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DESCRIPTION

TECHNICAL FIELD

[0001] This disclosure relates generally to hearing assistance devices, and more particularly to frequency translation by high-frequency spectral envelope warping in hearing assistance devices.

BACKGROUND

[0002] Hearing assistance devices, such as hearing aids, include, but are not limited to, devices for use in the ear, in the ear canal, completely in the canal, and behind the ear. Such devices have been developed to ameliorate the effects of hearing losses in individuals. Hearing deficiencies can range from deafness to hearing losses where the individual has impairment responding to different frequencies of sound or to being able to differentiate sounds occurring simultaneously. The hearing assistance device in its most elementary form usually provides for auditory correction through the amplification and filtering of sound provided in the environment with the intent that the individual hears better than without the amplification.

[0003] In order for the individual to benefit from amplification and filtering, they must have residual hearing in the frequency regions where the amplification will occur. If they have lost all hearing in those regions, then amplification and filtering will not benefit the patient at those frequencies, and they will be unable to receive speech cues that occur in those frequency regions. Frequency translation processing recodes high-frequency sounds at lower frequencies where the individual's hearing loss is less severe, allowing them to receive auditory cues that cannot be made audible by amplification.

[0004] One way of enhancing hearing for a hearing impaired person was proposed by Hermansen, Fink, and Hartmann in 1993. "Hearing Aids for Profoundly Deaf People Based on a New Parametric Concept," Hermansen, K.; Fink, F.K.; Hartmann, U; Hansen, V.M., Applications of Signal Processing to Audio and Acoustics, 1993. "Final Program and Paper Summaries," 1993 IEEE Workshop on, Vol., Iss, 17-20 Oct. 1993, pp. 89-92. They proposed that a vocal tract (formant) model be constructed by linear predictive analysis of the speech signal and decomposition of the prediction filter coefficients into formant parameters (frequency, magnitude, and bandwidth). A speech signal was synthesized by filtering the linear prediction residual with a vocal tract model that was modified so that any high frequency formants outside of the range of hearing of a hearing impaired person were transposed to lower frequencies at which they can be heard. They also suggested that formants in low-frequency regions may not be transposed. However, this approach is limited in the amount of transposition that can be performed without distorting the low frequency portion of the spectrum (e.g., containing the first two formants). Since the entire signal is represented by a

formant model, and resynthesized from the modified (transposed) formant model, the entire signal may be considerably altered in the process, especially when large transposition factors are used for patients having severe hearing loss at mid and high frequencies. In such cases, even the part of the signal that was originally audible to the patient is significantly distorted by the transposition process.

[0005] In U.S. Patent 5,571,299, Melanson presented an extension to the work of Hermansen *et. al.* in which the prediction filter is modified directly to warp the spectral envelope, thereby avoiding the computationally expensive process of converting the filter coefficients into formant parameters. Allpass filters are inserted between stages in a lattice implementation of the prediction filter, and the fractional-sample delays introduced by the allpass filters determine the nature of the warping that is applied to the spectral envelope. One drawback of this approach is that it does not provide direct and complete control over the shape of the warping function, or the relationship between input frequency and transposed output frequency. Only certain input-output frequency relationships are available using this method.

[0006] In U.S. Patent 5,014,319, Leibman relates a frequency transposition hearing aid that classifies incoming sound according to frequency content, and selects an appropriate transposition factor on the basis of that classification. The transposition is implemented using a variable-rate playback mechanism (the sound is played back at a slower rate to transpose to lower frequencies) in conjunction with a selective discard algorithm to minimize loss of information while keeping latency low. This scheme was implemented in the AVR TranSonic™ and ImpaCt™ hearing aids. However, in at least one study, this variable-rate playback approach has been shown to lack effectiveness in increasing speech intelligibility. See, for example, "Preliminary results with the AVR ImpaCt Frequency-Transposing Hearing Aid," McDermott, H.J.; Knight, M. R.; *J. Am. Acad. Audiol.*, 2001 Mar.; 12 (3); 121-7 11316049 (P, S, E, B), and "Improvements in Speech Perception with use of the AVR TranSonic Frequency-Transposing Hearing Aid," McDermot, H. J.; Dorkos, V. P.; Dean, M. R.; Ching, T. Y.; *J. Speech Lang. Hear. Res.* 1999 Dec.; 42(6): 1323-35. Some disadvantages of this approach are that the entire spectrum of the signal is transposed, and that the pitch of the signal is, therefore, altered. To address this deficiency, this method uses a switching system that enables transposition when the spectrum is dominated by high-frequency energy, as during consonants. This switching system may introduce errors, especially in noisy or complex audio environments, and may disable transposition for some signals which could benefit from it.

[0007] In U.S. Patent Application Publication 2004 0264721 (issued as U.S. Patent 7,248,711), Allegro *et. al.* relate a method for frequency transposition in a hearing aid in which a nonlinear frequency transposition function is applied to the spectrum. In contrast to Leibman, this algorithm does not involve any classification or switching, but instead transposes low frequencies weakly and linearly and high frequencies more strongly. One drawback of this method is that it may introduce distortion when transposing pitched signals having significant energy at high frequencies. Due to the nonlinear nature of the transposition function (the input-output frequency relationship), transposed harmonic structures become inharmonic. This artifact is especially noticeable when the inharmonic transposed signal overlaps the spectrum

of the non-transposed harmonic structure at lower frequencies.

[0008] The Allegro algorithm is described as a frequency domain algorithm, and resynthesis may be performed using a vocoder-like algorithm, or by inverse Fourier transform. Frequency domain transposition algorithms (in which the transposition processing is applied to the Fourier transform of the input signal) are the most-often cited in the patent and scholarly literature (see for example Simpson *et. al.*, 2005, and Turner and Hurtig, 1999, U.S. Patent 6,577,739, U.S. Patent Application Publication 2004 0264721 (issued as U.S. Patent 7,248,711) and PCT Patent Application WO 0075920). "Improvements in speech perception with an experimental nonlinear frequency compression hearing device," Simpson, A.; Hersbach, A. A.; McDermott, H.J.; *Int J Audiol.* 2005 May;44(5):281-92; and "Proportional frequency compression of speech for listeners with sensorineural hearing loss," Turner, C.W.; Hurtig, R.R.; *J Acoust Soc Am.* 1999 Aug;106(2):877-86. Not all of these methods render transposed harmonic structure inharmonic, but they all share the drawback that the pitch of transposed harmonic signals are altered.

[0009] Kuk *et. al.* (2006) discuss a frequency transposition algorithm implemented in the Widex Inteo hearing aid, in which energy in the one-octave neighborhood of the highest-energy peak above a threshold frequency is transposed downward by one or two octaves (a factor of two or four) and mixed with the original unprocessed signal. "Linear Frequency Transposition: Extending the Audibility of High-Frequency Information," Francis Kuk; Petri Korhonen; Heidi Peeters; Denise Keenan; Anders Jessen; and Henning Andersen; *Hearing Review* 2006 October. As in other frequency domain methods, one drawback of this approach is that high frequencies are transposed into lower frequencies, resulting in unnatural pitch transpositions of the sound. Additional artifacts are introduced when the harmonic structure of the transposed signal overlaps the spectrum of the non-transposed harmonic structure at lower frequencies.

[0010] Therefore, an improved system for improved intelligibility without a degradation in natural sound quality in hearing assistance devices is needed.

[0011] US4051331 discloses a method for processing an audio signal received by a hearing assistance device, the method comprising filtering the audio signal to generate a high frequency filtered signal, the filtering performed at a splitting frequency; and transposing at least a portion of an audio spectrum of the filtered signal to a lower frequency range by a transposition process to produce a transposed audio signal, wherein the transposition process uses a prototype spectral peak modulated by a sinusoid having a time varying amplitude and frequency to control a center frequency and magnitude of a translated spectral feature.

[0012] EP2249587 discloses the preservation of perception relevant information in the low frequency part of an audio signal.

[0013] The invention is in the method of Claim 1 and the system of Claim 10.

[0014] Disclosed herein, among other things, is a system for frequency translation by high-

frequency spectral envelope warping in a hearing assistance device for a wearer. Furthermore a method for processing an audio signal received by a hearing assistance device as defined in claim 1 is disclosed.

[0015] In various embodiments, the pole frequencies are translated toward the knee frequency and may be done so linearly using a warping factor or non-linearly, such as using a logarithmic or other non-linear function. Such translations may be limited to poles above the knee frequency.

[0016] In various examples that are not part of the invention, the excitation signal is a prediction error signal, produced by filtering the high-pass signal with an inverse of the estimated all-pole spectral envelope. The present subject matter includes randomizing a phase of the prediction error signal, including translating the prediction error signal to the frequency domain using a discrete Fourier Transform; randomizing a phase of components below a Nyquist frequency; replacing components above the Nyquist frequency by a complex conjugate of the corresponding components below the Nyquist frequency to produce a valid spectrum of a purely real time domain signal; inverting the DFT to produce a time domain signal; and using the time domain signal as the excitation signal. It is understood that in various embodiments the prediction error signal is processed by using, among other things, a compressor, peak limiter, or other nonlinear distortion to reduce a peak dynamic range of the excitation signal. In various examples the excitation signal is a spectrally shaped or filtered noise signal.

[0017] In various embodiments the system includes combining the transposed signal with a low-pass filtered version of the audio signal to produce a combined output signal, and in some examples the transposed signal is adjusted by a gain factor prior to combining.

[0018] The system also provides the ability to modify pole magnitudes and frequencies.

[0019] In various examples, the system includes different uses of line spectral frequencies to simplify computations of the frequency translation process.

