die auf eine Minimierung der Differenz (e(k)) zwischen dem geschätzten durchschnittlichen oder geglätteten Eingangspufferpegel (l(t)) und einem Sollpufferpegel (L) abzielt,

ein Vergleichselement, das dazu eingerichtet ist, um einen Parameter, der eine Differenz zwischen dem geschätzten durchschnittlichen oder geglätteten Eingangspufferpegel und dem Sollpufferpegel (L) oder eine Va-

riationsgeschwindigkeit der Differenz zwischen dem durchschnittlichen Eingangspufferpegel und dem Sollpegel (L) repräsentiert, mit einer vorbestimmten Schwelle zu vergleichen, und

ein Verhältnissteuerelement, das dazu eingerichtet ist, um ein Umwandlungsverhältnis zwischen der Eingangsrate und der Abtastrate gemäß einer aus einer ersten Betriebsart und einer zweiten Betriebsart, die gemäß einem Ergebnis des Vergleichs zwischen dem Parameter und der vorbestimmten Schwelle ausgewählt ist, zu steuern,

wobei bei der ersten Betriebsart das Umwandlungsverhältnis durch Durchführung eines Weglass- und/oder Wiederholbetriebs zur Minimierung der Differenz zwischen dem durchschnittlichen Eingangspufferpegel und dem Sollpufferpegel gesteuert ist, und bei der zweiten Betriebsart das Umwandlungsverhältnis durch Änderung eines Hochabtastfaktors und/oder eines Runterabtastfaktors gesteuert ist,

wobei die erste Betriebsart zur Steuerung des Umwandlungsverhältnisses ausgewählt ist, wenn der Parameter unterhalb der vorbestimmten Schwelle ist, und die zweite Betriebsart zur Steuerung des Umwandlungsverhältnisses ausgewählt ist, wenn der Parameter die vorbestimmte Schwelle überschreitet und dadurch indiziert, dass die Differenz zwischen dem durchschnittlichen Eingangspufferpegel und dem Sollpufferpegel zu groß oder eine Änderung zu schnell ist für eine Minimierung der Differenz durch Weglass- und/oder Wiederholbetriebe.

9. Hörgerät nach Anspruch 8, wobei der Abtastprozessor (103) ferner dazu eingerichtet ist, um die gepufferten Abtastungen (x(n)) hochabzutasten, um ein hochabgetastetes Signal (z(r)) bereitzustellen, und das hochabgetastetes Signal (z(r)) runterabzutasten, und wobei das Verhältnissteuerelement (105) ferner dazu eingerichtet ist, um den Abtastprozessor (103) dazu zu bringen, Betriebe bezüglich des hochabgetasteten Signals (z(r)) durchzuführen.

10. Hörgerät nach Anspruch 9, wobei der Abtastprozessor (103) einen Polyphasenfilter ($P_u$, $Q_d$) für das Hochabtasten und/oder das Runterabtasten umfasst.

11. Hörgerät nach einem der Ansprüche 8 bis 10, wobei der Abtastprozessor (103) ferner dazu eingerichtet ist, um jede Abtastung des Ausgangssignals (y(m)) als ein Skalarprodukt eines Untersatzes der gepufferten Abtastungen (x(n)) und eines Untersatzes von Filterkoeffizienten (h(j)) zu berechnen, die Untersätze neu für jede berechnete Abtastung (y(m)) aus jeweils dem Eingangspuffer (102) und einem Satz von vorbestimmten Koeffizienten (h(j)) auszuwählen, und den Untersatz der gepufferten Abtastungen (x(n)) in Abhängigkeit von dem geschätzten durchschnittlichen oder geglätteten Eingangspufferpegel (l(t)) auszuwählen.

12. Hörgerät nach Anspruch 11, wobei der Abtastprozessor (103) ferner dazu eingerichtet ist, um den Untersatz der gepufferten Abtastungen (x(n)) in Abhängigkeit von dem geschätzten durchschnittlichen oder geglätteten Eingangspufferpegel (l(t)) in der Zeit zu verschieben.

13. Hörgerät nach einem der Ansprüche 8 bis 12, wobei das Verhältnissteuerelement (105) ein Proportionalableitungssteuerelement oder ein Proportionalintegralableitungssteuerelement umfasst.

14. Hörgerät nach einem der Ansprüche 8 bis 13, ferner mit einem Latenzsteuerelement (109), das dazu eingerichtet ist, um den Sollpufferpegel (L) in Abhängigkeit von einer Verringerung einer geschätzten Qualität eines Empfangs des Digitalaudiosignals zu erhöhen und andersherum.

15. Hörgerät nach einem der Ansprüche 8 bis 14, wobei das Latenzsteuerelement (109) ferner dazu eingerichtet ist, um die Qualität des Empfangs basierend auf wiederholten Bestimmungen der Anzahl (N) von unverarbeiteten Abtastungen (x(n)) in dem Eingangspuffer (102) zu schätzen.

