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DESCRIPTION

[0001] The invention regards a binaural hearing system comprising a left hearing device, a right hearing device, and a link between the two hearing devices and a method for operating a binaural hearing system.

[0002] Hearing devices generally comprise a microphone, a power source, electric circuitry and a speaker (receiver). Binaural hearing devices typically comprise two hearing devices, one for a left ear and one for a right ear of a listener. The sound received by a listener through his ears often consists of a complex mixture of sounds coming from all directions. The healthy auditory system possesses a remarkable ability to separate the sounds originating from different sources. Furthermore, normal-hearing (NH) listeners have an amazing ability to follow the conversation of a single speaker in the presence of others, a phenomenon known as the "cocktail-party problem".

[0003] The single most common complaint among people with hearing loss is the difficulty in understanding speech in complex acoustic environments, such as background noise, reverberation or competing talkers. Although compensating for the reduced sensitivity (e.g., by hearing aids) largely improves the ability to understand speech in quiet and to some extent in noisy environments many hearing-impaired (HI) listeners still show great difficulties in adverse conditions.

[0004] Normal-hearing (NH) listeners can use Interaural Time Difference (ITD), the difference in arrival time of a sound between the two ears, and Interaural Level Difference (ILD), the difference in level of a sound between the two ears caused by shadowing of the sound by the head, to cancel sounds in the left ear which are coming from the right side of the listener and sounds in the right ear which are coming from the left side of the listener. This phenomenon is called binaural Equalization-Cancellation (EC) and was first described in "Equalization and Cancellation Theory of Binaural Masking-Level Differences", N. I. Durlach, J. Acoust. Soc. Am. 35, 1206 (1963). The result of this is that the signal-to-noise ratio (SNR) of the right source is improved in the right ear while the SNR of the left source is improved in the left ear. Accordingly, the listener can select which source to attend to. Normal-hearing (NH) listeners can do this rather effectively, while hearing-impaired (HI) listeners often have problems doing this, leading to significantly reduced speech intelligibility in adverse conditions.

[0005] C. Kim, K. Kumar, and R. M. Stern, "Binaural sound source separation motivated by auditory processing", Proc. ICASSP, pp. 5072-5075 (2011) presents a method of signal processing for speech recognition using two microphones. Speech signals detected by two microphones are passed through bandpass filtering in a filter bank. Interaural cross-correlation is used to generate a spatial masking function. The spatial masking function and a temporal mask are combined and applied on the speech signals.

[0006] J. Li, S. Sakamoto, S. Hongo, M. Akagi, and Y. Suzuki, "Two-stage binaural speech

enhancement with Wiener filter based on equalization-cancellation model", in Proc. IEEE WASPAA, 2009, pp.133-136 shows a method for binaural speech enhancement. The method is based on the equalization-cancellation (EC) model. In a first stage interfering signals are estimated by equalizing and cancelling a target signal based on the EC model. A time-variant Wiener filter is applied to enhance the target signal given noisy mixture signals in a second stage.

[0007] In J. Li, S. Sakamoto, S. Hongo, M. Akagi, and Y. Suzuki, "Two-stage binaural speech enhancement with Wiener filter for high-quality speech communication", Speech Commun. 53, pp. 677-689 (2011) a two-input two-output system for speech communication is presented. The system comprises a two-stage binaural speech enhancement with Wiener filter approach. In a first stage interference signals are estimated by equalization and cancellation processes for a target signal. The cancellation is performed for interference signals. In a second stage a time-variant Wiener filter is applied to enhance the target signal given noisy mixture signals.

[0008] WO 2004/114722 A1 presents a binaural hearing aid system with a first and second hearing aid, each comprising a microphone, an A/D converter, a processor, a D/A converter, an output transducer, and a binaural sound environment detector. The binaural sound environment detector determines a sound environment surrounding a user of the binaural hearing aid system based on at least one signal from the first hearing aid and at least one signal from the second hearing aid. The binaural sound environment determination is used for provision of outputs for each of the first and second hearing aids for selection of the signal processing algorithm of each of the hearing aid processors. This allows the binaural hearing aid system to perform coordinated sound processing. WO2008006401A1 deals with a method for manufacturing an audible signal to be perceived by an individual in dependency from an acoustical signal source, whereby the individual wears a right-ear and a left-ear hearing device, respectively with a right-ear and with a left-ear microphone arrangement and with a right-ear and with a left-ear speaker arrangement. The input signal of the right-ear and left-ear speaker arrangement is generally dependent from the output signal of the right-ear and left-ear microphone arrangement, respectively. Only when an acoustical signal source to be perceived is located laterally of the individual's head in a range of DOA which is $45° < DOA < 135°$ or $225° < DOA < 315°$ relative to the individual's horizontal straight-ahead direction, a predominant dependency of the input signal of the contra-lateral speaker arrangement from the output signal of the ipsi-lateral microphone arrangement is established.

[0009] It is an object of the invention to provide an improved binaural hearing system and an improved method for processing binaural sound signals.

[0010] This object is achieved by a binaural hearing system comprising a first hearing device and a second hearing device as defined by claim 1. Each of the hearing devices comprises a power source, an output transducer, an environment sound input, a link unit and electric circuitry. The environment sound input is configured to receive sound from an acoustic environment and to generate an environment sound signal. The link unit is configured to transmit the environment sound signal from the hearing device comprising the link unit to a link

unit of the other hearing device of the binaural hearing system and to receive a transmitted environment sound signal from the other hearing device of the binaural hearing system. The electric circuitry comprises a filter bank. The filter bank is configured to process the environment sound signal and the transmitted environment sound signal by generating processed environment sound signals and processed transmitted environment sound signals. Each of the processed environment sound signals and processed transmitted environment sound signals corresponds to a frequency channel determined by the filter bank. The electric circuitry of each of the hearing devices is configured to use the processed environment sound signals of the respective hearing device and the processed transmitted environment sound signals from the other hearing device to estimate a respective time delay between the environment sound signal and the transmitted environment sound signal. The electric circuitry is configured to apply the respective time delay to the transmitted environment sound signal to generate a time delayed transmitted environment sound signal. The time delays estimated in the respective hearing devices using the processed environment sound signal of the respective hearing device and the processed transmitted environment sound signal of the other hearing device can be different, e.g., as the shadowing effect of the head can depend on the sound source location and on degree of symmetry of a head between the hearing devices.

[0011] The electric circuitry is configured to scale the time delayed transmitted environment sound signal by a respective interaural level difference to generate an equalized transmitted environment sound signal. The electric circuitry is configured to subtract the equalized transmitted environment sound signal from the environment sound signal to receive an equalized-cancelled environment sound signal. And the electric circuitry is configured to determine a target signal as either the equalized-cancelled environment sound signal of the first hearing device or the equalized-cancelled environment sound signal of the second hearing device based on whichever has the strongest pitch, and to use the target signal to generate an output sound signal, and to apply at the other hearing device the respective time delay to the target signal and to scale the target signal by the respective interaural level difference generating the output sound signal at the other hearing device. The respective equalized-cancelled environment sound signals, the respective output sound signals and therefore also the output sounds can be different for each of the hearing devices.

[0012] One aspect of the invention is the improvement of left environment sound signal in the right ear and right environment sound signal in the left ear when in use in a binaural hearing system of a left hearing device worn at the left ear and a right hearing device worn at the right ear. Another aspect of the invention is an increase of intelligibility for hearing impaired (HI) listeners, who are not able to perform this task without a binaural hearing system.

[0013] The electric circuitry can comprise processing units, which can perform one, some or all of the tasks (signal processing) of the electric circuitry. Preferably the electric circuitry comprises a time delay estimation unit configured to use the processed environment sound signals of the respective hearing device and the processed transmitted environment sound signals from the other hearing device to estimate a respective time delay between the environment sound signal and the transmitted environment sound signal. In one embodiment

the electric circuitry comprises a time delay application unit configured to apply the respective time delay to the transmitted environment sound signal to generate a time delayed transmitted environment sound signal. In one embodiment the electric circuitry comprises an interaural level difference scaling unit configured to scale the time delayed transmitted environment sound signal by a respective interaural level difference to generate an equalized transmitted environment sound signal. The interaural level difference scaling is used to scale target or masking components of an environment sound signal. Masking components are noise components which decrease the signal quality and target components are signal components which increase the signal quality. In one embodiment the electric circuitry comprises a subtraction unit configured to subtract the equalized transmitted environment sound signal from the environment sound signal to receive an equalized-cancelled environment sound signal. In one embodiment the electric circuitry comprises an output signal generation unit which is configured to use the target signal to generate an output sound signal, which can be converted into an output sound by the output transducer.

[0014] In a preferred embodiment the filter banks of the electric circuitry comprise a number of band-pass filters. The band-pass filters are preferably configured to divide the environment sound signal and transmitted environment sound signal into a number of environment sound signals and transmitted environment sound signals each corresponding to a frequency channel determined by one of the band-pass filters. The band-pass filters preferably each generate a copy of the respective signal and perform band-pass filtering on the copy of the respective signal. Each band-pass filter has a predetermined center frequency and a predetermined frequency bandwidth which correspond to a frequency channel. The band-pass filter passes only frequencies within a certain frequency range defined by the center frequency and the frequency bandwidth. Frequencies outside the frequency range defined by the center frequency and the frequency bandwidth of the band-pass filter are removed by the band-pass filtering. The center frequencies of the band-pass filters are preferably linearly spaced according to Equivalent Rectangular Bandwidth (ERB). The center frequencies of the band-pass filters are preferably between 0 Hz and 8000 Hz, e.g. between 100 Hz and 2000 Hz, such as between 100 Hz and 600 Hz. The fundamental frequency of voices or speech of individuals can have a broad range with high fundamental frequencies for women and children with up to 600 Hz. The fundamental frequencies of interest are those below approximately 600 Hz, preferably below approximately 300 Hz including speech modulations and pitch of voiced speech.

[0015] Preferably the electric circuitry of each of the hearing devices comprises a rectifier. The rectifier is preferably configured to half-wave rectify respective sound signals of each of the frequency channels. The rectifier can also be configured to rectify a respective incoming sound signal.