[0020] This Summary is an overview of some of the teachings of the present application and not intended to be an exclusive or exhaustive treatment of the present subject matter. Further details about the present subject matter are found in the detailed description and appended claims. The scope of the present invention is defined by the appended claims.

BRIEF DESCRIPTION OF THE DRAWINGS

[0021]

FIG. 1 is a block diagram of a hearing assistance device including a frequency translation element according to one embodiment of the present subject matter.

FIG. 2 is a signal flow diagram of a frequency translation system according to one embodiment of the present subject matter.

FIG. 3 is a signal flow diagram of a frequency translation system according to one embodiment of the present subject matter.

FIG. 4 illustrates a frequency warping function used in the frequency translation system according to one embodiment of the present subject matter.

FIGS. 5-7 demonstrate data for various frequency translations using different combinations of splitting frequency, knee frequency and warping ratio, according to various embodiments of the present subject matter.

FIGS. 8A and 8B demonstrate one example of the effect of warping on the spectral envelope using a frequency translation system according to one embodiment of the present subject matter.

FIG. 9 is a signal flow diagram demonstrating a time domain spectral envelope warping process for the frequency translation system according to one embodiment of the present subject matter.

FIG. 10 is a signal flow diagram demonstrating a frequency domain spectral envelope warping process for the frequency translation system according to one embodiment of the present subject matter.

FIG.11 is a signal flow diagram demonstrating a time domain spectral envelope warping process for the frequency translation system combining the whitening and shaping filters according to one embodiment of the present subject matter.

FIGS. 12A and 12B show magnitude and frequency plots as a function of normalized frequency, according to one embodiment of the present subject matter.

FIGS. 13A and 13B show spectral envelope ($A(k)$) roots before and after warping according to one embodiment of the present subject matter.

FIGS. 14A and 14B show roots of $P(k)$ (o's) and $Q(k)$ (x's) before and after warping according to one embodiment of the present subject matter.

FIG. 15 shows a plot of the roots of a spectral envelope constructed from warped line spectral frequencies according to one embodiment of the present subject matter.

FIG. 16 shows one example of a prototype spectral peak according to one embodiment of the present subject matter.

FIG. 17 shows two spectral peaks constructed by replicating a prototype spectral peak, such as the one in FIG. 16, according to one embodiment of the present subject matter.

FIG. 18 shows two spectral peaks constructed by replicating a prototype spectral peak, such as the one in FIG. 16, according to one embodiment of the present subject matter.

FIGS. 19A, 19B and 19C show spectra for partials having different amounts of spectral line widening due to bandwidth enhancement, according to various embodiments of the present subject matter.

DETAILED DESCRIPTION

[0022] The following detailed description of the present subject matter refers to subject matter in the accompanying drawings which show, by way of illustration, specific aspects and embodiments in which the present subject matter may be practiced. These embodiments are described in sufficient detail to enable those skilled in the art to practice the present subject matter. References to "an", "one", or "various" embodiments in this disclosure are not necessarily to the same embodiment, and such references contemplate more than one embodiment. The following detailed description is demonstrative and not to be taken in a limiting sense. The scope of the present subject matter is defined by the appended claims.

[0023] The present subject matter relates to improved speech intelligibility in a hearing assistance device using frequency translation by high-frequency spectral envelope warping. The system described herein implements an algorithm for performing frequency translation in an audio signal processing device for the purpose of improving perceived sound quality and speech intelligibility in an audio signal when presented using a system having reduced bandwidth relative to the original signal, or when presented to a hearing-impaired listener sensitive to only a reduced range of acoustic frequencies.

[0024] One goal of the proposed system is to improve speech intelligibility in the reduced-bandwidth presentation of the processed signal, without compromising the overall sound quality, that is, without introducing undesirable perceptual artifacts in the processed signal. In embodiments implemented in a real-time listening device, such as a hearing aid, the system must conform to the computation, latency, and storage constraints of such real-time signal processing systems.

HEARING ASSISTANCE DEVICE APPLICATION

[0025] In one application, the present frequency translation system is incorporated into a hearing assistance device to provide improved speech intelligibility without undesirable perceptual artifacts in the processed signal. FIG. 1 demonstrates a block diagram of a hearing assistance device including a frequency translation element according to one embodiment of the present subject matter. The hearing assistance device includes a microphone 110 which provides signals to the electronics 120. The electronics 120 provide a processed signal for speaker 112. The electronics 120 include, but are not limited to, hearing assistance device

system 124 and frequency translation system 122. It is understood that such electronics and systems may be implemented in hardware, software, firmware, and various combinations thereof. It is also understood that certain applications may not employ this exact set of components and/or arrangement. For example, in the application of cochlear implants, no speaker 112 is necessary. In the example of hearing aids, speaker 112 is also referred to as a "receiver." In the hearing aid example, electronics 120 may be implemented in different examples, including analog hardware, digital hardware, or various combinations thereof. In digital hearing aids, electronics 120 may be a digital signal processor or other form of processor. It is understood that electronics 120 in various examples may include additional devices such as memory or other circuits. In one digital hearing aid example, hearing assistance device system 124 is implemented using a time domain approach. In one digital hearing aid example, hearing assistance device system 124 is implemented using a frequency domain approach. In various examples the hearing assistance device system 124 may be programmed to perform hearing aid functions including, but not limited to, programmable frequency-gain, acoustic feedback cancellation, peak limiting, environment detection, and/or data logging, to name only a few. In hearing aid applications with rich digital signal processor designs, the frequency translation system 122 and hearing assistance device system 124 are implemented by programming the digital signal processor to perform the desired algorithms on the signal received from microphone 110. Thus, it is understood that such systems perform both frequency translation and hearing aid processing in a common digital signal processor. It is understood that such systems perform frequency translation and hearing aid processing using different processors. Variations of hardware, firmware, and software may be employed without departing from the scope of the claimed subject matter.

FREQUENCY TRANSLATION SYSTEM EXAMPLE

[0026] FIG. 2 is a signal flow diagram of a frequency translation system 122 according to one embodiment of the present subject matter. The diagram in FIG. 2 depicts a two-branch algorithm in which the spectral envelope of the signal in the high-pass branch is warped such that peaks in the spectral envelope are translated to lower frequencies. In one embodiment, the spectral envelope of the signal in the high-pass branch is estimated by linear predictive analysis, and the frequencies of the peaks in the spectral envelope are determined from the coefficients of the filter so derived. Various linear predictive analysis approaches are possible. One source of information about linear prediction is provided by John Makhoul in Linear Prediction: A Tutorial Review, Proceedings of the IEEE, Vol. 63, No. 4, April 1975. Linear prediction includes, but is not limited to, autoregressive modeling or all-pole modeling. The peak frequencies are translated to new (lower) frequencies and used to specify a synthesis filter, which is applied to the residue signal obtained by inverse-filtering the analyzed signal by the unmodified (before warping) prediction filter. The (warped) filtered residue signal, possibly with some gain applied, is combined with the signal in the lower branch (not processed by frequency translation) of the algorithm to produce the final output signal. This combination of distinct high-pass and pass-through branches with spectral envelope warping in the high-pass frequency translation branch guarantees that signals that should not be translated (for

example, low-frequency voiced speech) pass through the system without artifacts or alteration, and allows explicit and controlled balancing of the processed and unprocessed signals. Moreover, by processing a high-pass signal, instead of the full-bandwidth signal, no computational burden (linear prediction coefficients or pole frequencies, for example) is incurred due to the relatively higher-energy part of the signal that should not be translated in frequency.

[0027] The system of FIG. 2 includes two signal branches. The upper branch in the block diagram in FIG. 2 contains the frequency translation processing 220 performed on the audio signal. In this embodiment, frequency translation processing 220 is applied only to the signal in a highpass (or high bandpass) region of the spectrum passed by filter 214. The signal in the lower branch is not processed by frequency translation. The filter 210 in the lower branch of the diagram may have a lowpass or allpass characteristic, and should, at a minimum, pass all of the energy rejected by the filter in the upper branch, so that all of the spectral energy in the signal is represented in at least one of the branches of the algorithm. The processed and unprocessed signals are combined in the summing block 212 at the right edge of the block diagram to produce the overall output of the system. A gain control 230 may be optionally included in the upper branch to regulate the amount of the processed signal energy in the final output.

[0028] In one example, the filter 210 in the lower block is omitted. In one example the filter 210 is replaced by a simple delay compensating for the delay incurred by filtering in the upper processing branch. FIG. 3 shows more detail of one frequency translation system of FIG. 2 according to one embodiment of the present subject matter. In FIG. 3 the leftmost block of the processing branch of frequency translation system 322 is called a splitting filter 314. The function of the splitting filter 314 is to isolate the high-frequency part of the input audio signal for frequency translation processing. The cutoff frequency of this high-pass (or high bandpass) filter 314 is one of the parameters of the system, and we will call it the splitting frequency. The motivation for employing a splitting filter 314 in our system is to leave unaltered the low-frequency part of the audio signal, which is the part that lies *within* the limited-bandwidth region in which the signal will be presented or received, and that usually dominates the sound quality of the overall signal. Frequency translation processing is to be applied primarily to parts of the signal that would otherwise be inaudible, or fall outside of the limited available bandwidth. In speech processing applications it is intended that primarily the parts of speech having substantial high-frequency content, such as fricative and sibilant consonants, are frequency translated. Other spectral regions, such as the lower-frequency regions containing harmonic information, critical for the perceived voice quality, and the first two vowel formants, critical for vowel perception, may be unaffected by the processing, because they will be suppressed by the splitting filter 314.