## Revendications

1. Procédé de fonctionnement d'un dispositif auditif (100, 200), le procédé comprenant les étapes consistant à :

   • recevoir des échantillons d'un signal audio numérique ;
   • alimenter un tampon d'entrée (102) en échantillons reçus comme signal d'entrée numérique (x(n)) ;
   • traiter les échantillons tamponnés (x(n)) pour fournir des échantillons d'un signal de sortie numérique (y(m)) de sorte que le signal de sortie numérique (y(m)) soit une représentation convertie en taux d'échantillonnage du signal d'entrée numérique (x(n)) avec un taux d'échantillonnage cible prédéterminé ($F_y$) ;
   • estimer un niveau de tampon d'entrée moyen ou lissé (l(t)) indiquant le nombre (N) d'échantillons non traités

(x(n)) dans le tampon d'entrée (102) ; et

• contrôler le traitement des échantillons tamponnés (x(n)) de manière à minimiser la différence (e(k)) entre le niveau de tampon d'entrée moyen ou lissé (I(t)) estimé et un niveau de tampon cible (L) ;

• comparer un paramètre représentatif d'une différence entre le niveau de tampon d'entrée moyen ou lissé estimé et le niveau de tampon cible (L), ou une vitesse de variation de la différence entre le niveau de tampon d'entrée moyen et le niveau cible (L), à un seuil prédéterminé ;

• contrôler un rapport de conversion entre le taux d'entrée et le taux d'échantillonnage selon l'un d'un premier mode et d'un second mode, qui est sélectionné en fonction d'un résultat de la comparaison entre le paramètre et le seuil prédéterminé, où :

• dans le premier mode, le rapport de conversion est contrôlé en effectuant une opération de saut et/ou de répétition afin de minimiser la différence entre le niveau de tampon d'entrée moyen et le niveau de tampon cible,

• dans le second mode, le rapport de conversion est contrôlé en modifiant un facteur de sur-échantillonnage et/ou un facteur de sous-échantillonnage ;

• le premier mode est sélectionné pour contrôler le rapport de conversion lorsque le paramètre est inférieur au seuil prédéterminé, et

• le second mode est sélectionné pour contrôler le rapport de conversion lorsque le paramètre dépasse le seuil prédéterminé, indiquant ainsi que la différence entre le niveau de tampon d'entrée moyen et le niveau de tampon cible est trop grande ou change trop rapidement pour permettre de minimiser la différence par des opérations de saut et/ou de répétition.

2. Procédé selon la revendication 1, dans lequel ledit traitement comprend un sur-échantillonnage des échantillons tamponnés (x(n)) pour fournir un signal sur-échantillonné (z(r)) et un sous-échantillonnage du signal sur-échantillonné (z(r)), et où le contrôle dudit traitement comprend l'exécution d'opérations sur le signal sur-échantillonné (z(r)).

3. Procédé selon la revendication 2, dans lequel ledit sur-échantillonnage et/ou ledit sous-échantillonnage comprend l'utilisation d'un filtre polyphasé ($P_u$, $Q_d$).

4. Procédé selon l'une quelconque des revendications précédentes, dans lequel ledit traitement comprend un calcul de chaque échantillon du signal de sortie (y(m)) en tant que produit scalaire d'un sous-ensemble des échantillons tamponnés (x(n)) et d'un sous-ensemble de coefficients de filtre (h(j)), où lesdits sous-ensembles sont sélectionnés à nouveau pour chaque échantillon calculé (y(m)) à partir respectivement du tampon d'entrée (102) et d'un ensemble de coefficients prédéterminés (h(j)), et où la sélection dudit sous-ensemble des échantillons tamponnés (x(n)) est faite en fonction du niveau de tampon d'entrée moyen ou lissé (I(t)) estimé.

5. Procédé selon l'une quelconque des revendications précédentes, dans lequel un contrôle du traitement des échantillons tamponnés (x(n)) comprend un contrôle à dérivée proportionnelle ou un contrôle à dérivée proportionnelle-intégrale.

6. Procédé selon l'une quelconque des revendications précédentes, comprenant en outre une augmentation du niveau de tampon cible (L) en fonction d'une qualité estimée de réception du signal audio numérique décroissant et vice versa.

7. Procédé selon la revendication 6, dans lequel ladite qualité de réception est estimée sur la base de déterminations répétées du nombre (N) d'échantillons non traités (x(n)) dans le tampon d'entrée (102).