[0016] Preferably the electric circuitry of each of the hearing devices comprises a low-pass filter. The low-pass filter is preferably configured to low-pass filter respective sound signals of each of the frequency channels. Low-pass filtering here means that amplitudes of signals with frequencies above a cut-off frequency of the low-pass filter are removed and low-frequency

signals with a frequency below the cut-off frequency of the low-pass filter are passed.

[0017] Preferably each of the electric circuitries is configured to generate a processed environment sound signal and a processed transmitted environment sound signal in each of the frequency channels by using the filter bank, the rectifier, and the low-pass filter. Each of the electric circuitries can also be configured to use only the filter bank or the filter bank and the rectifier or the filter bank and the low-pass filter to generate a processed environment sound signal and a processed transmitted environment sound signal in each of the frequency channels.

[0018] In one embodiment the electric circuitry of each of the hearing devices is configured to determine a cross-correlation function between the processed environment sound signals and the processed transmitted environment sound signals of each of the frequency channels. The cross-correlation function can be determined on a frame base (frame based cross-correlation) or continuously (running cross-correlation). Preferably all cross correlation functions are summed and a time delay is estimated from the peak with smallest lag or lag of the largest peak of the summed cross-correlation functions. Alternatively the time delay of each frequency channel can also be estimated as the peak with smallest lag or lag of the largest peak. A time delay between the environment sound signals and the transmitted environment sound signals can then be determined by averaging the time delays of each frequency channel across all frequency channels. The electric circuitry of one of the respective hearing devices can also be configured to determine the time delay with a different method than the electric circuitry of the other hearing device.

[0019] A respective time delay determined in the first hearing device can be different from a respective time delay determined in the second hearing device, as the first hearing device determines the respective time delay based on sound coming from a second half plane and the second hearing device determines the respective time delay based on sound coming from a first half plane. To understand the half planes we consider a head wearing the first and second hearing device on two sides of the head. A first sound source is located on a first side of the head, representing the first half plane and a second sound source is located on a second side of the head, representing the second half plane. Therefore, e.g., a shadowing effect by a head can be different for the two hearing devices, and also the location of sound sources is typically not symmetric. This can lead to different time delays between the environment sound signal and the transmitted environment sound signal in the first hearing device and second hearing device.

[0020] In a preferred embodiment the electric circuitry of each of the hearing devices comprises a lookup table with a number of predetermined scaling factors. Each of the predetermined scaling factors preferably corresponds to a time delay range or time delay. The lookup tables with predetermined scaling factors can be different for each of the hearing devices, e.g., the predetermined scaling factors can be different and/or the lookup table time delay ranges or time delays can be different for the lookup tables. The predetermined scaling factors can be determined in a fitting step to determine the respective interaural level

difference of sound between the two hearing devices of the binaural hearing system. Alternatively some standard predetermined scaling factors can be used, which are preferably determined in a standard setup with a standard head and torso simulator (HATS). The interaural level difference can also be determined from the processed environment sound signals and the processed transmitted environment sound signals using the determined time delays. The interaural level difference can be determined for target sound or masking sound or sound comprising both target and masking sound in dependence of the predetermined scaling factors. Preferably the predetermined scaling factors are determined such that the interaural level difference of masking sound is determined. The interaural level difference results from the difference in sound level of sound received by the two hearing devices due to a different distance to the sound source and a possible shadowing effect of a head between the hearing devices of a binaural hearing system. The respective interaural level difference is preferably determined by the respective lookup table in dependence of the respective time delay between the environment sound signal and the transmitted environment sound signal.

[0021] In an embodiment, the first hearing device determines the respective interaural level difference based on sound coming from a second half plane and the second hearing device determines the respective interaural level difference based on sound coming from a first half plane.

[0022] The electric circuitry of each of the hearing devices is configured to delay and attenuate the transmitted environment sound signal with the time delay and interaural level difference determined by the hearing device and subtract this signal from the environment sound signal of the hearing device to generate a equalized-cancelled environment sound signal.

[0023] The filter bank of the electric circuitry of each of the hearing devices of the binaural hearing system is configured to process the equalized-cancelled environment sound signal by generating processed equalized-cancelled environment sound signals. Each of the processed equalized-cancelled environment sound signals corresponds to a frequency channel determined by the filter bank. The electric circuitry of each of the hearing devices is preferably configured to determine an auto-correlation function of the processed equalized-cancelled environment sound signals in each frequency channel. The auto-correlation function is preferably determined in short time frames or by using a sliding window. The electric circuitry of each of the hearing devices is preferably configured to determine a summed auto-correlation function of the processed equalized-cancelled environment sound signals of each frequency channel by summing the auto-correlation function of the processed equalized-cancelled environment sound signals of each frequency channel across all frequency channels at each time step. The time steps result from the duration of the short time frames or from a predefined time step of the sliding window. The electric circuitry of each of the hearing devices is preferably configured to determine a pitch from a lag of a largest peak in the summed auto-correlation function and to determine the pitch strength by the peak-to-valley ratio of the largest peak. The electric circuitry of each of the hearing devices is preferably configured to provide the pitch and pitch strength to the link unit of the respective hearing device. The link unit is preferably configured to transmit the pitch and pitch strength to the link unit of the other

hearing device of the binaural hearing system and to receive the pitch and pitch strength from the other hearing device. Alternatively the electric circuitry of each of the hearing devices can also be configured to provide the summed auto-correlation function to the link unit of the respective hearing device. In this case the link unit can be configured to transmit the summed auto-correlation to the link unit of the other hearing device of the binaural hearing system and to receive a transmitted summed auto-correlation function from the other hearing device. The electric circuitry of each of the hearing devices can then be configured to determine a pitch from a lag of a largest peak in the summed auto-correlation function and the transmitted summed auto-correlation function and to determine the pitch strength by the peak-to-valley ratio of the largest peak.

[0024] The electric circuitries is configured to compare the pitches of the equalized-cancelled environment sound signals of both hearing devices to determine a strongest and/or weakest pitch. A target signal is determined as the processed equalized-cancelled environment sound signal or the processed transmitted equalized-cancelled environment sound signal with the strongest pitch by the electric circuitry of each of the hearing devices. Preferably each of the electric circuitries is configured to provide the target signal to the link unit of the respective hearing device. Each of the link units is preferably configured to transmit the target signal to the link unit of the other hearing device.

[0025] Alternatively the equalized-cancelled environment sound signal of a respective hearing device can be transmitted to the other hearing device and a transmitted equalized-cancelled environment sound signal can be received by the respective hearing device from the other hearing device, such that both hearing devices contain an equalized-cancelled environment sound signal and a transmitted equalized-cancelled environment sound signal.

[0026] A noise signal can be determined as the equalized-cancelled environment sound signal or transmitted equalized-cancelled environment sound signal with the weakest pitch by the electric circuitry of each of the hearing devices.

[0027] In another preferred embodiment each of the electric circuitries is configured to process the equalized-cancelled environment sound signal by generating processed equalized-cancelled environment sound signals in each of the frequency channels by using the filter bank, the rectifier, and the low-pass filter. Each of the electric circuitries can also be configured to use only the filter bank or the filter bank and the rectifier or the filter bank and the low-pass filter to generate a processed equalized-cancelled environment sound signal in each of the frequency channels. The filter bank is configured to process the equalized-cancelled environment sound signal in an equivalent way to the environment sound signal and the transmitted environment sound signal. The processed equalized-cancelled environment sound signals of the frequency channels of the two hearing devices can be used to determine a target signal and a noise signal. Preferably the pitch and pitch strengths of the processed equalized-cancelled environment sound signals are determined and transmitted to the other hearing device to determine a target signal and a noise signal. Alternatively the processed equalized-cancelled environment sound signals can be transmitted to the other hearing device to

determine a target signal and a noise signal.

[0028] The electric circuitry of each of the hearing devices is configured to apply the respective time delay to the target signal. The electric circuitry is also configured to scale the target signal by a respective interaural level difference. The electric circuitry is further configured to generate an output sound signal by applying the respective time delay to the target signal and scaling the target signal received from the other hearing device. As an example we consider a situation with a left hearing device, respectively a first hearing device and right hearing device, respectively a second hearing device. If the target signal is the equalized-cancelled environment sound signal of the right hearing device the target signal is transmitted to the left hearing device, where it is time delayed according to a time delay determined in the left hearing device and scaled according to an interaural level difference determined in the left hearing device. The target signal of the right hearing device is the output sound signal in the right hearing device and the transmitted time delayed and scaled target signal is the output sound signal in the left hearing device. If the target signal is the equalized-cancelled environment sound signal of the left hearing device the target signal is transmitted to the right hearing device, where it is time delayed according to a time delay determined in the right hearing device and scaled according to an interaural level difference determined in the right hearing device. The target signal of the left hearing device is the output sound signal in the left hearing device and the transmitted time delayed and scaled target signal is the output sound signal in the right hearing device. The respective output sound signal can be converted to output sound by an output transducer, e.g., a speaker, a bone anchored transducer, a cochlear implant or the like.

[0029] Preferably the electric circuitry of each of the hearing devices is configured to determine a noise signal as the equalized-cancelled environment sound signal with the weakest pitch. As an example we consider a situation with a left hearing device, respectively a first hearing device and right hearing device, respectively a second hearing device. If the noise signal is the equalized-cancelled environment sound signal of the right hearing device the noise signal is transmitted to the left hearing device, where it is time delayed according to a time delay determined in the left hearing device and scaled according to an interaural level difference determined in the left hearing device. If the noise signal is the equalized-cancelled environment sound signal of the left hearing device the noise signal is transmitted to the right hearing device, where it is time delayed according to a time delay determined in the right hearing device and scaled according to an interaural level difference determined in the right hearing device. Preferably the overall level of the noise signal is reduced in order to improve a signal-to-noise ratio (SNR) in both a left output sound signal and a right output sound signal.