[0029] In one example the frequency translation processor 320 is programmed to perform a piecewise linear frequency warping function. Greater detail is provided in FIG. 4, which depicts an input-output frequency relationship. The warping function consists of two regions: a low-frequency region 410 in which no warping is applied, and a high-frequency warping region 420,

in which energy is translated from higher to lower frequencies. The frequency corresponding to the breakpoint in this function, dividing the two regions, is called the knee point, or knee frequency 430, in the warping curve. Energy above this frequency is translated towards, but not below, the knee frequency 430. The amount by which this energy is translated in frequency is determined by the slope of the frequency warping curve in the warping region called a warping ratio. Precisely, the warping ratio is the inverse of the slope of the warping function above the knee point. In processor-based implementations, the knee point and warping ratio are parameters of the frequency translation algorithm.

[0030] The three algorithm parameters described above, the splitting frequency, the warping function knee frequency, and the warping ratio, determine which parts of the spectral envelope are processed by frequency translation, and the amount of translation that occurs. FIGS. 5 through 7 depict the frequency translation processing for three different configurations of the three parameters. The abscissa represents increasing frequency, the units on the ordinate are arbitrary. The line having large dashes represents a hypothetical input frequency envelope, and the line with small dots represents the corresponding translated spectral envelope. In FIG. 5, the splitting frequency and knee frequency are both 2 kHz, so energy in the envelope above 2 kHz is warped toward that frequency. The overall signal bandwidth is reduced and the peaks in the envelope have been translated to lower frequencies. FIG. 6 depicts the case of the splitting frequency, at 1 kHz, being lower than the knee frequency in the warping function. In this case energy above 1 kHz is processed by frequency translation, but energy below 2 kHz is not translated, so one of the peaks in the spectral envelope is translated as shown in FIG. 6. Thus, in FIG. 6, some of the energy in the processing branch, the energy between 1 kHz (the splitting frequency) and 2 kHz (the knee frequency), is not translated to lower frequencies because it is below the knee frequency. In FIG. 7, the knee frequency in the frequency warping curve is 1 kHz, lower in frequency than the splitting frequency, which remains at 2 kHz. As in FIG. 5, only energy above 2 kHz is processed, but in this case, the envelope energy is translated towards 1 kHz, so one of the peaks in the envelope is translated to a frequency lower than the splitting frequency. Thus, in FIG. 7 some energy (or part of the envelope) is translated to a region below the splitting frequency. Consequently, before translation the processing branch included only spectral peaks above the splitting frequency, and after translation a peak was present at a frequency below the splitting frequency. The examples provided in FIGS. 5-7 show how the various settings of the algorithm parameters translate peaks in the spectral envelope. In various embodiments, these figures depict changes to the signal in the highpass branch only. In such embodiments, there is no overall signal bandwidth reduction in general, because the processed signal is ultimately mixed in with the original signal.

[0031] The frequency warping function governs the behavior of the frequency translation processor, whose function is to alter the shape of the spectral envelope of the processed signal. In such embodiments, the pitch of the signal is not changed, because the spectral envelope, and not the fine structure, is affected by the frequency translation process. This process is depicted in FIGS. 8A and 8B, which shows the spectral envelope for a short segment of speech before (FIG. 8A) and after (FIG. 8B) frequency translation processing. The

spectral envelope is estimated for a short-time segment of the input signal by a method of linear prediction (also known as autoregressive modeling), in which a signal is decomposed into an all-pole (recursive, or autoregressive) filter describing the spectral envelope of the signal, and a whitened (spectrally-flattened) excitation signal that can be processed by the all-pole filter to recover the original signal. The frequencies of the filter's complex pole pairs determine the location of peaks in the spectral envelope. There are three peaks in the spectral envelope depicted in FIGS 8A and 8B, corresponding to three pairs of poles (six non-trivial filter coefficients) in the estimated all-pole filter. Consequently, the number of coefficients used in the estimation of the spectral envelope is a parameter of the algorithm.

[0032] In one example of the present system a whitened excitation signal, derived from linear predictive analysis, is processed using a warped spectral envelope filter to construct a new signal whose spectral envelope is a warped version of the envelope of the input signal, having peaks above the knee frequency translated to lower frequencies. In one embodiment, the peak frequencies are computed directly from the values of the complex poles in the filter derived by linear prediction. In one example the peak frequencies are estimated by examination of the frequency response of the filter. Other approaches for determining the peak frequencies are possible without departing from the scope of the claimed subject matter.

[0033] By translating the peak frequencies according to the frequency warping function described above, a new warped spectral envelope is specified which is used to determine the coefficients of the warped spectral envelope filter. In one embodiment, the filter pole frequencies can be modified directly, so that the spectral envelope described by the filter is warped, and peak frequencies above the knee frequency (such as 2 kHz shown in FIGS. 8A and 8B) in the warping function are translated toward, but not below, that frequency. It is understood that in some cases, two filter poles can be close together in frequency, creating a peak in the spectral envelope at a frequency that is different from the two pole frequencies. It is understood that various approaches to translating peak frequencies can be applied. In one example, new pole frequencies are specified to produce a desired translation of envelope peak frequencies. In one example, a new envelope peak frequency is specified. Other approaches are possible without departing from the scope of the claimed subject matter.

[0034] The whitened excitation signal, derived from linear predictive analysis, may be subjected to further processing to mitigate artifacts that are introduced when the high-frequency part of the input signal contains very strong tonal or sinusoidal components. For example, the excitation signal may be made maximally noise-like (and less impulsive) by a phase randomization process. This can be achieved in the frequency domain by computing the discrete Fourier transform (DFT) of the excitation signal, and expressing the complex spectrum in polar form (magnitude and phase, or angle). The phase of components at and below the Nyquist frequency (half the sampling frequency) are replaced by random values, and the components above the Nyquist frequency are made equal to the complex conjugate of corresponding (mirrored about the Nyquist component) components below the Nyquist frequency, so that the representation corresponds to a real time domain signal. This frequency domain representation is then inverted to obtain new excitation signal.

[0035] In various alternative examples that are not part of the invention, the excitation signal may be replaced by a shaped (filtered) noise signal. The noise may be shaped to behave like a speech-like spectrum, or may be shaped by a highpass filter, and possibly using the same splitting filter used to isolate the high-frequency part of the input signal. In such an implementation, it is generally not necessary to compute the excitation (prediction error) signal in the linear predictive analysis stage.

[0036] In other alternative examples, the excitation signal may be subjected to dynamics processing, such as dynamic range compression or limiting, or to non-linear waveform distortion to reduce its impulsiveness, and the artifacts associated with frequency transposition of signals with strongly tonal high-frequency components.

[0037] The output of the frequency translation processor, consisting of the high-frequency part of the input signal having its spectral envelope warped so that peaks in the envelope are translated to lower frequencies, and optionally scaled by a gain control, is combined with the original, unmodified signal to produce the output of the algorithm.

[0038] The present system provides the ability to govern in very specific ways the energy injected at lower frequencies according to the presence of energy at higher frequencies.

TIME DOMAIN SPECTRAL ENVELOPE WARPING EXAMPLE

[0039] FIG. 9 shows a time domain spectral envelope warping process according to one embodiment of the present subject matter. It is understood that this example is not intended to be limiting or exclusive, but rather demonstrative of one way to implement a time domain warping process.

[0040] In the time domain process of FIG. 9, sound is sampled from a microphone or other sound source ($x(t)$) and provided to the spectral envelope warping system 900. The input samples are applied to a linear prediction analysis block 903 and a finite-impulse-response filter 904 ("FIR filter 904"). The outputs of the linear prediction analysis block 902 are filter coefficients (h_k) which are used by the FIR filter 904 to filter the input samples ($x(t)$) to produce the prediction error, or excitation signal, $e(t)$. The filter coefficients (h_k) are used to find polynomial roots (\tilde{P}_k) 905 which are then warped to provide warped poles ($\{\tilde{\tilde{P}}_k\}$) 907. The excitation signal, $e(t)$, and warped poles ($\{\tilde{\tilde{P}}_k\}$) are used by an all pole filter 908, such as a biquad filter arrangement, to filter the excitation signal with the warped all pole filter. The resultant output is a sampled warped spectral envelope signal ($\{\tilde{x}(t)\}$).

[0041] It is understood that variations in process order and particular filters may be substituted in systems without departing from the scope of the claimed subject matter.

FREQUENCY DOMAIN SPECTRAL ENVELOPE WARPING EXAMPLE

[0042] FIG. 10 shows a frequency domain spectral envelope warping process according to one embodiment of the present subject matter. It is understood that this example is not intended to be limiting or exclusive, but rather demonstrative of one way to implement a frequency domain warping process.

[0043] In the frequency domain process of FIG. 10, sound is sampled from a microphone or other sound source ($x(t)$) and converted into frequency domain information, such as subbands ($X(w_k)$), before it is provided to the spectral envelope warping system 1000. One such conversion approach is the use of a fast Fourier Transform (FFT) 1001. The input sub-band ($X(w_k)$) samples are applied to

a spectral domain pole estimation block 1003 to perform spectral domain pole estimation and to a divider 1004. "Linear Prediction: A Tutorial Review", John Makhoul, Proceedings of the IEEE, Vol. 63, No. 4, April 1975. The spectral domain pole estimation block 1003 is used to find polynomial roots (P_k) which are then converted into a complex frequency response $H(w_k)$ by process 1005. The input sub-band signals $X(w_k)$ are divided by the complex frequency response $H(w_k)$ by divider 1004 to whiten the spectrum of the input sub-band signals $X(w_k)$ and to produce a complex sub-band prediction error, or complex sub-band excitation signal, $E(w_k)$. The polynomial roots (P_k) are then warped to provide warped poles ($\{P_k\}$) 1007. The warped poles ($\{P_k\}$) are converted to a complex frequency response ($H(w_k)$) 1009.