8. Dispositif auditif (100, 200) comprenant un récepteur (101), un tampon d'entrée (102) et un processeur d'échantillons (103), le récepteur (101) étant conçu pour recevoir des échantillons d'un signal audio numérique et alimenter le tampon d'entrée (102) en échantillons reçus comme signal d'entrée numérique (x(n)), le processeur d'échantillons (103) étant conçu pour traiter les échantillons tamponnés (x(n)) afin de fournir des échantillons d'un signal de sortie numérique (y(m)) de sorte que le signal de sortie numérique (y(m)) soit une représentation convertie en taux d'échantillonnage du signal d'entrée numérique (x(n)) avec un taux d'échantillonnage cible prédéterminé ($F_y$), **caractérisé en ce que** le dispositif auditif (100, 200) comprend en outre :

• un estimateur (104) conçu pour estimer un niveau de tampon d'entrée moyen ou lissé (I(t)) indiquant le nombre (N) d'échantillons non traités (x(n)) dans le tampon d'entrée (102) ;

• un contrôleur de rapport (105) conçu pour contrôler le processeur d'échantillons (103) de manière à minimiser la différence (e(k)) entre le niveau de tampon d'entrée moyen ou lissé (I(t)) estimé et un niveau de tampon cible (L) ;

• un comparateur conçu pour comparer un paramètre représentatif d'une différence entre le niveau de tampon d'entrée moyen estimé ou lissé et le niveau de tampon cible (L), ou une vitesse de variation de la différence entre le niveau de tampon d'entrée moyen et le niveau cible (L), à un seuil prédéterminé ; et

• un contrôleur de rapport conçu pour contrôler un rapport de conversion entre le taux d'entrée et le taux d'échantillonnage selon l'un d'un premier mode et d'un second mode, qui est sélectionné en fonction d'un résultat de la comparaison entre le paramètre et le seuil prédéterminé,

où, dans le premier mode, le rapport de conversion est contrôlé en effectuant une opération de saut et/ou de répétition afin de minimiser la différence entre le niveau de tampon d'entrée moyen et le niveau de tampon cible, et dans le second mode, le rapport de conversion est contrôlé en modifiant un facteur de sur-échantillonnage et/ou un facteur de sous-échantillonnage ;

où le premier mode est sélectionné pour contrôler le rapport de conversion lorsque le paramètre est inférieur au seuil prédéterminé, et le second mode est sélectionné pour contrôler le rapport de conversion lorsque le paramètre dépasse le seuil prédéterminé, indiquant, ainsi que la différence entre le niveau de tampon d'entrée moyen et le niveau de tampon cible est trop grande ou change trop rapidement pour permettre de minimiser la différence par des opérations de saut et/ou de répétition.

9. Dispositif auditif selon la revendication 8, dans lequel le processeur d'échantillons (103) est en outre conçu pour sur-échantillonner les échantillons tamponnés $(x(n))$ afin de fournir un signal sur-échantillonné $(z(r))$ et pour sous-échantillonner le signal sur-échantillonné $(z(r))$, et dans lequel le contrôleur de rapport (105) est en outre conçu pour amener le processeur d'échantillons (103) à effectuer des opérations sur le signal sur-échantillonné $(z(r))$.

10. Dispositif auditif selon la revendication 9, dans lequel le processeur d'échantillons (103) comprend un filtre polyphasé $(P_u, Q_d)$ pour ledit sur-échantillonnage et/ou sous-échantillonnage.

11. Dispositif auditif selon l'une quelconque des revendications 8 à 10, dans lequel le processeur d'échantillons (103) est en outre conçu pour calculer chaque échantillon du signal de sortie $(y(m))$ en tant que produit scalaire d'un sous-ensemble des échantillons tamponnés $(x(n))$ et d'un sous-ensemble de coefficients de filtre $(h(j))$, afin de sélectionner à nouveau lesdits sous-ensembles pour chaque échantillon calculé $(y(m))$ à partir respectivement du tampon d'entrée (102) et d'un ensemble de coefficients prédéterminés $(h(j))$, et de sélectionner ledit sous-ensemble des échantillons tamponnés $(x(n))$ en fonction du niveau de tampon d'entrée moyen ou lissé $(I(t))$ estimé.

12. Dispositif auditif selon la revendication 11, dans lequel le processeur d'échantillons (103) est en outre conçu pour décaler ledit sous-ensemble des échantillons tamponnés $(x(n))$ dans le temps en fonction du niveau de tampon d'entrée moyen ou lissé $(I(t))$ estimé.

13. Dispositif auditif selon l'une quelconque des revendications 8 à 12, dans lequel le contrôleur de rapport (105) comprend un contrôleur à dérivée proportionnelle ou un contrôleur à dérivée proportionnelle-intégrale.

14. Dispositif auditif selon l'une quelconque des revendications 8 à 13 et comprenant en outre un contrôleur de latence (109) conçu pour augmenter le niveau de tampon cible (L) en fonction d'une qualité estimée de réception d'un signal audio numérique décroissant et vice versa.