[0030] The electric circuitry can be configured to apply the time delay to the noise signal. Preferably the electric circuitry is configured to reduce the overall level of the noise signal. The electric circuitry can be configured to combine the noise signal and the target signal to generate an output sound signal or add the noise signal to an output sound signal comprising the target signal to generate an output sound signal comprising the target signal and the noise signal. One electric circuitry can also be configured to provide an output sound signal to the

output transducer of one of the hearing devices and the other electric circuitry can be configured to provide a noise signal to the output transducer on the other of the hearing devices.

[0031] In a preferred embodiment the electric circuitry of each of the hearing devices is configured to determine a gain in each time-frequency region based on the energy of the target signal or on the signal-to-noise ratio (SNR) of the target signal and the noise signal. The time-frequency regions are defined by the time steps and frequency channels. Preferably the electric circuitry is configured to apply the gain to the environment sound signal. Preferably a high gain is applied in time-frequency regions where the target signal is above a certain threshold and a low gain in time-frequency regions where the target signal is below a certain threshold. This removes time-frequency regions with noise and keeps time-frequency regions with target signal, therefore removing most of the noise. The gain can also be applied as a function of energy of the target signal and time-frequency region, i.e., with the gain depending on the value of the energy of the target signal.

[0032] In one embodiment the link unit of each of the hearing devices is a wireless link unit, e.g., a bluetooth transceiver, an infrared transceiver, a wireless data transceiver or the like. The wireless link unit is preferably configured to transmit and receive sound signals and data signals, e.g., environment sound signals, processed environment sound signals, equalized-cancelled sound signals, processed equalized-cancelled sound signals, auto-correlation functions, cross-correlation functions, gain functions, scaling parameters, pitches, pitch strengths or the like via a wireless link between the wireless link unit of one hearing device and the wireless link unit of the other hearing device of the binaural hearing system. Alternatively or additionally the link unit can comprise a wired link, e.g., a cable, a wire, or the like between the two link units of the binaural hearing system, which is configured to transmit and receive sound signals and data signals. The wired link can for example be enclosed in a pair of glasses, a frame of a pair of glasses, a hat, or other devices obvious to the person skilled in the art.

[0033] In a preferred embodiment the environment sound input of each of the hearing devices is a microphone. Preferably a left microphone is configured to receive sound and generate a left microphone signal at a left side of the binaural hearing system and a right microphone is configured to receive sound and generate a right microphone signal at a right side of the binaural hearing system.

[0034] The objective of the invention is further achieved by a method for processing of binaural sound signals as defined in claim 13. The method comprises the following steps: Receiving a first environment sound signal and a second environment sound signal. Processing the first environment sound signal and the second environment sound signal by generating processed first environment sound signals and processed second environment sound signals wherein each of the processed first environment sound signals and processed second environment sound signals corresponds to a frequency channel. Determining a cross-correlation function between the processed second environment sound signals and the processed first environment sound signals as a function of the delay of the processed first environment sound

signals in order to determine a first time delay, which is the time delay in the second hearing device of a sound source coming from a same side as the processed first environment sound signals. Determining a cross-correlation function between the processed first environment sound signals and the processed second environment sound signals as a function of the delay of the processed second environment sound signals in order to determine a second time delay, which is the time delay in the first hearing device of a sound source coming from a same side as the processed second environment sound signals. Alternatively, the first and second time delay can also be determined after summing all the cross-correlation functions. Applying the second time delay to the second environment sound signal to generate a time delayed second environment sound signal. Applying the first time delay to the first environment sound signal to generate a time delayed first environment sound signal. Scaling the time delayed second environment sound signal by a second interaural level difference to generate an equalized second environment sound signal. Scaling the time delayed first environment sound signal by a first interaural level difference to generate an equalized first environment sound signal. Subtracting the equalized second environment sound signal from the first environment sound signal to receive an equalized-cancelled first environment sound signal. Subtracting the equalized first environment sound signal from the second environment sound signal to receive an equalized-cancelled second environment sound signal. Using the equalized-cancelled first environment sound signal and the equalized-cancelled second environment sound signal comprises the steps of: determining a target signal as either the equalized-cancelled first environment sound signal or the equalized-cancelled second environment sound signal based on whichever has the strongest pitch, and using the target signal to generate a first output sound signal, and to apply the respective time delay to the target signal and to scale the target signal by the respective interaural level difference to generate the second output sound signal.

[0035] This delay is a part of the calculation in the hearing device. Basically, the hearing device generates a cross-correlation function which is defined for a range of different delays. This function is e.g. obtained by shifting one of the signals by one sample at the time and for each shift calculating the cross correlation coefficient. In this case it is the processed first environment sound signals that is shifted/delayed in order to calculate the delay of the first sound source in the second hearing device.

[0036] The method using the equalized-cancelled first environment sound signal and equalized-cancelled second environment sound signal comprises the steps of processing the equalized-cancelled first environment sound signal by generating processed equalized-cancelled first environment sound signals with each of the processed equalized-cancelled first environment sound signals corresponding to a frequency channel. Processing the equalized-cancelled second environment sound signal by generating processed equalized-cancelled second environment sound signals with each of the processed equalized-cancelled second environment sound signals corresponding to a frequency channel. In a preferred embodiment the method comprises the steps of determining an auto-correlation function of the processed equalized-cancelled first environment sound signals in each frequency channel and determining an auto-correlation function of the processed equalized-cancelled second environment sound signals in each frequency channel. Determining a first summed auto-

correlation function of the processed equalized-cancelled first environment sound signals of each frequency channel by summing the auto-correlation function of the processed equalized-cancelled first environment sound signals of each frequency channel across all frequency channels and determining a second summed auto-correlation function of the processed equalized-cancelled second environment sound signals of each frequency channel by summing the auto-correlation function of the processed equalized-cancelled second environment sound signals of each frequency channel across all frequency channels. Determining a pitch from a lag of a largest peak in the first summed auto-correlation function and the second summed auto-correlation function. The pitch can also be determined by other methods known in the art. Determining a pitch strength by the peak-to-valley ratio of the largest peak. The pitch strength can also be determined by other methods known in the art. The method according to claim 1 determines a target signal as the equalized-cancelled first environment sound signal (or a processed version thereof) or equalized-cancelled second environment sound signal (or a processed version thereof) with the strongest pitch. In addition, the method may comprise determining a noise signal as the equalized-cancelled first environment sound signal or equalized-cancelled second environment sound signal with the weakest pitch.

[0037] A preferred embodiment of the method comprises the step determining a gain in each time-frequency region based on the energy of the target signal or based on the signal-to-noise ratio (SNR) between the target signal and the noise signal. Preferably it also comprises the step applying the gain to the first environment sound signal and applying the gain to the second environment sound signal.

[0038] An embodiment of a binaural hearing system can be used to perform an embodiment of a method for processing of binaural sound signals.

[0039] The present invention will be more fully understood from the following detailed description of embodiments thereof, taken together with the drawings in which:

Fig. 1 shows a schematic illustration of a binaural hearing system;

Fig. 2 shows a schematic illustration of a block diagram of an auditory pre-processing stage;

Fig. 3 shows a block diagram of an equalization and cancellation stage;

Fig. 4 shows a block diagram of a target selection and gain calculation stage.

Fig. 5 shows an example of the use/processing of the equalized-cancelled microphone signals in the left and right hearing devices.

[0040] Fig. 1 shows a binaural hearing system 10 with a left hearing device 12 and a right hearing device 14. Each of the hearing devices 12 and 14 has a microphone 16, 16', a bluetooth transceiver 18, 18', electric circuitry 20, 20', a power source 22, 22', and a speaker

24, 24'.

[0041] The microphone 16 receives ambient sound from the environment on the left side of the binaural hearing system 10 and converts the ambient sound into a left microphone signal 26. The microphone 16' receives ambient sound from the environment on the right side of the binaural hearing system 10 and converts the ambient sound into a right microphone signal 26'. The bluetooth transceiver 18 is connected wirelessly to the bluetooth transceiver 18' via a link 28. The link can also be a wired link, e.g., a cable or wire and the bluetooth transceiver 18, 18' can also be any other form of transceiver, e.g., Wi-Fi, infrared, or the like. The bluetooth transceiver 18 transmits the left microphone signal 26 to the bluetooth transceiver 18' and receives the right microphone signal 26' from the bluetooth transceiver 18'. The electric circuitries 20 and 20' process the left and right microphone signals 26 and 26' and generate output sound signals 30 and 30', which are converted into output sound by the speakers 24 and 24'.

[0042] The method of processing of binaural sound signals can be performed by the binaural hearing system 10 presented in Fig. 1. The method can be divided into three stages: an auditory pre-processing stage (Fig. 2), an equalization and cancellation stage (Fig. 3), and a target selection and gain calculation stage (Fig. 4). The gain calculation can be optional. In the following we will describe the method of processing of binaural sound signals in the hearing devices 12 and 14. The method for the right hearing device 14 in this embodiment is synchronously performed to the method of the left hearing device 12. In other embodiments different methods can be performed in the left hearing device 12 and in the right hearing device 14, e.g., not all of the steps of the method have to be the same. It is also possible to have a time delay between performing a method in the left hearing device 12 and the right hearing device 14.

[0043] In the auditory pre-processing stage (Fig. 2) the left microphone signal 26 and the right microphone signal 26', are divided into a number of frequency channels using a filterbank 32 with a number of band-pass filters 34, which are followed by a rectifier 36 and a low-pass filter 38. The band-pass filters 34 process a copy of the left microphone signal 26 and the right microphone signal 26' by dividing the respective signal into frequency channels through band-pass filtering with center frequencies corresponding to a specific band-pass filter 34. The center frequencies of the band-pass filters 34 are preferably between 0 Hz and 8000 Hz, e.g. between 100 Hz and 2000 Hz, or between 100 Hz and 600 Hz. The respective band-pass-filtered microphone signal 40, respectively 40' (not shown), in one of the frequency channels is half-wave rectified by the rectifier 36 and low-pass filtered by the low-pass filter 38 in order to extract periodicities below a certain cut-off frequency of the low-pass filter 38 to generate a processed microphone signal 42, respectively 42' (Fig. 3). For frequency channels with low center frequencies the extracted periodicity corresponds to a temporal fine structure (TFS) of the signal while it corresponds to the envelope of the signal for frequency channels with higher center frequencies.