[0044] The complex sub-band excitation signal, $E(w_k)$, and complex frequency response $\{H(w_k)\}$ are multiplied 1010 to provide a sampled warped spectral envelope signal in the frequency domain $\{X(w_k)\}$. This sampled warped spectral envelope signal in the frequency domain $\{X(w_k)\}$ can be further processed in the frequency domain by other processes and ultimately converted into the time domain for transmission of processed sound.

Examples of Combined Whitening and Shaping Filters

[0045] In some examples, computational savings can be achieved by combining the application of the all-zero FIR filter, to generate the prediction error signal, and the application of the all-pole warped spectral envelope filter to the excitation signal, into a single filtering step.

[0046] The all-pole spectral envelope filter is normally implemented as a cascade (or sequence) of second-order filter sections, so-called biquad sections or biquads. Those practiced in the art will recognize that, for reasons of numerical stability and accuracy, as well as efficiency, high-order recursive filters should be implemented as a cascade of low-order filter sections. In the implementation of an all-pole filter, each biquad section has only two poles in its transfer functions, and no (non-trivial) zeros. However, the zeros in the FIR filter can be

implemented in the biquad sections along with the spectral envelope poles, and in this case, the FIR filtering step in the original frequency translation algorithm can be eliminated entirely. An example is provided by the system 1100 in FIG. 11.

[0047] In FIG. 11, input samples $x(t)$ are provided to the linear prediction block 1103 and biquad filters (or filter sections) 1108. The output of linear prediction block 1103 is provided to find the polynomial roots 1105, P_k . The polynomial roots P_k , are provided to biquad filters 1108 and to the pole warping block 1107. The roots P_k specify the zeros in the biquad filter sections. The resulting output of pole warping block 1107, $\{P_k\}$, is applied to the biquad filters 1108 to produce the warped output $\{x(t)\}$. The warped roots $\{P_k\}$ specify the poles in the biquad filter sections.

[0048] In one example, the zeros corresponding to (unwarped) roots of the predictor polynomial should be paired in a single biquad section with their counterpart warped poles in the frequency translation algorithm. Since not all poles in the spectral envelope are transformed in the frequency translation algorithm (only complex poles above a specified knee frequency), some of the biquad sections that result from this pairing will have unity transfer functions (the zeros and unwarped poles will coincide). Since the application of these sections ultimately has no effect on a signal, they can be omitted entirely, resulting in computational savings and improved filter stability.

[0049] In the present frequency translation algorithm, the highpass splitting filter makes poles on the positive real axis uncommon, but it frequently happens that poles are found on the negative real axis (poles at the Nyquist frequency, or half the sampling frequency) and these poles should not be warped, but should rather remain real poles (at the Nyquist frequency) in the warped spectral envelope. Moreover, it may happen that a pole is found below the knee frequency in the warping function, and such a pole need not be warped. Poles such as these whose frequencies are not warped can be omitted entirely from the filter design. In the case of a predictor of order 8, for example, if one pole pair is found on the negative real axis, a 25% savings in filtering costs can be achieved by omitting one second order section. If additionally one of the poles is below the knee frequency, the savings increases to 50%.

[0050] In addition to achieving some computational savings, this modification may make the biquad filter sections more numerically stable. In some examples, for reasons of numerical stability and accuracy, filter sections including both poles and zeros are implemented, rather than only poles.

[0051] It is understood that the system of FIG. 11 can be implemented in the frequency domain by combining the frequency response $H(w_k)$ and the warped frequency response $\{H(w_k)\}$ of FIG. 10 before performing the multiply 1010. Other frequency domain variations are possible without departing from the scope of the claimed subject matter.

[0052] In various examples, the processes for performing frequency translation depicted in the

block 122 of FIG. 1 can be performed using different approaches. Some examples provide less computational cost associated with the core frequency translation algorithm than others. In various examples, a method is employed for warping the parameters of the spectral envelope that does not require that the predictor polynomial to be factored to identify its roots. In the preceding approaches, the identification of spectral envelope poles requires finding the roots of the polynomial described by the predictor coefficients (for example, see block 905 of FIG. 9). Arbitrary polynomial roots are found using one of a variety of successive approximation algorithms, such as the Newton-Raphson algorithm or Laguerre's method. These algorithms may be more costly to implement, may be more sensitive to numerical errors and may have convergence issues or give erroneous results.

[0053] One approach that eases computational complexity is to find the line spectral frequencies that describe the predictor polynomial $A(k)$. They are the angles of the roots of the palindromic and anti-palindromic polynomials defined by:

$$P(m) = A(m) + A(M+1-m),$$

and

$$Q(m) = A(m) - A(M+1-m)$$

for $m=0..M$, where M is the order of the polynomial $A(k)$, and $A(M+1)$ is equal to 0. The roots of these polynomials are guaranteed to lie on the unit circle in the complex plane, and therefore can be found using one-dimensional search techniques (rather than two dimensional searching, as is necessary to find the roots of $A(k)$). The original polynomial can be reconstructed as:

$$A(m) = (P(m) + Q(m))/2$$

[0054] The polynomials P and Q have at least two advantages over the predictor polynomial A . One advantage is that they are less sensitive to quantization errors. The corruption of the coefficients that occurs in quantization has little effect on the stability or shape of the polynomial function, whereas small errors in the coefficients of A may introduce large distortions in the spectral envelope, and may make the all-pole filter unstable (may move a pole outside the unit circle). Moreover, all the coefficients of P and Q are approximately equally sensitive to errors, whereas in the polynomial A , the higher order coefficients are much more sensitive to errors.

[0055] Another advantage that motivates their use in spectral envelope warping, is that all of the roots of both P and Q are on the unit circle in the Z -plane. For speech coding, this is an advantage, because it means that only the root frequencies need to be stored and transmitted (hence the term "line spectral frequencies"), the magnitudes are always unity. In our application, this property implies that the roots of these polynomials are very much easier to find than those of A itself. For example, the roots can be identified as the zeros in the magnitude of the discrete Fourier transform (or its efficient implementation, the FFT) of the polynomial coefficients. In this way, the precision with which the roots are found can be easily traded against computational cost through the length of the DFT (a longer DFT gives more precise root frequencies at the cost of more computation). Other one-dimensional search

techniques can be employed to find the roots of the polynomials P and Q, since they are known to lie on the line that describes the unit circle in the complex plane. Such techniques for estimating the line spectral frequencies have been shown to be very efficient, and in the case of low-order polynomials, well-known closed-form solutions exist for computing the roots (such as the quadratic formula for computing roots of a second-order polynomial).

[0056] In this approach the process of spectral envelope warping is carried out in the line spectral domain, by transforming the line spectral frequencies, rather than the predictor polynomial root frequencies.

[0057] FIGS 12A and 12B show the magnitude and phase response of a spectral envelope having three prominent peaks. The poles of the corresponding all-pole filter are shown on the Z-plane plot of FIG 13A. The Z-plane plot of FIG. 13B shows the poles in the warped all-pole filter that would result from warping by a factor of 2 all poles in the original polynomial having frequency greater than $\pi/10$. The normalized (to the range 0...1) frequencies before warping are:

0.0670	0.2445	0.6457
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and after warping are

0.0670	0.1722	0.3729
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[0058] FIGS. 14A and 14B show the roots of the corresponding polynomials P(k) and Q(k) before and after warping. The normalized frequencies for the polynomials P(k) and Q(k) are:

P: 0.0668	0.2410	0.6248	1.0000
Q: 0	0.1402	0.2907	0.6569

before warping, and

P: 0.0658	0.1719	0.3667	1.0000
Q: 0	0.1189	0.2343	0.4061

after warping. Clearly, the frequencies of the roots of P(k) are quite closely related to the frequencies of the poles of A(k), and therefore they undergo a very similar transformation. Thus, if a suitable transformation of the root frequencies of Q(k) can be identified, then spectral envelope warping can be performed on the line spectral pairs, which are easy to find, rather than the poles of the predictor polynomial itself.

[0059] Since the frequencies of the roots of P(k) correspond to the frequencies of the roots of A(k), it follows that the frequencies of the roots of Q(k) must correspond in some way to the magnitudes of the roots of A(k) (recall that the magnitudes of the roots of both P(k) and Q(k) are always unity). This relationship is found through the so-called "difference parameters," the difference between the frequencies of the roots of P(k) and the nearest (in frequency) root of Q(k). The difference parameters for the example polynomials can be found to be:

0.0668	0.0497	0.0321	0.3431
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before warping, and

0.0531	0.0530	0.0394	0.5939
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after warping. It is known that smaller values of the difference parameters correspond to sharper peaks in the spectral envelope, and larger values to broader peaks. (The peaks in this example were all chosen to be fairly sharp to make them easier to see.) Note that the difference parameters are not much affected by the warping process.

[0060] In order to preserve the bandwidth of the spectral peaks, one could attempt to preserve, as nearly as possible, the difference parameters in the warping process, transforming only the frequencies of the roots of $P(k)$, and re-computing the frequencies of the roots of $Q(k)$ from the difference parameters. In some applications, it may not be considered necessary to preserve the original peak bandwidths, and in such cases, suitable difference parameters can be chosen arbitrarily, or chosen to satisfy some other properties of the warped spectral envelope (for example, they may be chosen to avoid unnaturally sharp peaks in the spectral envelope). FIG. 15 shows the Z-plane plot of the roots of an all-pole spectral envelope constructed from the warped roots of $P(k)$ and using difference parameters all chosen equal to 0.15. The normalized frequencies of the poles are found to be:

0.0671	0.1733	0.3758
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which is in good agreement with the frequencies of the poles obtained through the original warping procedure.