15. Dispositif auditif selon l'une quelconque des revendications 8 à 14, dans lequel le contrôleur de latence (109) est en outre conçu pour estimer ladite qualité de réception sur la base de déterminations répétées du nombre (N) d'échantillons non traités $(x(n))$ dans le tampon d'entrée (102).
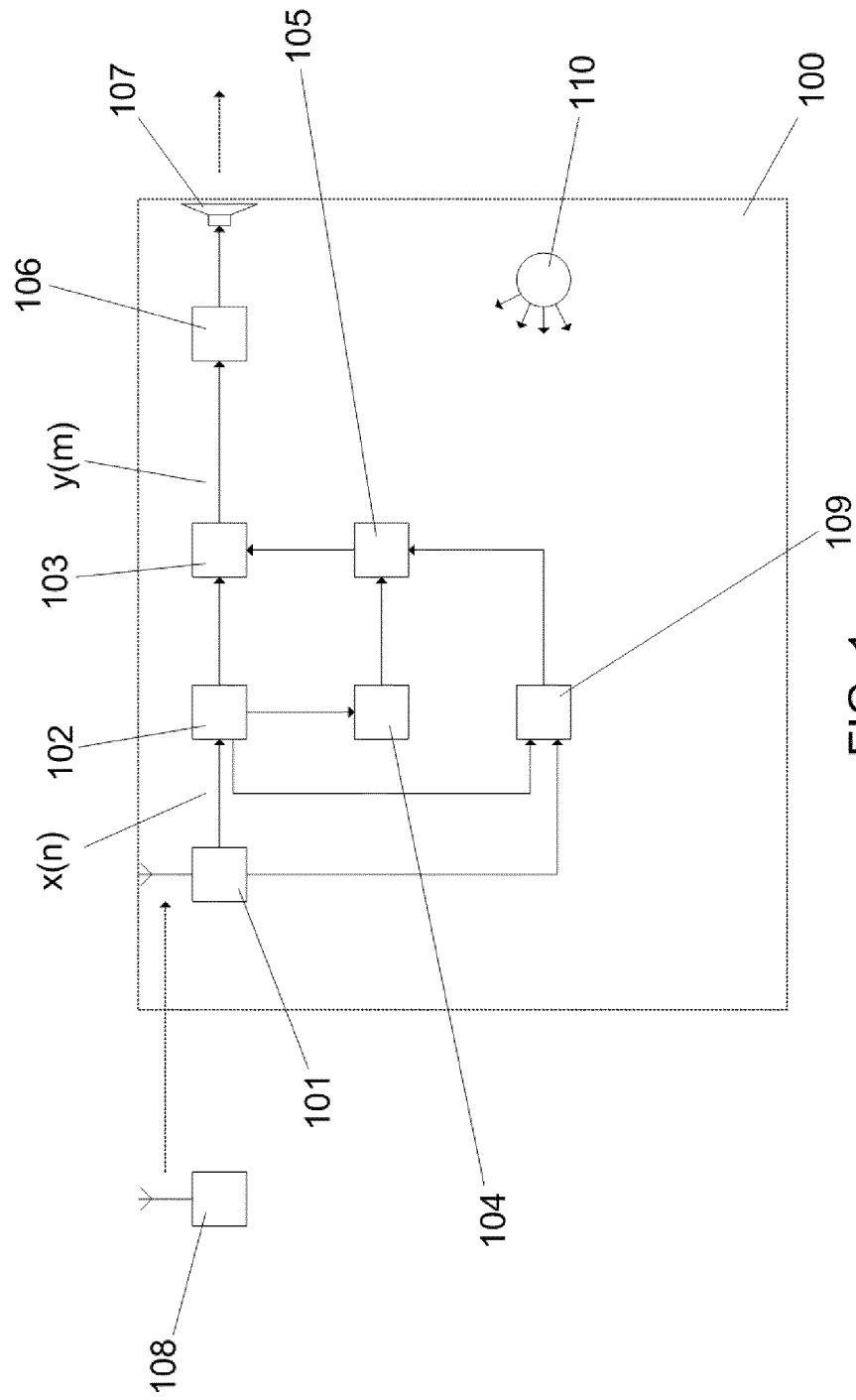
FIG. 1