[0044] In the equalization and cancellation stage (Fig. 3) a cross-correlation function between

the processed left 42 and processed right microphone signals 42' is determined in each frequency channel. The cross-correlation function is either determined on a frame base or continuously. The determination of the cross-correlation function is divided in time steps determined by the time frame step size or a predefined time step duration for the continuously (running) cross-correlation function determination. The cross-correlation function can be determined in a cross-correlation unit 44 or by an algorithm which is performed by the electric circuitry 20.

[0045] A time delay in each frequency channel is estimated as lag with a largest peak or the peak with smallest lag. A right time delay is determined based on the cross-correlation function between the processed left microphone signal 42 and the processed right microphone signal 42' as a function of the delay of the processed right microphone signal 42'. A left time delay is determined based on the cross-correlation function between the processed right microphone signal 42' and the processed left microphone signal 42 as a function of the delay of the processed left microphone signal 42. At each time step the respective time delay between the processed left microphone signal 42 and processed right microphone signals 42' is determined as an average across all frequency channels. The time delay can be determined by a time delay averaging unit 46 or by an algorithm which is performed by the electric circuitry 20. The time delay is updated slowly over time. Alternatively, the first and second time delay is determined after summing the cross-correlation functions of the frequency channels.

[0046] In an embodiment, the hearing device generates a cross-correlation function which is defined for a range of different delays. This function is e.g. obtained by shifting one of the signals by one sample at the time and for each shift calculating the cross correlation coefficient. In this case it is the processed first environment sound signals that is shifted/delayed in order to calculate the delay of the first sound source at the second hearing device.

[0047] The left time delay is then applied to the left microphone signal 26 at the right side and the right time delay is then applied to the right microphone signal 26' at the left side generating a time delayed left microphone signal 48 at the right side and a time delayed right microphone signal 48' at the left side. Applying the left and/or right time delay can be performed by a time delay application unit 50 or by an algorithm which is performed by the electric circuitry 20.

[0048] The left microphone signal 26 at the right side is scaled by an interaural level difference determined by the right hearing device 14 and the right microphone signal 26' at the left side is scaled by an interaural level difference determined by the left hearing device 12 resulting in an equalized left microphone signal 52 and an equalized right microphone signal 52'. In this embodiment each of the interaural level differences determined by the left hearing device 12 and right hearing device 14 is determined from a lookup table based on the time delay determined by the left hearing device 12 and right hearing device 14 and thereby the direction of the sound. In this embodiment the interaural level differences determined by the left hearing device 12 and right hearing device 14 correspond to the level differences of masking components, e.g., noise or the like, between the left and right side. The interaural level

difference can also correspond to the level difference of target components. The scaling can be performed by a scaling unit 54 or by an algorithm which is performed by the electric circuitry 20.

[0049] The equalized right microphone signal 52' is then subtracted from the left microphone signal 26 at the left side generating an equalized-cancelled left microphone signal 56 and the equalized left microphone signal 52 is then subtracted from the right microphone signal 26' at the right side generating an equalized-cancelled right microphone signal 56'. The subtraction can be performed by a signal addition unit 58 or by an algorithm which is performed by the electric circuitry 20.

[0050] After this stage the equalized-cancelled microphone signals 56, 56' generated through the equalization-cancellation stage could in principle be presented to a listener by hearing devices 12 and 14 (Fig. 1), but the equalized-cancelled microphone signals 56, 56' do not comprise any spatial cues. The equalized-cancelled microphone signals 56, 56' have an improved left sound signal in the left ear and an improved right sound signal in the right ear, as masking components were removed. The spatial cues can also be regained in the target selection and gain calculation stage. Also a noise signal can be generated by the equalization-cancellation stage, if the interaural level difference corresponds to the level difference of target components. If noise signal and target signal are generated preferably one hearing device will have the target signal and the other hearing device will have the noise signal. Basically, the left hearing device cancel out sound coming from the right and the right hearing device cancel out sound coming from the left. Thus, if the target is coming from the left, the left hearing device will have the target and the right hearing device will have the masker.

[0051] In the target selection and gain calculation stage, the target signal is determined and a gain based on the target signal. The stage begins with determining which of the equalized-cancelled left microphone signal 56 or equalized-cancelled right microphone signals 56' is the target signal (cf. also block 66 in FIG. 5).

[0052] The target signal is determined as the equalized-cancelled microphone signal 56, 56' with the strongest pitch. To determine the equalized-cancelled microphone signal 56, 56' with the strongest pitch the auditory pre-processing stage using the filter bank 32 with band-pass filters 34, the rectifier 36, and the low-pass filter 38 is performed on each of the equalized-cancelled microphone signals 56, 56' generating processed equalized-cancelled microphone signals 60, 60' (cf. Fig. 4). An auto-correlation function of the respective processed equalized-cancelled microphone signal 60, 60' is determined for short time frames or by using sliding windows in each frequency channel. Determining the auto-correlation can be performed by an auto-correlation unit 62, 62' or by an algorithm which is performed by the electric circuitry 20 (cf. Fig. 1).

[0053] At each time step the auto-correlation functions are summed across all frequency channels and a pitch is determined from the lag of the largest peak in the summed auto-correlation function. The pitch strength is determined by the peak-to-valley ratio of the largest

peak. The pitch and pitch strength are updated slowly across time. The summation of the auto-correlation functions and determination of the pitch and pitch strength can be performed by a summation and pitch determination unit 64 (Fig. 4) or by an algorithm which is performed by the electric circuitry 20 (Fig. 1).

[0054] Finally the target signal 68 is chosen as the processed equalized-cancelled microphone signal 60, 60' with the strongest pitch. The noise signal 70 is chosen as the processed equalized-cancelled microphone signal 60, 60' with the weakest pitch. The target and noise selection can be performed by a target selection unit 66 or by an algorithm which is performed by the electric circuitry 20.

[0055] An example of the further use/processing of the equalized-cancelled microphone signals 56, 56' (Fig. 3) in the left and right hearing devices 12, 14 is illustrated in Fig. 5.

[0056] In order to determine the target signal 68 and noise signal 70 the pitch and pitch strength of the left hearing device 12 is transmitted to the right hearing device 14 and vice versa. The pitch strength of the respective equalized-cancelled microphone signal 56 or 56' is compared to the transmitted pitch strength of the equalized-cancelled microphone signal 56' or 56 and depending on the result, meaning which signal has the strongest/weakest pitch, the following steps are performed (cf. block 66 in Fig. 4, 5):

[0057] If the target signal 68 (cf. Fig. 4) is the processed equalized-cancelled left microphone signal 60, meaning that the equalized-cancelled left microphone signal 56 has the strongest pitch, the equalized-cancelled left microphone signal 56 is transmitted to the right hearing device 14 where it is time delayed (cf. blocks ΔT in Fig. 5) according to the time delay determined in the right hearing device 14 and scaled according to the interaural level difference determined in the right hearing device 14 (cf. multiplication factors α_{LR} in Fig. 5) generating a right output sound signal 30'. The left output sound signal 30 is the equalized-cancelled left microphone signal 56.

[0058] If the target signal 68 is the processed equalized-cancelled right microphone signal 60', meaning that the equalized-cancelled right microphone signal 56' has the strongest pitch, the equalized-cancelled right microphone signal 56' is transmitted to the left hearing device 12 where it is time delayed (cf. blocks ΔT in Fig. 5) according to the time delay determined in the left hearing device 12 and scaled according to the interaural level difference determined in the left hearing device 12 (cf. multiplication factors α_{RL} in Fig. 5) generating a left output sound signal 30. The right output sound signal 30' is the equalized-cancelled right microphone signal 56'.

[0059] The left output sound signal 30 is converted to a left output sound at the left side and the right output sound signal 30' is converted to a right output sound at the right side. The conversion of output sound signal 30, 30' to output sound is preferably performed synchronously.

[0060] The noise signal can also be added to the output sound signals 30, 30' or used as one or both of the output sound signals 30, 30'.

[0061] If the noise signal 70 is the processed equalized-cancelled left microphone signal 60, the equalized-cancelled left microphone signal 56 is transmitted to the right hearing device where it is time delayed according to the time delay determined in the right hearing device 14 and scaled according to the interaural level difference determined in the right hearing device 14 generating a right output sound signal 30'. The left output sound signal 30 is the equalized-cancelled left microphone signal 56.

[0062] If the noise signal 70 is the processed equalized-cancelled right microphone signal 60', the equalized-cancelled right microphone signal 56' is transmitted to the left hearing device where it is time delayed according to the time delay determined in the left hearing device 12 and scaled according to the interaural level difference determined in the left hearing device 12 generating a left output sound signal 30. The right output sound signal 30' is the equalized-cancelled right microphone signal 56'.

[0063] Preferably, the noise signal, which can either be the equalized-cancelled left microphone signal 56 or the equalized-cancelled right microphone signal 56', is attenuated compared to the target signal. This attenuation is applied by β_L if the noise signal is determined as the equalized-cancelled left microphone signal 56 and by β_R if the noise signal is determined as the equalized-cancelled right microphone signal 56'.

[0064] If the target signal (68, 68') is determined as the processed equalized-cancelled environment sound signal (60; 60') of the hearing device (12; 14), the hearing device (12; 14) is configured to apply a high gain, β_L, to the equalized-cancelled environment sound signal (56; 56') of hearing device (12; 14) before it is provided to the link unit (18; 18') and the hearing device (14; 12) is configured to apply a low gain, β_R, to the equalized-cancelled environment sound signal (56'; 56) of hearing device (14; 12) before it is provided to the link unit (18'; 18).