[0061] Various warping approaches are possible without departing from the scope of the claimed subject matter. In one approach, the line spectral frequencies are warped in the same way as the linear prediction frequencies. This has the effect of sharpening all of the poles of the reconstructed polynomial (moving them closer to the unit circle). In one alternative approach, the difference between the line spectral frequencies that bracket a pole are preserved in the warping. This tends to preserve the shape of the peaks in the spectral envelope, but can introduce problems with surrounding line spectral frequencies. This method highlights the added benefit of omitting extra line spectral frequencies from the warped set.

[0062] Another variation includes implementing only the spectral envelope peak finding function in the line spectral frequency domain. This can be done by computing the line spectral frequencies from $B(n)$, estimating poles or biquad coefficients from the line spectral frequencies, and performing warping of the poles or biquad coefficients as set forth in the earlier embodiments.

[0063] Computing line spectral frequencies is relatively computationally quick and efficient compared to the earlier methods of finding roots of the LPC polynomial. The line spectral frequencies are not exactly the roots or poles of the spectral envelope, but pairs of line spectral frequencies bracket spectral envelope poles. Larger magnitude poles are more tightly bracketed by pairs of line spectral frequencies. In various applications, spectral envelope peaks are translated by translating the corresponding line spectral frequencies. Peaks can be sharpened by moving the corresponding line spectral frequencies closer together. In various applications, line spectral frequencies that do not bracket a pole can be eliminated.

[0064] It is understood that one variation of the present process includes, but is not limited to:

performing linear prediction on the input signal to get coefficients, h_k

obtaining line spectral frequencies from the coefficients h_k ;

obtaining from the line spectral frequencies an estimate of the roots of the predictor polynomial described by the coefficients h_k ;

warping the resulting estimated roots; and

filtering the resulting input signal with a filter having the transfer function $H(n) = B(n)/A(n)$,

where $B(n)$ are the coefficients of a polynomial having roots equal to those estimated from the line spectral frequencies and $A(n)$ are coefficients of a polynomial having roots equal to the warped estimated roots (found at, for example, block 908 of FIG. 9).

[0065] It is understood that one variation of the present process includes, but is not limited to:

performing linear prediction on the input signal to get coefficients, h_k

obtaining line spectral frequencies from the coefficients h_k ;

warping the line spectral frequencies; and

filtering the resulting input signal with a filter having the transfer function $H(n) = B(n)/A(n)$,

where $B(n)$ are the coefficients of the predictor polynomial (the coefficients h_k for at, for example, block 904 of FIG.9) and $A(n)$ are coefficients of a polynomial constructed from the warped line spectral frequencies.

[0066] In this variation, an N-order ARMA filter can be implemented directly, without conversion to biquad sections. In a variation of this approach, when constructing the warped line spectral frequencies some of the frequencies that do not correspond to poles can be optionally eliminated. This creates an $A(n)$ of lower order than $B(n)$. Further variations can remove the corresponding line spectral frequencies from the non-warped set to reduce the order of $B(n)$.

[0067] It is understood that one variation of the present process includes a hybrid approach, which includes, but is not limited to:

performing linear prediction on the input signal to get coefficients, h_k

obtaining line spectral frequencies from the coefficients h_k ;

warping the line spectral frequencies;

filtering the input signal with a FIR filter having coefficients h_k (as shown, for example, in block 904 in FIG. 9) to obtain a whitened excitation signal; and

filtering the whitened excitation signal (for example, $e(t)$ in FIG. 9) with a IIR filter having coefficients $A(n)$, where $A(n)$ are coefficients of a polynomial constructed from the warped line spectral frequencies.

[0068] It is understood that variations in process order and particular conversions may be substituted in systems without departing from the scope of the claimed subject matter.

[0069] The present subject matter includes a method for processing an audio signal received by a hearing assistance device, including: filtering the audio signal to generate a high frequency filtered signal, the filtering performed at a splitting frequency; transposing at least a portion of an audio spectrum of the filtered signal to a lower frequency range by a transposition process to produce a transposed audio signal; and summing the transposed audio signal with the audio signal to generate an output signal, wherein the transposition process includes: estimating an all-pole spectral envelope of the filtered signal from a plurality of line spectral frequencies; applying a warping function to the all-pole spectral envelope of the filtered signal to translate the poles above a specified knee frequency to lower frequencies, thereby producing a warped spectral envelope; and exciting the warped spectral envelope with an excitation signal to synthesize the transposed audio signal. It also provides for the estimation of the line spectral frequencies estimated from a set of linear prediction coefficients. It also provides for application of warping functions to the line spectral frequencies. It also provides for scaling the transposed audio signal and summing the scaled transposed audio signal with the audio signal. It is contemplated that the filtering includes, but is not limited to high pass filtering or high bandpass filtering. In various embodiments, the estimating includes performing linear prediction. In various examples the estimating is done in the frequency domain. In various examples the estimating is done in the time domain.

[0070] In various embodiments, the pole frequencies are translated toward the knee frequency and may be done so linearly using a warping factor or non-linearly, such as using a logarithmic or other non-linear function. Such translations may be limited to poles above the knee frequency.

[0071] In various examples that are not part of the invention, the excitation signal is a prediction error signal, produced by filtering the high-pass signal with an inverse of the estimated all-pole spectral envelope. The present subject matter in various examples includes randomizing a phase of the prediction error signal, including translating the prediction error signal to the frequency domain using a discrete Fourier Transform; randomizing a phase of components below a Nyquist frequency; replacing components above the Nyquist frequency by a complex conjugate of the corresponding components below the Nyquist frequency to

produce a valid spectrum of a purely real time domain signal; inverting the DFT to produce a time domain signal; and using the time domain signal as the excitation signal. It is understood that the prediction error signal is processed by using, among other things, a compressor, peak limiter, or other nonlinear distortion to reduce a peak dynamic range of the excitation signal. In various examples that are not part of the invention the excitation signal is a spectrally shaped or filtered noise signal.

[0072] In various embodiments the system includes combining the transposed signal with a low-pass filtered version of the audio signal to produce a combined output signal, and in some examples the transposed signal is adjusted by a gain factor prior to combining.

[0073] The system also provides the ability to modify pole magnitudes and frequencies.

[0074] In various examples, the system includes different uses of line spectral frequencies to simplify computations of the frequency translation process.

[0075] In various embodiments, a system and method of frequency translation uses additive synthesis of frequency translation spectra. This approach enhances frequency translation processing to provide greater variation in the translated spectral envelope with different parameter settings, and more distinct and less confusable translated spectra for different translated sounds. The method includes, but is not limited to one or more of the following aspects: additive synthesis of spectral peaks by means of a single, modulated prototype spectrum having fixed spectral shape; fine control of the translated spectral envelope; and/or independence of the spectral envelope excitation signal from the input signal, specifically, from the low frequency input spectrum.

[0076] A process from musical sound synthesis, known as *bandwidth-enhanced additive synthesis* (see for example, K. Fitz and L. Haken, "A new algorithm for bandwidth association in bandwidth-enhanced additive sound modeling," in Proc. ICMC, 2000, and see also K. Fitz, "The reassigned bandwidth-enhanced method of additive synthesis," Thesis (PhD). University Of Illinois At Urbana-Champaign, 2000, is adapted for the present system to synthesize translated spectral features. The envelopes of individual tones in a sinusoidal synthesis model are modulated by narrowband noise to produce noisy sinusoids. This allows noisy sounds, like flutes and clarinets, to be synthesized using a simple additive algorithm.

[0077] In this approach the narrowband noise is modulated by a tone to translate it to a desired spectral region. The frequency of the tone becomes the center frequency of a synthesized noise band, and the amplitude of the tone becomes the peak spectral magnitude of the noise band. A noise signal is filtered to obtain a narrowband, lowpass noise signal. This narrowband noise can be considered a "prototype" spectral peak that will be replicated at the desired feature center frequencies, In various examples, the approach is performed in the time domain. In various examples, the approach is performed in the frequency domain or subband domain. In limited bandwidth implementations of the present subject matter some computation can be conserved in its generation. In various embodiments, random samples can be

generated at a highly decimated rate and upsampled by simple interpolation or smoothing. Some embodiments can require that only a single fixed-coefficient prototype lowpass filter be generated (rather than updating filter coefficients at each block for each spectral peak, as in other approaches), and since the excitation for that filter is already lowpass (being generated at a lower rate), the stopband constraints on the filter can be relaxed (allowing the use of a cheaper, lower-order filter). An example prototype spectral peak is shown in Figure 16. The prototype spectral peak is modulated (multiplied) by a sinusoid having time-varying amplitude and frequency to control the center frequency and magnitude of the translated spectral feature. Multiplication by a sum of two or more sinusoids is performed to render multiple translated peaks simultaneously in various embodiments. In some embodiments, the computation required does *not* scale with the number of spectral peaks rendered (that is, it grows more slowly than the number of rendered peaks), since the synthesis of the prototype spectral peak consumes most of the computation, and the modulation is relatively inexpensive (one example is a table lookup and a couple of multiplies per sample). Figures 17 and 18 show two two-peak spectra generated by modulating the prototype shown in Figure 16.

[0078] In addition to potential computational efficiencies achieved by synthesizing a single prototype spectral peak, the present subject matter offers other potential advantages. In various embodiments, the synthesized spectrum does not depend on the presence of low frequency energy in the input signal. In other approaches, the algorithm suppresses low frequency energy in the input with the splitting filter, and then boosts it again with the translated spectral envelope filter. This has consequences for the sound quality of the translated sound, since one can consider that the suppressed low frequency noise was represented in a small number of bits before being amplified by the spectral envelope filter. Under such other approaches, this situation can be aggravated when the input signal has little low frequency energy to begin with. Some dependence on the input signal level is useful, but an /s/ sound, for example, may have very little low frequency energy to excite the spectral envelope filter. In contrast, the embodiments of the present subject matter allow more precise and reliable control over the frequency at which energy is introduced, and the amount of energy introduced, because the prototype spectral peak is synthesized, and does not need to be generated from the input signal.