FIG. 2

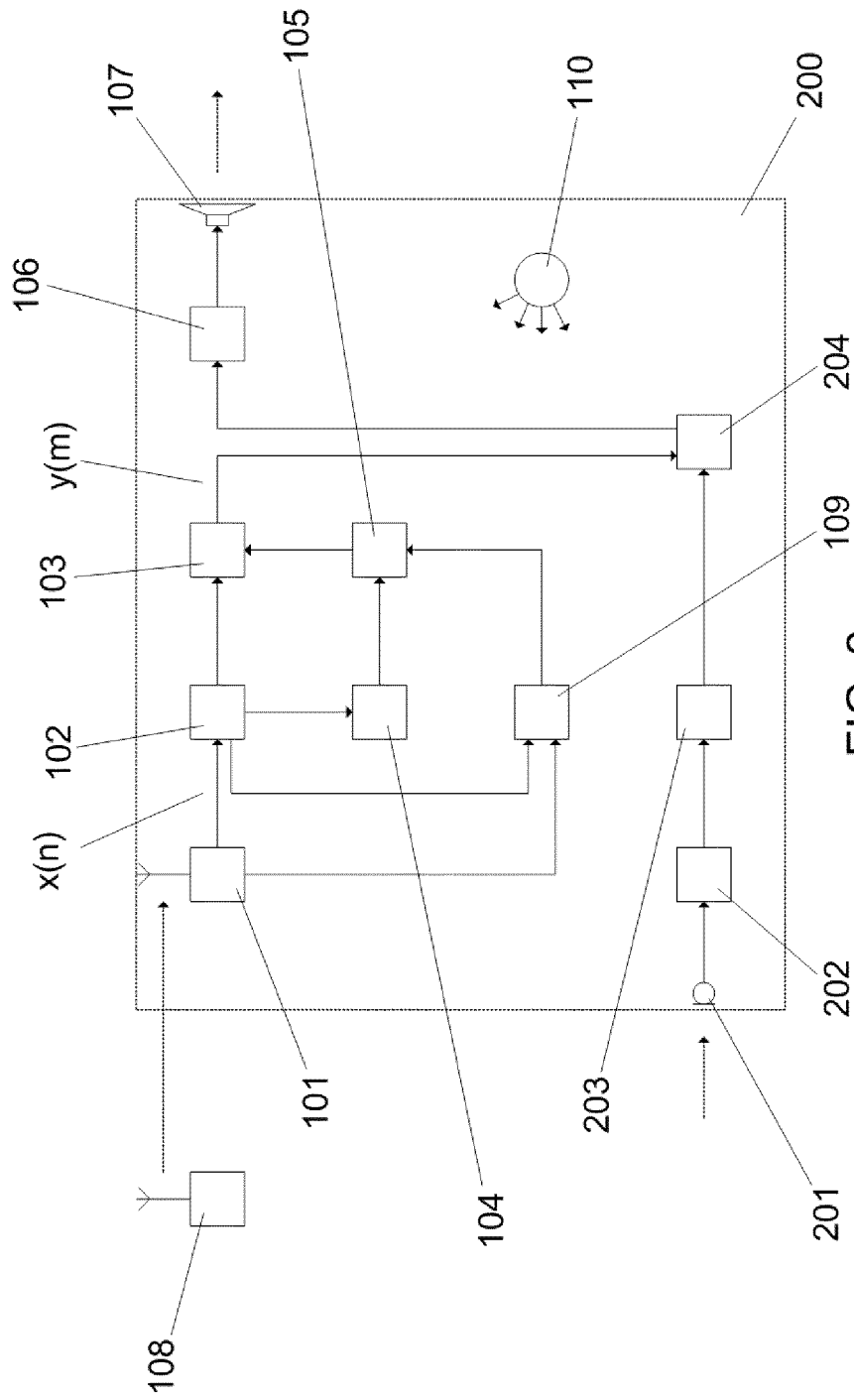