[0065] If the target signal (68', 68) is determined as the processed equalized-cancelled environment sound signal (60'; 60) of the hearing device (14; 12), the hearing device (14; 12) is configured to apply a high gain, β_R, to the equalized-cancelled environment sound signal (56'; 56) of hearing device (14; 12) before it is provided to the link unit (18'; 18) and the hearing device (12; 14) is configured to apply a low gain β_L, to the equalized-cancelled environment sound signal (56; 56') of hearing device (12; 14) before it is provided to the link unit (18; 18').

[0066] In another preferred embodiment a gain 72 in each time-frequency region is determined based on the energy of the target signal 68 or the signal-to-noise ratio (SNR) between the target signal 68 and the noise signal 70. The gain 72 can be determined by a gain determination unit 74 or by an algorithm which is performed by the electric circuitry 20.

[0067] Preferably a high gain is applied to the left microphone signal 26, respectively right microphone signal 26' in time-frequency regions where the target signal 68 is above a certain threshold or above a certain signal-to-noise ratio (SNR) between the target signal 68 and the noise signal 70 and a low gain is applied to the left 26, respectively right microphone signal 26' in time-frequency regions where the target signal 68 is below a certain threshold or below a certain signal-to-noise ratio (SNR) between the target signal 68 and the noise signal 70. The left output sound signal 30 is preferably converted to a left output sound at the left side synchronously with a conversion of the right output sound signal 30' to a right output sound at the right side. Only time-frequency regions of the target signal 68 are kept and most of the noise is removed. The gain application can be performed by a gain application unit 76, 76' or by an algorithm which is performed by the electric circuitry 20.

[0068] In this embodiment the processed microphone signals 42, 42' with applied gain in the frequency channels are summed across all frequency channels to generate the output sound signals 30, 30'. The summation of microphone signals with applied gain can be performed by a frequency channel summation unit 78, 78' or by an algorithm which is performed by the electric circuitry 20.

Reference signs

[0069]

10	binaural hearing system
12	left hearing device
14	right hearing device
16	microphone
18	bluetooth transceiver
20	electric circuitry
22	power source
24	speaker
26	microphone signal
28	link
30	

output sound signal

32

filter bank

34

band-pass filter

36

rectifier

38

low-pass filter

40

band-pass filtered microphone signal

42

processed microphone signal

44

cross-correlation unit

46

time delay averaging unit

48

time delayed microphone signal

50

time delay application unit

52

equalized microphone signal

54

scaling unit

56

equalized-cancelled microphone signal

58

signal addition unit

60

processed equalized-cancelled microphone signal

62

auto-correlation unit

64

summation and pitch determination unit

66

target selection unit

68

target signal

70

noise signal

72

gain

74

gain determination unit

76

gain application unit

78

frequency channel summation unit

REFERENCES CITED IN THE DESCRIPTION

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Patent documents cited in the description

- WO2004114722A1 **[0008]**
- WO2008006401A1 **[0008]**

Non-patent literature cited in the description

- **N. I. DURLACH**Equalization and Cancellation Theory of Binaural Masking-Level DifferencesJ. Acoust. Soc. Am., 1963, vol. 35, 1206- **[0004]**
- **C. KIMK. KUMARR. M. STERN**Binaural sound source separation motivated by auditory processingProc. ICASSP, 2011, 5072-5075 **[0005]**
- **J. LIS. SAKAMOTOS. HONGOM. AKAGIY. SUZUKI**Two-stage binaural speech enhancement with Wiener filter based on equalization-cancellation modelProc. IEEE WASPAA, 2009, 133-136 **[0006]**
- **J. LIS. SAKAMOTOS. HONGOM. AKAGIY. SUZUKI**Two-stage binaural speech enhancement with Wiener filter for high-quality speech communicationSpeech Commun, 2011, vol. 53, 677-689 **[0007]**

Patentkrav

1. Binauralt høresystem (10), der omfatter i det mindste en første høreanordning (12; 14) og en anden høreanordning (14; 12), der hver omfatter

en strømforsyningskilde (22; 22'), en udgangstransducer (24; 24'), en omgivelseslydindgang (16; 16') til lyd fra et akustisk miljø, der er konfigureret til at generere et omgivelseslydsignal (26; 26'),
en forbindelsesenhed (18; 18'), der er konfigureret til at transmittere omgivelseslydsignalet (26; 26') fra høreanordningen (12; 14), der omfatter forbindelsesenheden (18; 18'), til en forbindelsesenhed (18'; 18) på den anden høreanordning (14; 12) af det binaurale høresystem (10) og til at modtage et transmitteret omgivelseslydsignal (26'; 26) fra den anden høreanordning (14; 12) af det binaurale høresystem (10),
og elektriske kredsløb (20; 20') med en filterbank (32), der er konfigureret til at behandle omgivelseslydsignalet (26; 26') og det transmitterede omgivelseslydsignal (26'; 26) ved at generere behandlede omgivelseslydsignaler (42; 42') og behandlede transmitterede omgivelseslydsignaler (42'; 42), hvor hver af de behandlede omgivelseslydsignaler (42; 42') og behandlede transmitterede omgivelseslydsignaler (42'; 42) svarer til en frekvenskanal, der er bestemt af filterbanken (32),

hvor hvert af de elektriske kredsløb (20, 20') er konfigureret

- til at anvende de behandlede omgivelseslydsignaler (42; 42') af den respektive høreanordning (12; 14) og de behandlede transmitterede omgivelseslydsignaler (42'; 42) fra den anden høreanordning (14; 12) til at estimere en respektive tidsforsinkelse mellem omgivelseslydsignalet (26; 26') og det transmitterede omgivelseslydsignal (26'; 26),
- til at anvende den respektive tidsforsinkelse til det transmitterede omgivelseslydsignal (26'; 26) for at generere et tidsforsinket transmitteret omgivelseslydsignal (48'; 48),
- til at skalere det tidsforsinkede transmitterede omgivelseslydsignal (48'; 48') med en respektive interaural niveauforskel for at generere et udlignet transmitteret omgivelseslydsignal (52'; 52),
- til at subtrahere det udlignede transmitterede omgivelseslydsignal (52'; 52) fra omgivelseslydsignalet (26; 26') for at tilvejebringe et udlignet undertrykt omgivelseslydsignal (56; 56'),
- **KENDETEGNET VED, AT** hvert af de elektriske kredsløb (20, 20') er konfigureret
  - ∘ til at bestemme et målsignal (68, 68') som enten det udlignede undertrykte omgivelseslydsignal (56, 56') af den første høreanordning (12; 14) eller det udlignede undertrykte omgivelseslydsignal (56', 56) af den anden høreanordning (14; 12), der er baseret på den, der har den største tonehøjde, og
  - ∘ til at anvende målsignalet (68, 68') for at generere et udgangslydsignal (30; 30'), og til at anvende den respektive tidsforsinkelse for målsignalet ved den anden høreanordning (14, 12) og til at skalere målsignalet med den respektive interaurale

niveauforskel, der genererer udgangslydsignalet (30', 30) ved den anden høreanordning (14, 12).

2. Binauralt høresystem (10) ifølge krav 1, hvor hver af filterbankerne (32) omfatter et antal båndpasfiltre (34), der er konfigureret til at dele omgivelseslydsignalet (26, 26') og det transmitterede omgivelseslydsignal (26', 26) til et antal af omgivelseslydsignaler og transmitterede omgivelseslydsignaler, som hver svarer til en frekvenskanal, der er bestemt af et af båndpasfiltrene (34), og hvert af de elektriske kredsløb (20, 20') omfatter en ensretter (36), der er konfigureret til halvbølge-ensrette omgivelseslydsignalerne og de transmitterede omgivelseslydsignaler i frekvenskanalerne, og et lavpasfilter (38), der er konfigureret til at lavpasfiltrere omgivelseslydsignalerne og de transmitterede omgivelseslydsignaler i frekvenskanalerne, og hvor hvert af de elektriske kredsløb (20, 20') er konfigureret til at generere de behandlede omgivelseslydsignaler (42, 42') og de behandlede transmitterede omgivelseslydsignaler (42', 42) i frekvenskanalerne ved at anvende filterbanken (32), ensretteren (36) og lavpasfilteret (38).

3. Binauralt høresystem (10) ifølge i det mindste et af kravene 1 eller 2, hvor hvert af de elektriske kredsløb (20, 20') er konfigureret til at bestemme en krydskorrelationsfunktion mellem de behandlede omgivelseslydsignaler (42, 42') og de behandlede transmitterede omgivelseslydsignaler (42', 42) af hver af frekvenskanalerne, hvor hvert af de elektriske kredsløb (20, 20') er konfigureret til at summere krydskorrelationsfunktionerne af hver af frekvenskanalerne og til at estimere den respektive tidsforsinkelse fra en topværdi med den mindste forsinkelse eller fra en forsinkelse med en største topværdi af de summerede krydskorrelationsfunktioner.

4. Binauralt høresystem (10) ifølge i det mindste et af kravene 1 til 3, hvor hvert af de elektriske kredsløb (20, 20') omfatter en opslagstabel med et antal af forudbestemte skaleringsfaktorer, som hver svarer til et tidsforsinkelsesområde, og hvor den respektive interaurale niveauforskel bestemmes af opslagstabellen i afhængighed af den respektive tidsforsinkelse.

5. Binauralt høresystem (10) ifølge krav 4, hvor de forudbestemte skaleringsfaktorer, som hver svarer til et tidsforsinkelsesområde, bestemmes i et tilpasningstrin til at bestemme den respektive interaurale niveauforskel af maskeringslyd mellem de to høreanordninger (12, 14) af det binaurale høresystem (10).