[0079] Another consequence is that the dominant spectral effect appears just below the splitting filter cutoff frequency. Although the algorithm produces audible changes, its effect is mostly insensitive to input or algorithm parameters because the synthesis is confined mainly to the rising part of the splitting filter response. This can make the algorithm somewhat difficult and confusing to fit.

[0080] In such embodiments of the present approach, the bandwidth and shape of the rendered spectral peaks does not vary with the magnitude of the translated peak. Two (or more) peaks at different and arbitrary magnitudes can be produced without changing the shape of either (as shown in Figure 17 and Figure 18). This allows more distinctive spectra to be produced for different translated sounds. In the present approach, in certain embodiments, the prototype filter need only be computed once, so a more advantageous filter design can be

used.

Relation to bandwidth-enhanced additive synthesis

[0081] In bandwidth-enhanced additive synthesis (see for example, K. Fitz and L. Haken, "A new algorithm for bandwidth association in bandwidth-enhanced additive sound modeling," in Proc. ICMC, 2000, and see also K. Fitz, "The reassigned bandwidth-enhanced method of additive synthesis," Thesis (PhD). University Of Illinois At Urbana-Champaign, 2000), sinewaves in an additive synthesizer are amplitude-modulated using narrowband noise to add noise energy to the synthesized sound in the frequency neighborhood of the sinewave, using a synthesis equation like

$$y_t = (A_t + I_t \zeta_t) \cos(\omega_t t + \theta)$$

in which A_t is the time-varying carrier amplitude, I_t is the time-varying modulation index, ζ_t is the bandlimited noise modulator, and ω_t is the time-varying carrier (center) frequency. The amount of noise added to the spectrum is governed by the modulation index, and when I_t is zero, the synthesized component is a pure sinusoid at frequency ω_t (no noise is introduced at that component's frequency). The amplitude of the pure tone is governed by the carrier amplitude A_t . The effect is demonstrated in Figure 19 (copied from K. Fitz and L. Haken, "A new algorithm for bandwidth association in bandwidth-enhanced additive sound modeling," in Proc. ICMC, 2000).

[0082] Figure 19 shows spectra for partials having different amounts of spectral line widening due to bandwidth enhancement. Figure 19(a) corresponds to a sinusoid with no line widening, Figure 19(b) corresponds to a sinusoid with a moderate amount of line widening, and Figure 19(c) corresponds to a sinusoid with a large amount of line widening.

[0083] In the present approach, the intended *application* and *purpose* are different, and these differences motivate a somewhat different implementation. In the music synthesis application, stochastic modulation was used to add a little noisiness to a sound that was otherwise composed of pure sinusoids. In the present method, the tone at frequency ω_t (see the above equation) is not desired in the frequency translation output. This approach is not interested in hearing noisy sinewaves, so the implementation is not necessarily balancing noise energy and tonal energy. The present method in various embodiments uses modulation to place noise energy at a specific desired center frequency. Therefore, the modulation index is always unity and the carrier is always suppressed, because the pure sinusoid is not desired in the frequency translation output. The synthesis equation for each component is therefore reduced to modulation (multiplication) of narrowband noise by at tone at the desired center frequency, as in

$$y_t = M_n \zeta_t \cos(\omega_n t).$$

Here, M_n and ω_n are the magnitude and center frequency of the translated spectral envelope peak, which are estimated and updated each block in various embodiments (rather than each sample), and ζ_t is the prototype bandlimited noise. The final synthesized signal, the output of

the frequency translation algorithm, is a sum of these modulated noise components, as in

$$Y_t = \sum_{i=0}^K M_{i,n} \zeta_t \cos(\omega_{i,n} t)$$

where K is the number of estimated spectral envelope peaks (2 for example in certain other frequency translation algorithms).

In such embodiments of the present method, modulation is an efficient way to synthesize a noise spectrum, by adding up bandlimited noise components. If, instead, those bandlimited noise components were generated using bandpass filters with varying center frequencies, far more computation would be required just to update the time-varying filter coefficients. In such methods, a single pre-calculated and optimized filter may be used for all synthesized spectral envelope peaks, and the output of that single filter is multiplied by sinewaves at the component center frequencies (and having the desired component amplitudes).

[0084] In various examples that are not part of the invention, variations include algorithms where a noise-like excitation signal that is independent of the input signal spectrum is used to excite the spectral envelope filter used the earlier-described frequency translation algorithm approaches. One disadvantage to this approach, relative to the present approach, is that the excitation signal needs to have sufficient bandwidth to excite all parts of the spectral envelope filter that might have significant magnitude, so the synthesis of the noise sequence cannot be decimated to the same extent. The prototype noise in the proposed algorithm could be as narrow as, say 500 Hz, but the full bandwidth of the spectral envelope filter could be 2 to 3 kHz. Using a narrower prototype noise also controls the spread of spectral energy.

[0085] Composition of the translated spectral envelope can be achieved by modifying the earlier-described algorithms to use fixed bandwidth second order (biquad) sections and following each one with a gain scale. In this scheme, the biquad filter center frequencies and the gain scales would be updated, but the bandwidth (or Q factor) would remain constant.

[0086] Alternatively, the excitation signal could be constructed using a flat magnitude spectrum and the phase spectrum of the input signal, thereby preserving the fine structure of the input signal. This is a more aggressive whitening of the input signal than is achieved in the linear prediction decomposition currently used. This whitened signal could then excite a bank of fixed-bandwidth filters to produce the translated signal.

[0087] The present subject matter includes hearing assistance devices, including, but not limited to, cochlear implant type hearing devices, hearing aids, such as behind-the-ear (BTE), in-the-ear (ITE), in-the-canal (ITC), or completely-in-the-canal (CIC) type hearing aids. It is understood that behind-the-ear type hearing aids may include devices that reside substantially behind the ear or over the ear. Such devices may include hearing aids with receivers associated with the electronics portion of the behind-the-ear device, or hearing aids of the type having a receiver in-the-canal. Such devices may also be referred to as receiver-in-the-canal (RIC) or receiver-in-the-ear (RITE) devices. It is understood that other hearing assistance devices not expressly stated herein may fall within the scope of the claimed subject matter.

[0088] It is understood one of skill in the art, upon reading and understanding the present application will appreciate that variations of order, information or connections are possible without departing from the present teachings. This application is intended to cover adaptations or variations of the present subject matter. It is to be understood that the above description is intended to be illustrative, and not restrictive. The scope of the present subject matter should be determined with reference to the appended claims.

REFERENCES CITED IN THE DESCRIPTION

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Patentkrav

1. Fremgangsmåde til behandling af et lydsignal modtaget af et høreapparat, idet fremgangsmåden omfatter:

5 filtrering af lydsignalet for at generere et højfrekvent filtreret signal, hvor filtreringen sker ved en delefrekvens valgt for at isolere en højfrekvent del af lydsignalet til frekvenstransponering;

transponering af mindst en del af et lydspektrum af udelukkende det højfrekvente filtrerede signal til et lavere frekvensområde i en transponeringsproces for at frembringe et transponeret lydsignal; og

10 summering af det transponerede lydsignal med lydsignalet for at generere et udgangssignal,

hvor transponeringsprocessen omfatter vurdering af en alpolet indhyldningskurve af det højfrekvente filtrerede signal fra en flerhed af spektrallinjefrekvenser;

15 anvendelse af en warping-funktion på den vurderede alpoledede indhyldningskurve for at transponere polerne over en bestemt grænsefrekvens til lavere frekvenser, hvorved der frembringes en transponeret indhyldningskurve; idet fremgangsmåden **er kendetegnet ved**

20 at excitere den transponerede indhyldningskurve med et exciteringssignal til syntetisering af det transponerede lydsignal, idet exciteringssignalet syntetiseres ved anvendelse af en supplerende syntese af modulerede prototypespektralpeaks, hvor prototypespektralpeaket er et smalbåndet støjsignal, idet prototypespektralpeaket moduleres af en sinusbølge ved centerfrekvensen og magnituden af det transponerede indhyldningskurvepeak.

2. Fremgangsmåde ifølge krav 1, hvor det smalbandede støjsignal frembringes ved anvendelse af tilfældig sampling genereret ved reduceret hyppighed og upsampled ved anvendelse af interpolation eller udglatning.

3. Fremgangsmåde ifølge krav 1, hvor det smalbandede støjsignal frembringes ved anvendelse af et enkelt lavpasfilter med fast koefficient.

4. Fremgangsmåde ifølge et hvilket som helst af de foregående krav, hvor modulering af prototypespektralpeaket indeholder anvendelse af tabelopslag.