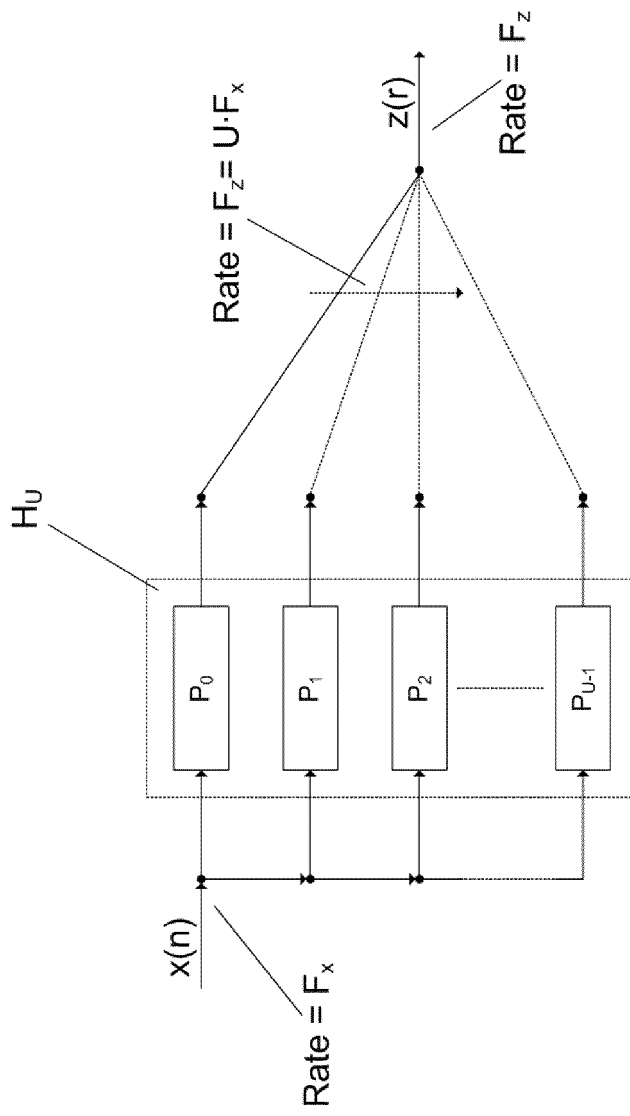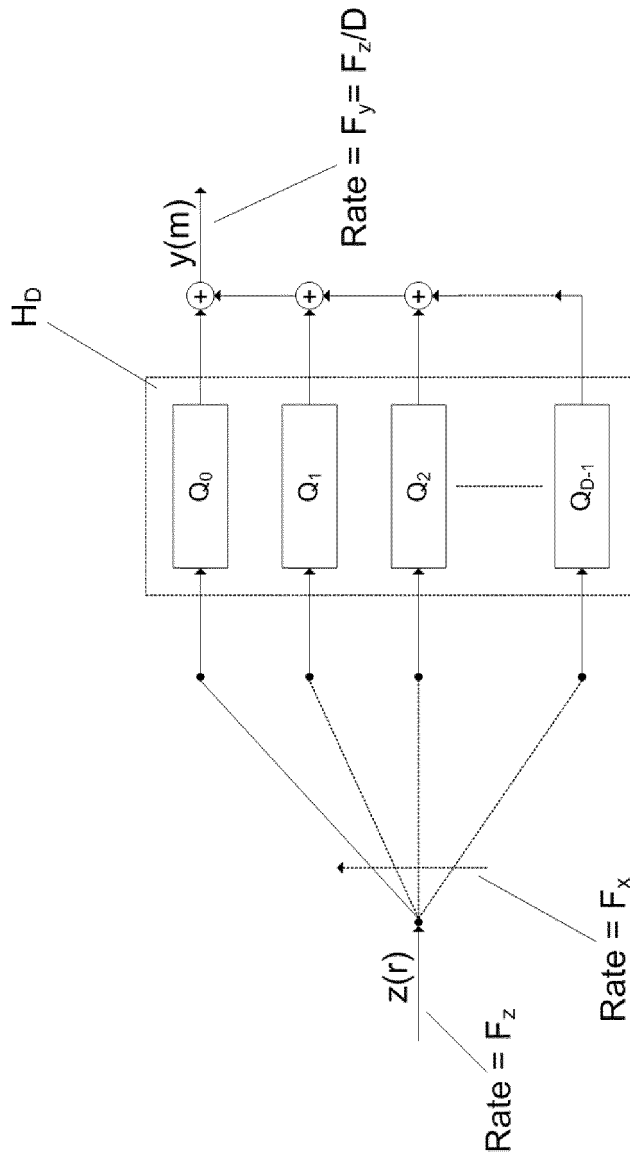FIG. 3