6. Binauralt høresystem (10) ifølge i det mindste et af kravene 1 til 5, hvor hver af filterbankerne (32) af hvert af de elektriske kredsløb (20, 20') er konfigureret til at behandle det udlignede undertrykte omgivelseslydsignal (56, 56') ved at generere behandlede udlignede undertrykkede omgivelseslydsignaler (60, 60'), hvor hver af de behandlede udlignede undertrykkede omgivelseslydsignaler (60, 60') svarer til en frekvenskanal, der

er bestemt af filterbanken (32), hvor hvert af de elektriske kredsløb (20, 20') er konfigureret til at bestemme en autokorrelationsfunktion af de behandlede udlignede undertrykkede omgivelseslydsignaler (60, 60') i hver frekvenskanal, til at bestemme en summeret autokorrelationsfunktion af de behandlede udlignede undertrykkede omgivelseslydsignaler (60, 60') af hver frekvenskanal ved at summere autokorrelationsfunktionen af de behandlede udlignede undertrykkede omgivelseslydsignaler (60, 60') af hver frekvenskanal på tværs af alle frekvenskanaler, til at bestemme en tonehøjde fra en forsinkelse med en største topværdi i den summerede autokorrelationsfunktion, og til at bestemme tonehøjdestyrken ved top-til-dal forholdet af den største topværdi, og hvor hvert af de elektriske kredsløb (20, 20') er konfigureret til at tilvejebringe tonehøjden og tonehøjdestyrken til forbindelsesenheden (18, 18'), hvor forbindelsesenheden (18, 18') er konfigureret til at transmittere tonehøjden og tonehøjdestyrken til forbindelsesenheden (18, 18') på den anden høreanordning (14, 12) af det binaurale høresystem (10) og til at modtage en tonehøjde og en tonehøjdestyrke fra den anden høreanordning (14, 12).

7. Binauralt høresystem (10) ifølge krav 6, hvor hvert af de elektriske kredsløb (20; 20') er konfigureret til at tilvejebringe målsignalet (68; 68') til forbindelsesenheden (18; 18'), hvor forbindelsesenheden (18; 18') er konfigureret til at transmittere målsignalet (68; 68') til forbindelsesenheden (18'; 18) på den anden høreanordning (14; 12), og hvor det elektriske kredsløb (20; 20') på den anden høreanordning (14; 12) er konfigureret til at anvende den respektive tidsforsinkelse til det modtagne målsignal (68; 68') og til at skalere det modtagne målsignal (68, 68') ved en respektive interaural niveauforskel, der genererer udgangslydsignalet (30'; 30).

8. Binauralt høresystem (10) ifølge krav 7, der er konfigureret til at tilvejebringe, at hvis målsignalet (68, 68') bestemmes som det udlignede undertrykte omgivelseslydsignal (56; 56') af den første høreanordning (12; 14), er den første høreanordning (12; 14) konfigureret til at anvende en høj forstærkning β_L til det udlignede undertrykte omgivelseslydsignal (56; 56') af den første høreanordning (12; 14), inden den er tilvejebragt til forbindelsesenheden (18; 18'), og den anden høreanordning (14; 12) er konfigureret til at anvende en lav forstærkning β_R til det udlignede undertrykte omgivelseslydsignal (56'; 56) af den anden høreanordning (14; 12) inden den er tilvejebragt til forbindelsesenheden (18'; 18).

9. Binauralt høresystem (10) ifølge krav 6, hvor hvert af de elektriske kredsløb (20, 20') er konfigureret til at bestemme en forstærkning (72) i hvert tidsfrekvensområde, der er baseret på en energi af målsignalet (68, 68') og til at anvende forstærkningen (72) til omgivelseslydsignalet (26, 26').

10. Binauralt høresystem (10) ifølge i det mindste et af kravene 1 til 9, hvor hver af forbindelsesenhederne (18, 18') er en trådløs forbindelsesenhed, der er konfigureret til at transmittere lydsignaler (26, 26'; 30, 30'; 42, 42'; 48, 48'; 52, 52'; 56, 56'; 60, 60'; 68, 68'; 70, 70') via en trådløs forbindelse (28) mellem den trådløse forbindelsesenhed (18; 18') af en høreanordning (12;

14) og den trådløse forbindelsesenhed (18'; 18) på den anden høreanordning (14; 12) af det binaurale høresystem (10).

11. Binauralt høresystem (10) ifølge i det mindste et af kravene 1 til 10, hvor omgivelseslydindgangen (16, 16') er en mikrofon (16, 16').

12. Binauralt høresystem (10) ifølge i det mindste et af kravene 1 til 11, hvor den første og anden høreanordning omfatter henholdsvis et første og andet høreapparat.

13. Fremgangsmåde til behandling af binaurale lydsignaler (26, 26'), der omfatter trinene:

- modtagelse af et første omgivelseslydsignal (26; 26') og et andet omgivelseslydsignal (26'; 26),
- behandling af det første omgivelseslydsignal (26; 26') og det andet omgivelseslydsignal (26'; 26) ved at generere behandlede første omgivelseslydsignaler (42; 42') og behandlede andet omgivelseslydsignaler (42'; 42), hvor hver af de behandlede første omgivelseslydsignaler (42; 42') og behandlede andet omgivelseslydsignaler (42'; 42) svarer til en frekvenskanal,
- bestemmelse af en krydskorrelationsfunktion mellem de behandlede første omgivelseslydsignaler (42; 42') og de behandlede andet omgivelseslydsignaler (42'; 42) til at bestemme en respektive tidsforsinkelse mellem det første omgivelseslydsignal (26; 26') og det andet omgivelseslydsignal (26'; 26),
- anvendelse af den respektive tidsforsinkelse til det andet omgivelseslydsignal (26'; 26) for at generere et tidsforsinket andet omgivelseslydsignal (48'; 48) og anvendelse af den respektive tidsforsinkelse til det første omgivelseslydsignal (26; 26') for at generere et tidsforsinket første omgivelseslydsignal (48; 48'),
- skalering af det tidsforsinkede andet omgivelseslydsignal (48'; 48) med en respektive interaural niveauforskel for at generere et udlignet andet omgivelseslydsignal (52'; 52) og skalering af det tidsforsinkede første omgivelseslydsignal (48; 48') med en respektive interaural niveauforskel for at generere et udlignet første omgivelseslydsignal (52; 52'),
- subtraktion af det udlignede andet omgivelseslydsignal (52'; 52) fra det første omgivelseslydsignal (26; 26') for at tilvejebringe et udlignet undertrykt første omgivelseslydsignal (56; 56') og subtraktion af det udlignede første omgivelseslydsignal (52; 52') fra det andet omgivelseslydsignal (26'; 26) til at modtage et udlignet undertrykt andet omgivelseslydsignal (56'; 56), og

KENDETEGNET VED, AT

- anvendelse af det udlignede undertrykte første omgivelseslydsignal (56; 56') og det udlignede undertrykte andet omgivelseslydsignal (56'; 56) omfatter trinene til:
- bestemmelse af et målsignal (68; 68') som enten det udlignede undertrykte første omgivelseslydsignal (56, 56') eller det udlignede

undertrykte andet omgivelseslydsignal (56', 56), der er baseret på den, der har det største tonehøjde, og
- anvendelse af målsignalet (68, 68') for at generere et første udgangslydsignal (30; 30'), og til at anvende den respektive tidsforsinkelse til målsignalet og til at skalere målsignalet med den respektive interaurale niveauforskel for at generere det andet udgangslydsignal (30', 30).

14. Fremgangsmåde ifølge krav 13, hvor anvendelse af det udlignede undertrykte første omgivelseslydsignal (56; 56') og udlignede undertrykte andet omgivelseslydsignal (56'; 56) omfatter trinene til:
- behandling af det udlignede undertrykte første omgivelseslydsignal (56; 56') ved at generere behandlede udlignede undertrykkede første omgivelseslydsignaler (60; 60'), hvor hver af de behandlede udlignede undertrykkede første omgivelseslydsignaler (60; 60') svarer til en frekvenskanal og behandling af det udlignede undertrykte andet omgivelseslydsignal (56'; 56) ved at generere behandlede udlignede undertrykkede andet omgivelseslydsignaler (60'; 60), hvor hver af de behandlede udlignede undertrykkede andet omgivelseslydsignaler (60'; 60) svarer til en frekvenskanal, og
- bestemmelse af en autokorrelationsfunktion af de behandlede udlignede undertrykkede første omgivelseslydsignaler (60; 60') i hver frekvenskanal og bestemmelse af en autokorrelationsfunktion af de behandlede udlignede undertrykkede andet omgivelseslydsignaler (60'; 60) i hver frekvenskanal,
- bestemmelse af en første summeret autokorrelationsfunktion af de behandlede udlignede undertrykkede første omgivelseslydsignaler (60; 60') af hver frekvenskanal ved at summere autokorrelationsfunktionen af de behandlede udlignede undertrykkede første omgivelseslydsignaler (60; 60') af hver frekvenskanal på tværs af alle frekvenskanaler og bestemmelse af en anden summeret autokorrelationsfunktion af de behandlede udlignede undertrykkede andet omgivelseslydsignaler (60'; 60) af hver frekvenskanal ved at summere autokorrelationsfunktionen af de behandlede udlignede undertrykkede andet omgivelseslydsignaler (60'; 60) af hver frekvenskanal på tværs af alle frekvenskanaler,
- bestemmelse af en første tonehøjde fra en forsinkelse med en største topværdi i den første summerede autokorrelationsfunktion og bestemmelse af en anden tonehøjde fra en forsinkelse med en største topværdi i den anden summerede autokorrelationsfunktion,
- bestemmelse af en første og anden tonehøjdestyrke af de behandlede udlignede undertrykkede første og andet omgivelseslydsignaler (60; 60') ved top-til-dal forholdet af den største topværdi.

15. Fremgangsmåde ifølge krav 14, der omfatter trinnet:

- bestemmelse af en forstærkning (72) i hvert tidsfrekvensområde, der er baseret på en energi af målsignalet (68; 68'), og
- anvendelse af forstærkningen (72) til det første omgivelseslydsignal (26; 26') og anvendelse af forstærkningen til det andet omgivelseslydsignal (26'; 26).

16. Anvendelse af et binauralt høresystem (10) ifølge i det mindste et af kravene 1 til 12 til at udføre en fremgangsmåde til behandling af binaurale lydsignaler (26, 26') ifølge i det mindste et af kravene 13 til 15.