- 5.** Fremgangsmåde ifølge krav 1, hvor prototypespektralpeaket er moduleret via en sum af to eller flere sinusbølger til at frembringe to eller flere transponerede peaks ved forskellige magnituder uden at ændre hverken det ene eller det andet peaks form.
- 5 **6.** Fremgangsmåde ifølge et hvilket som helst af de foregående krav, omfattende anvendelse af et 2. ordensfilter med fast båndbredde til at opnå den transponerede indhyldningskurve.
- 7.** Fremgangsmåde ifølge et hvilket som helst af de foregående krav, omfattende anvendelse af et transponeret indhyldningskurvefilter til at forhøje den lavfrekvente energi dæmpet ved
10 filtreringen af lydsignalet.
- 8.** Fremgangsmåde ifølge et hvilket som helst af de foregående krav, hvor centerfrekvensen og magnituden af de transponerede indhyldningskurvepeaks opdateres for hver blok af lydsignalet.
15
- 9.** Fremgangsmåde ifølge et hvilket som helst af de foregående krav, hvor prototypespektralpeaket har en bredde på 500 Hz og fremstilles ved anvendelse af et indhyldningskurvefilter med en båndbredde på 2 til 3 kHz.
- 20 **10.** System til behandling af et lydsignal modtaget af et høreapparat, idet systemet omfatter: en digital signalprocessor (322) programmeret til at udføre trinnene ifølge et hvilket som helst af kravene 1 til 9.
- 11.** System ifølge krav 10, hvor høreapparatet indeholder et hørehjælpemiddel.
25
- 12.** System ifølge krav 10, hvor høreapparatet indeholder et cochlear-implantat.