FIG. 4

FIG. 5

FIG. 6
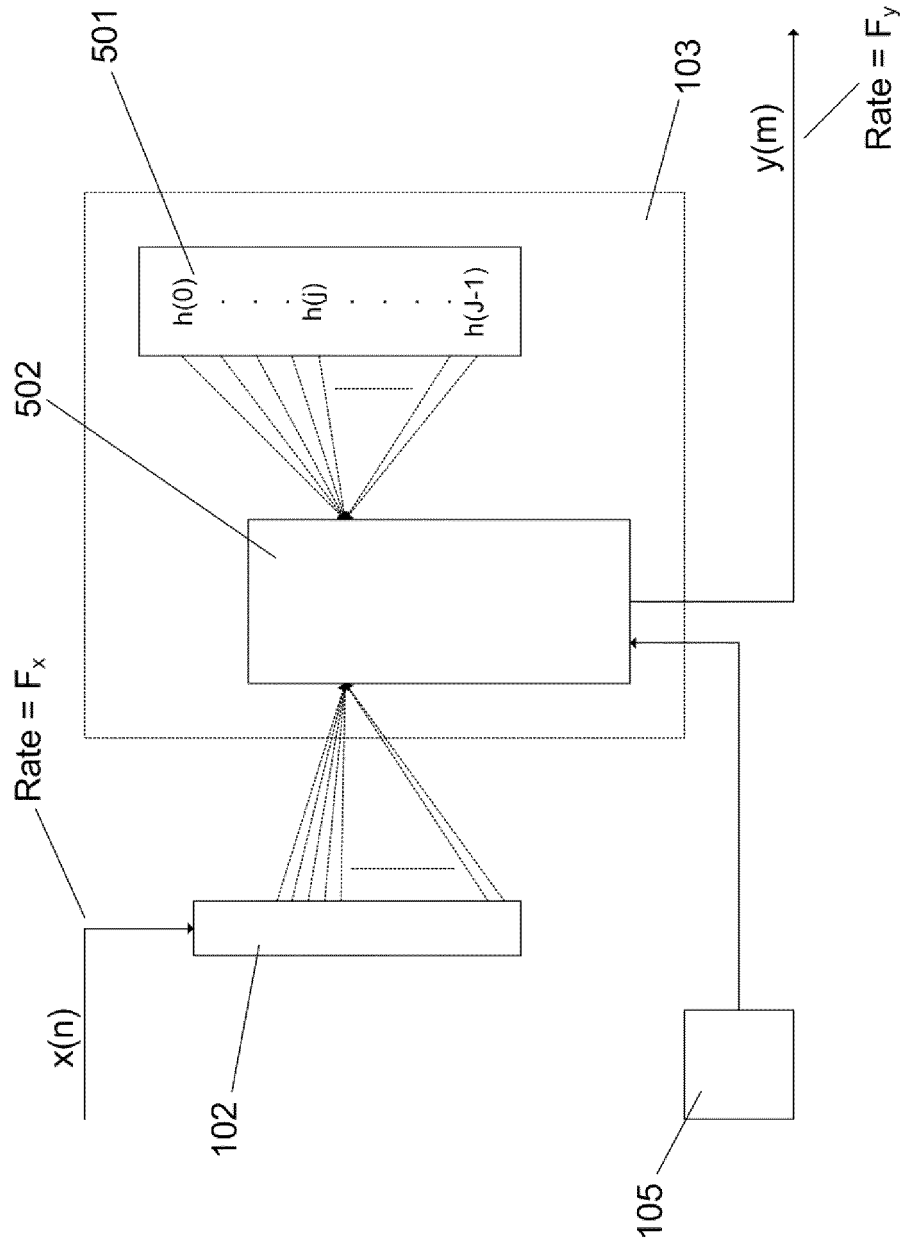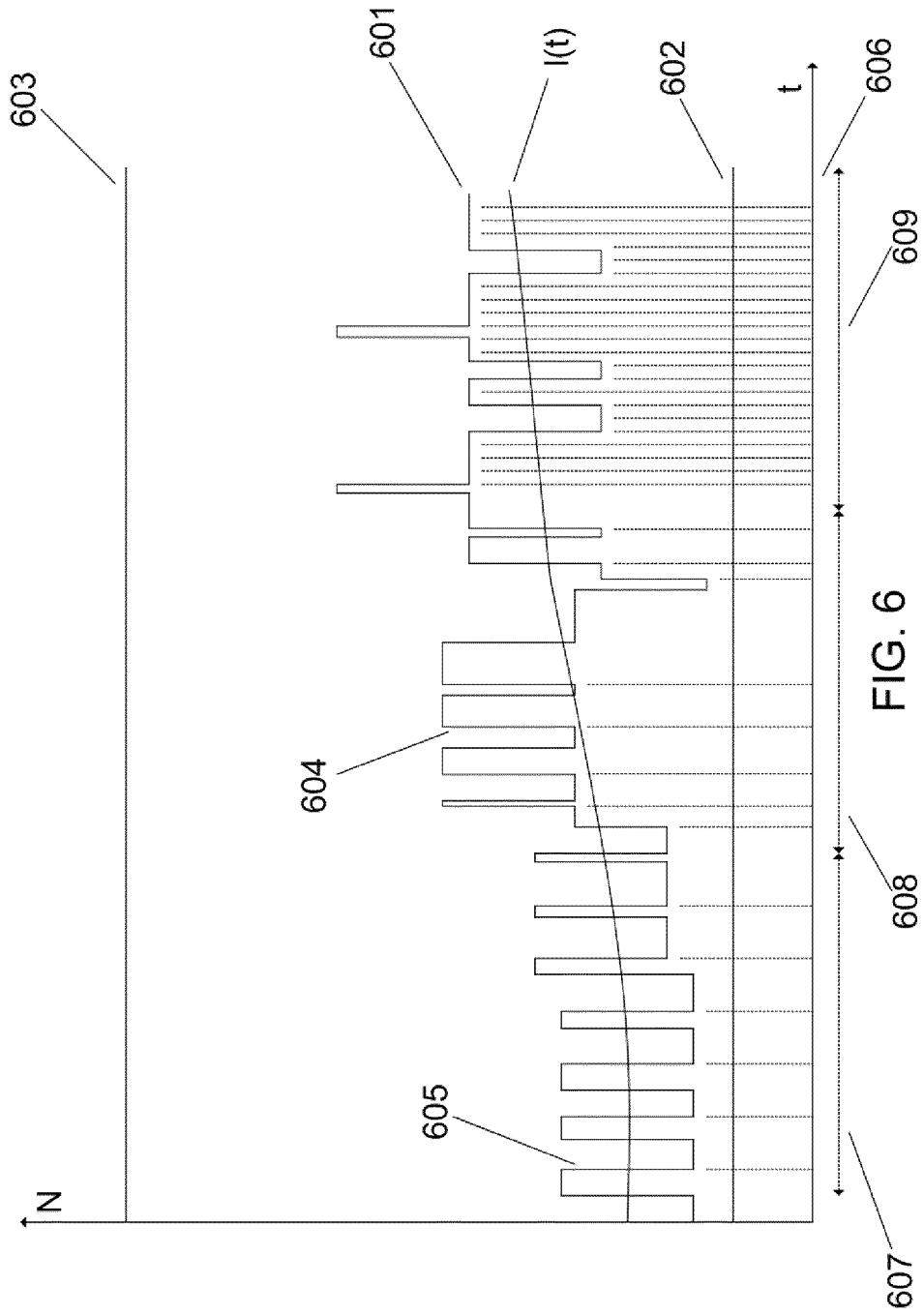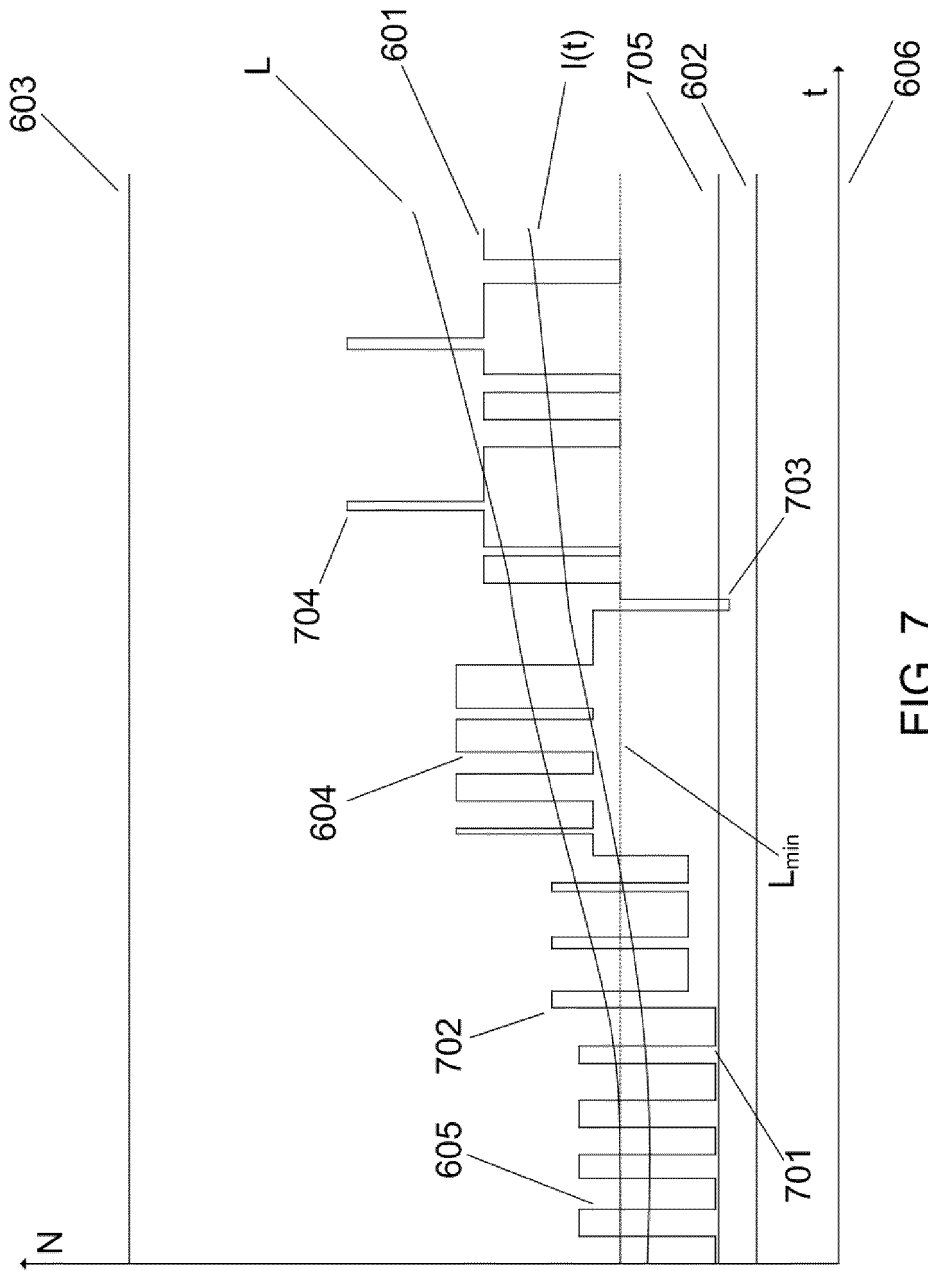
FIG. 7

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- **JOHN G. PROAKIS ; DIMITRIS K MANOLAKIS.** Digital Signal Processing. Prentice Hall, 07 April 2006 **[0076]**