DRAWINGS

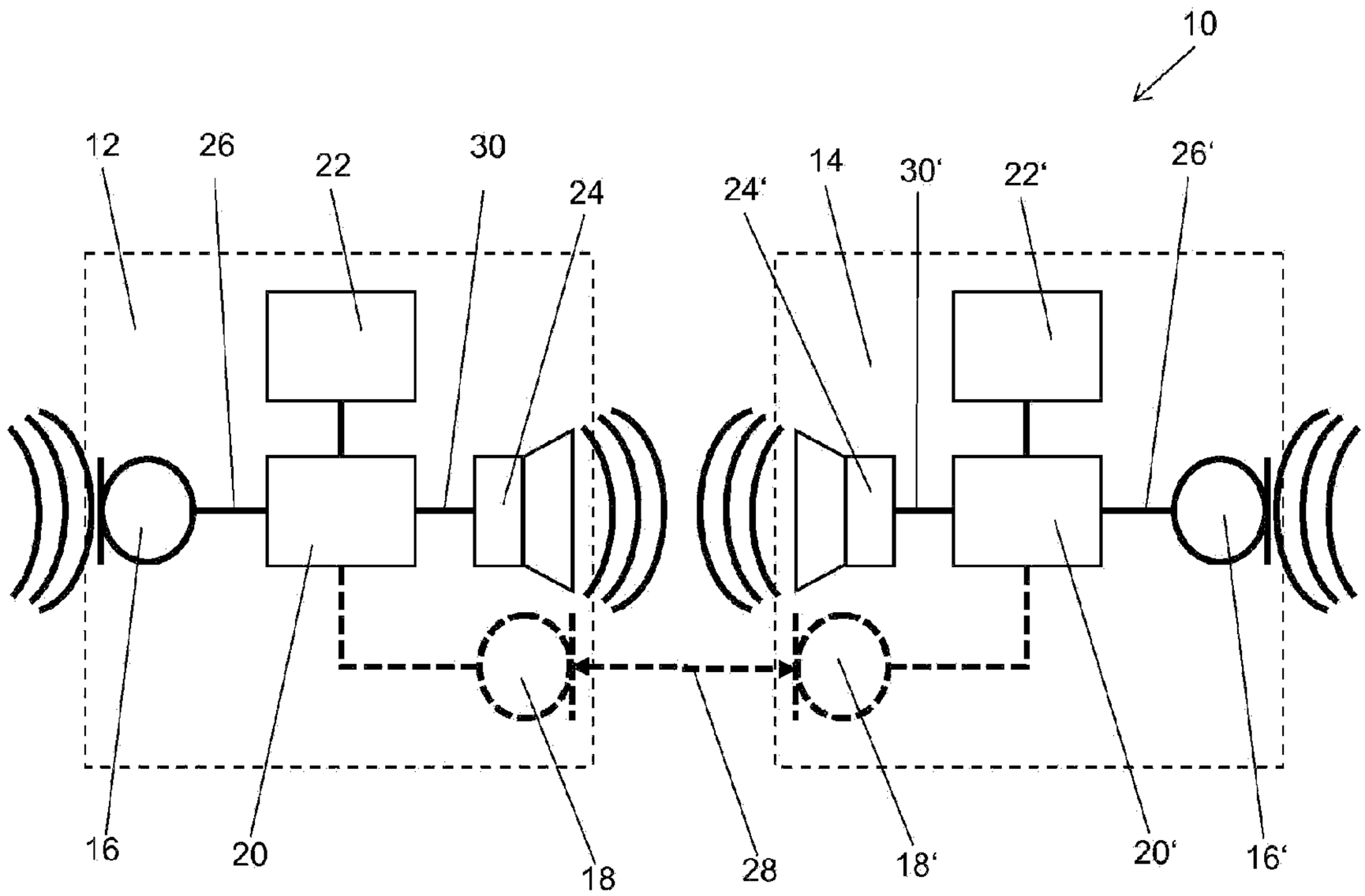


Figure 1

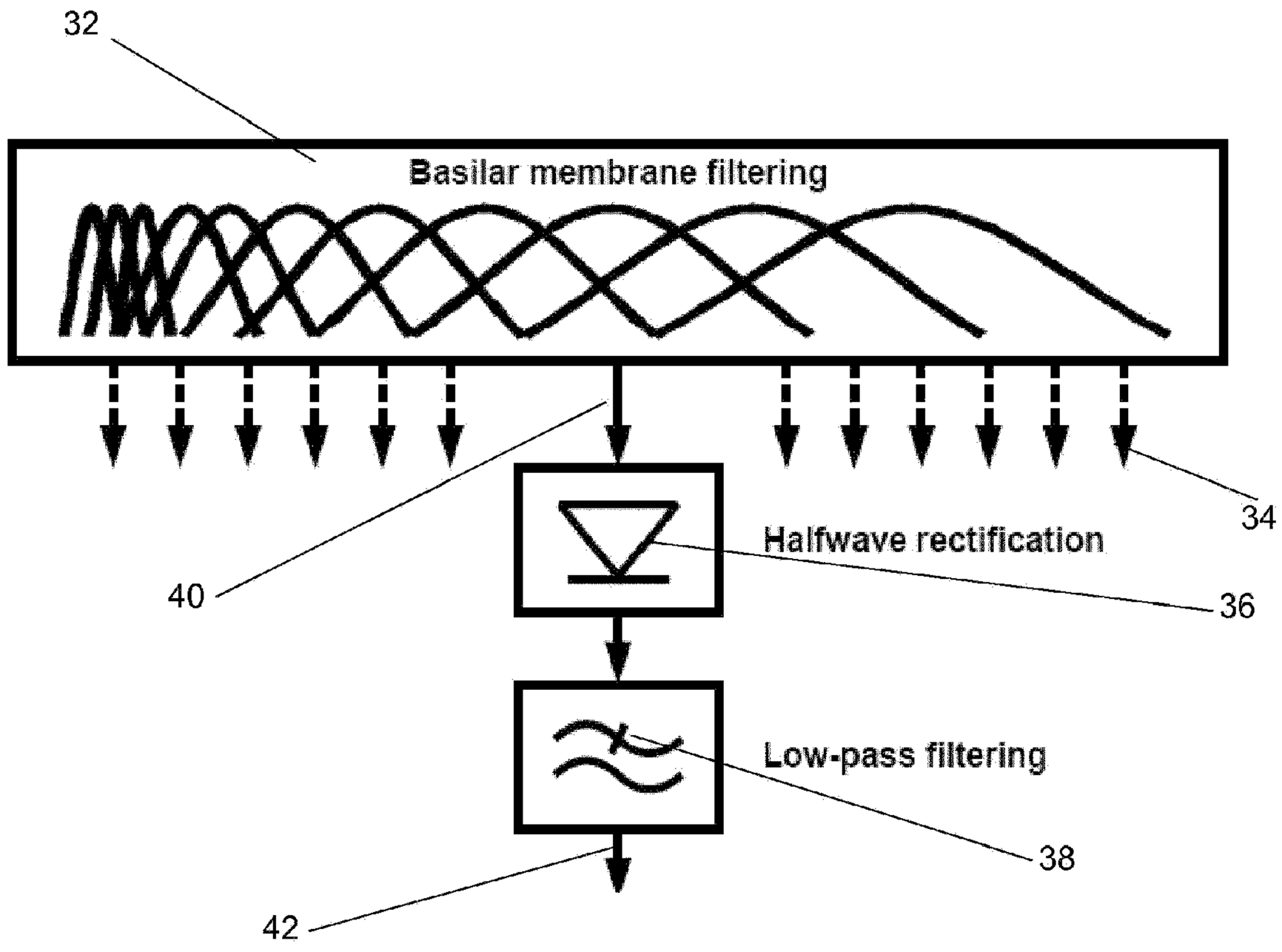

32
Basilar membrane filtering
34
Halfwave rectification
40
36
Low-pass filtering
38
42

Figure 2

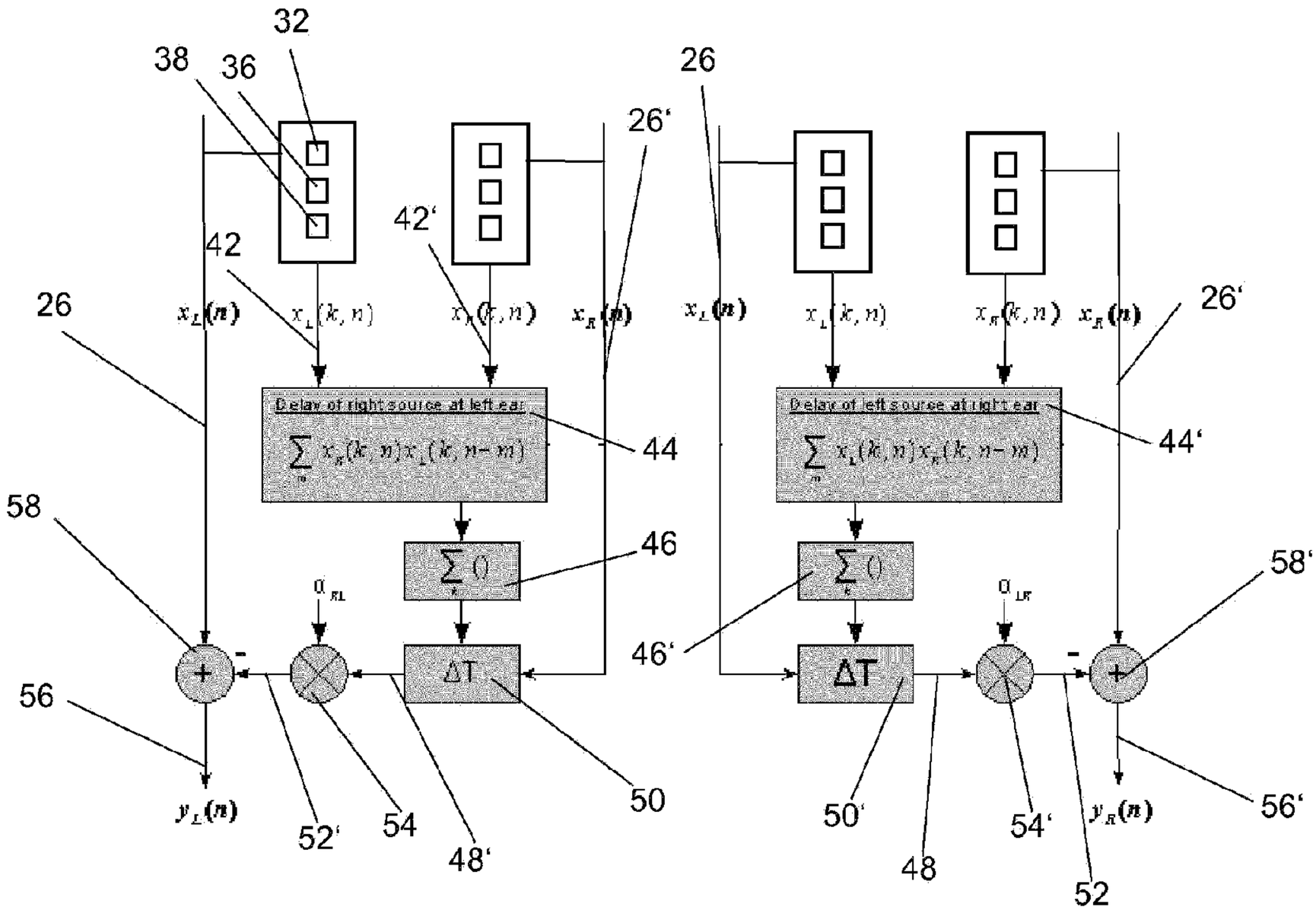

32
38
36
26
42'
26'
42
26
26'
Delay of right source at left ear
Delay of left source at right ear
44
44'
58
46
46'
58'
56
50
50'
52'
54
48'
54'
48
52
56'

Figure 3

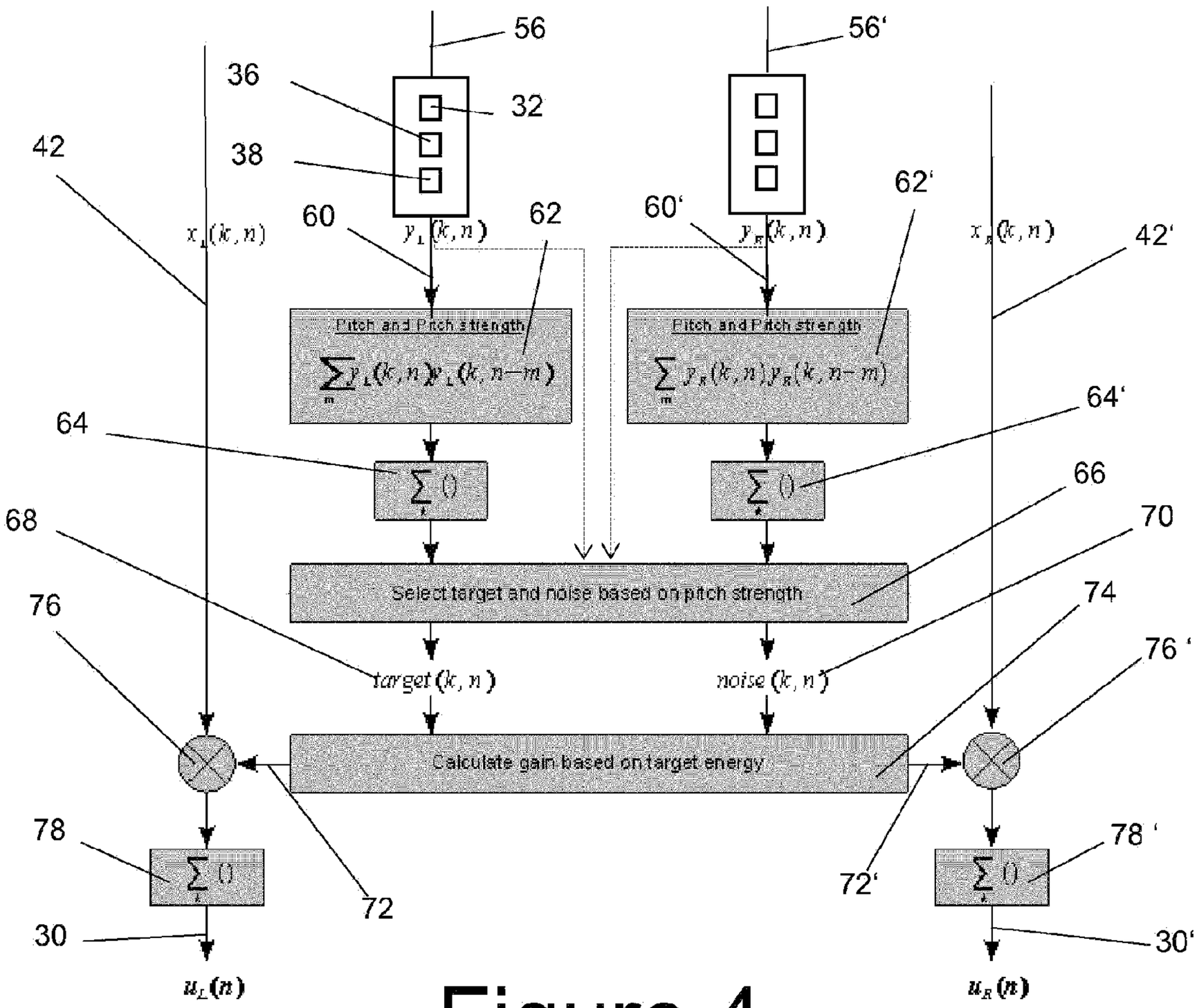

56
56'
36
32
38
42
60
62
60'
62'
42'
64
64'
66
68
70
74
76
76'
78
78'
72
72'
30
30'
Pitch and Pitch strength
Pitch and Pitch strength
Select target and noise based on pitch strength
target (k, n)
noise (k, n)
Calculate gain based on target energy
u_L(n)
u_R(n)

Figure 4

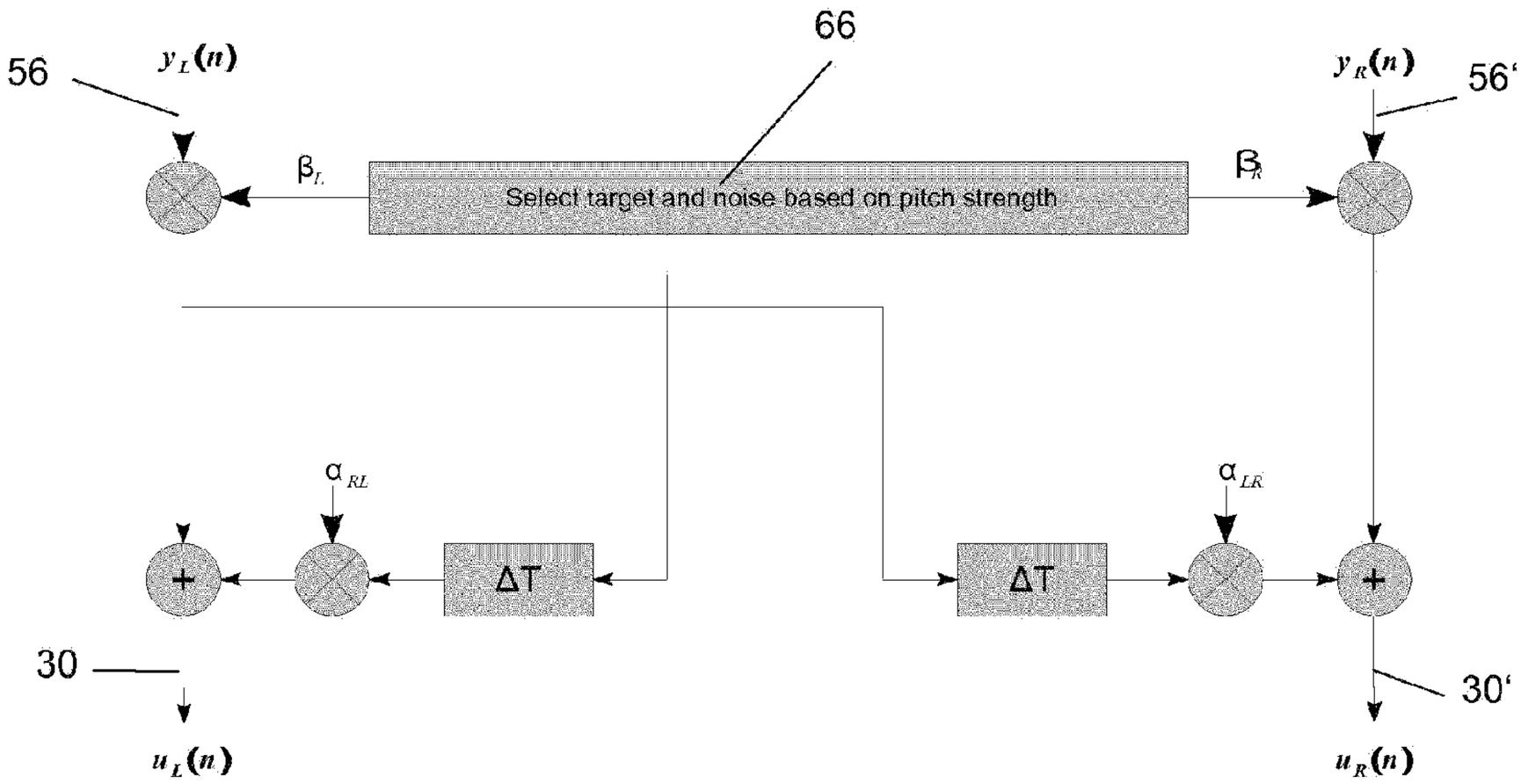

56
$y_L(n)$
66
$y_R(n)$
56'
β_L
Select target and noise based on pitch strength
β_R
α_{RL}
α_{LR}
+
ΔT
ΔT
+
30
$u_L(n)$
30'
$u_R(n)$

Figure 5