DRAWINGS

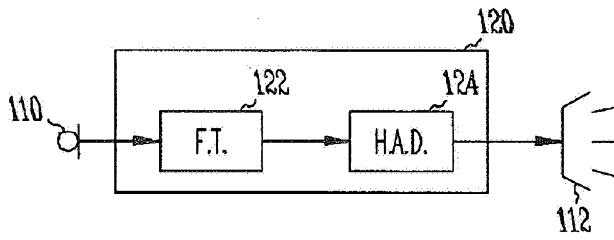


Fig. 1

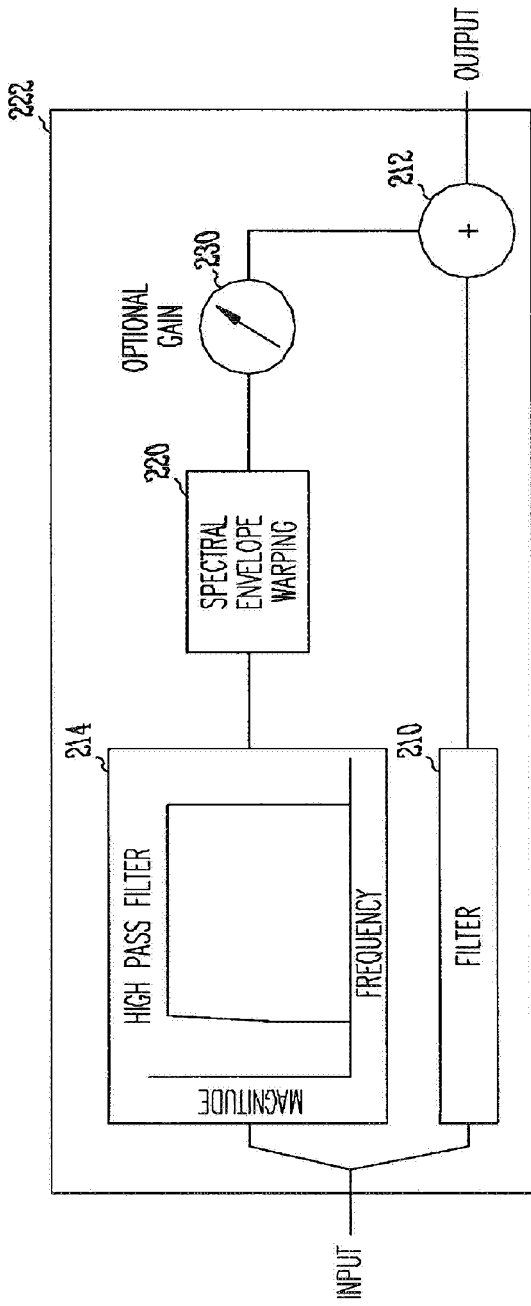


Fig. 2

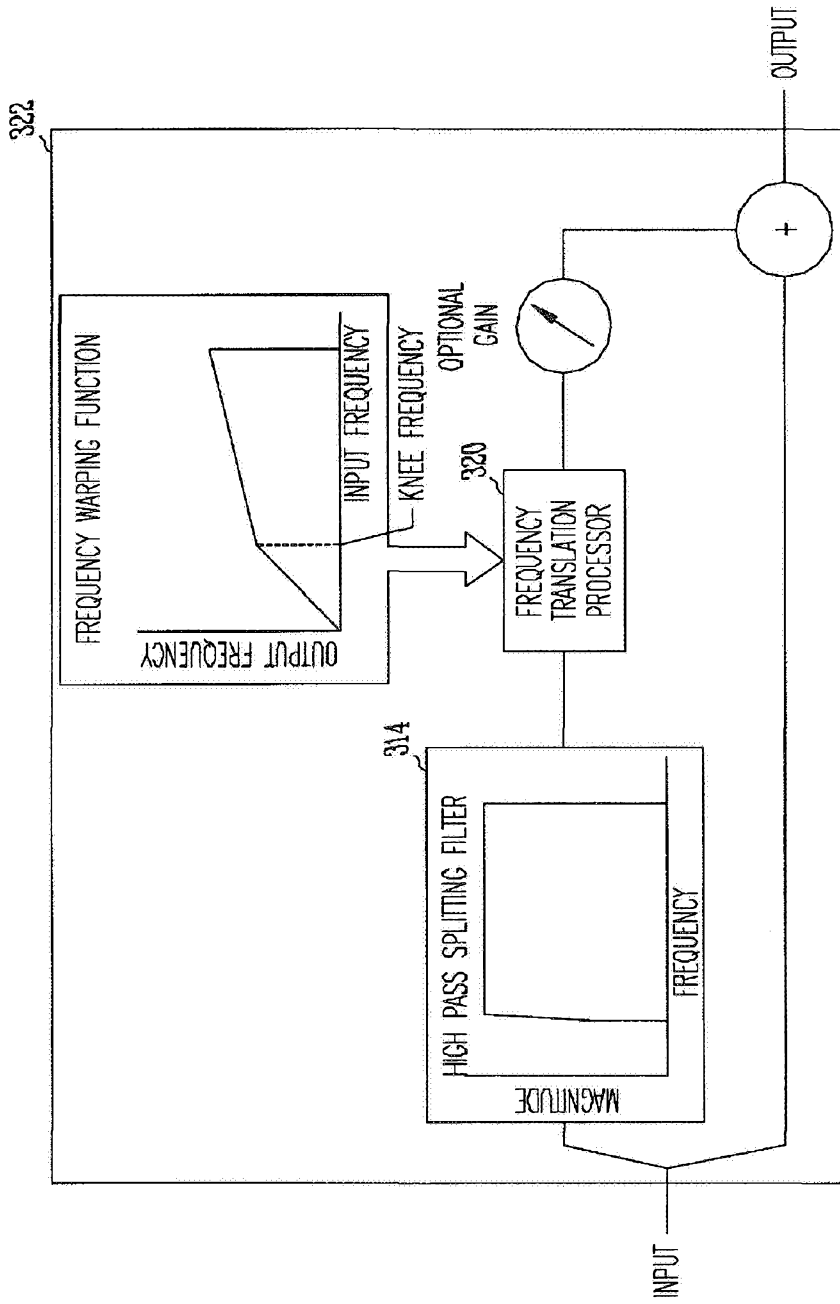


Fig. 3

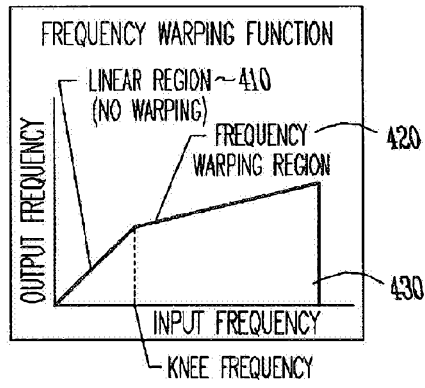


Fig. 4

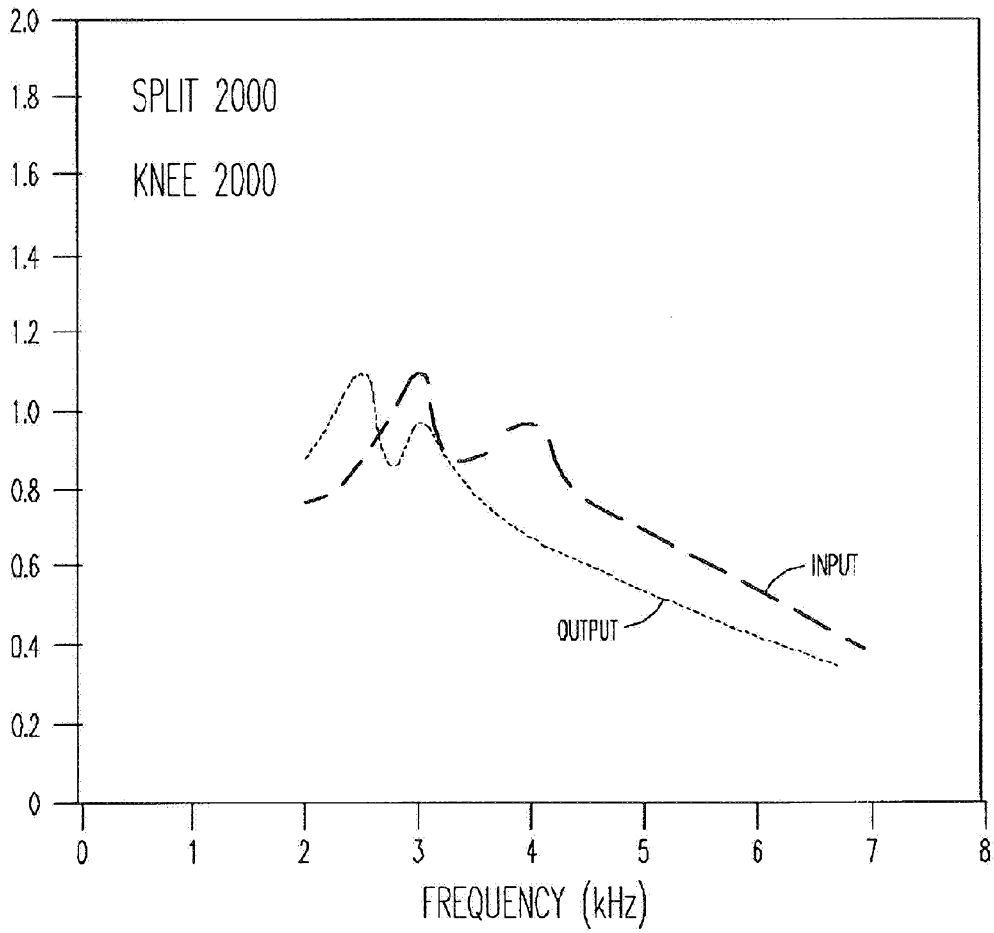


Fig. 5

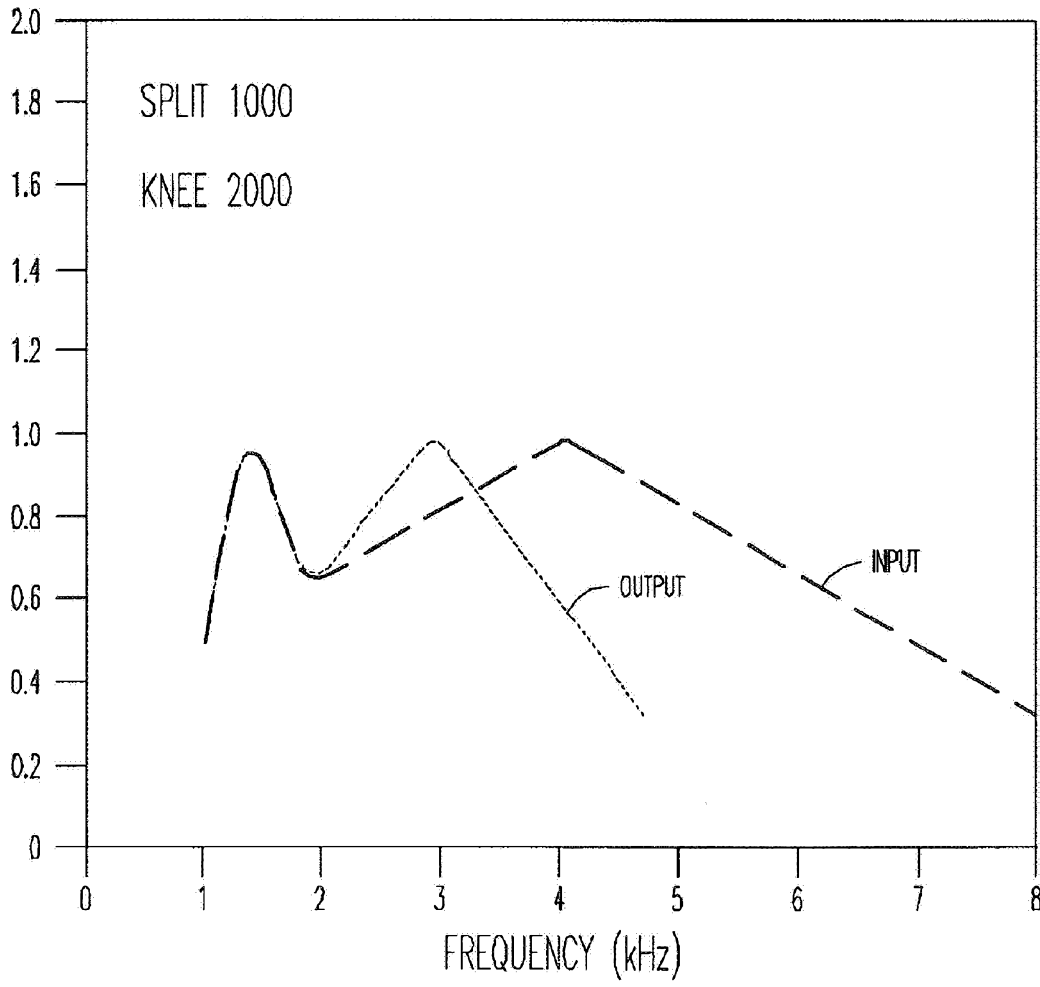


Fig. 6

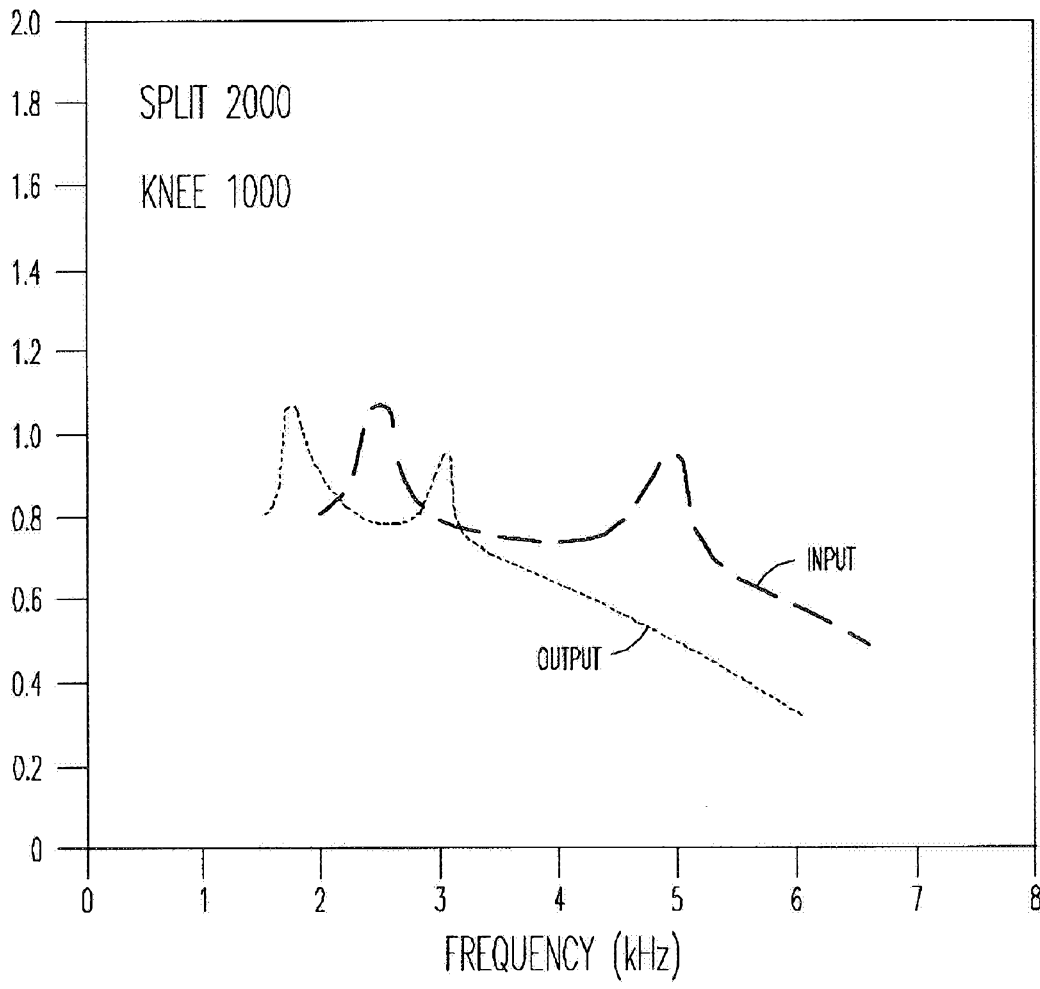


Fig. 7

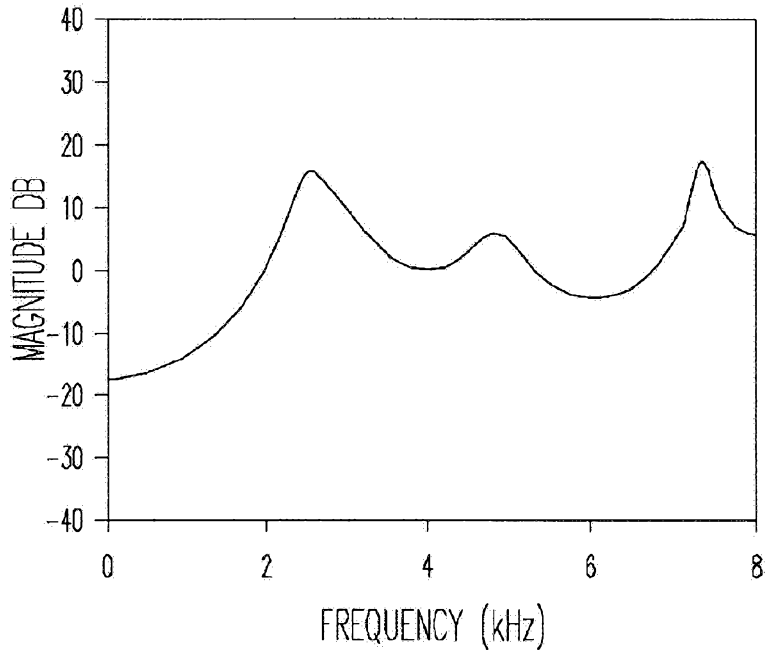


Fig. 8A

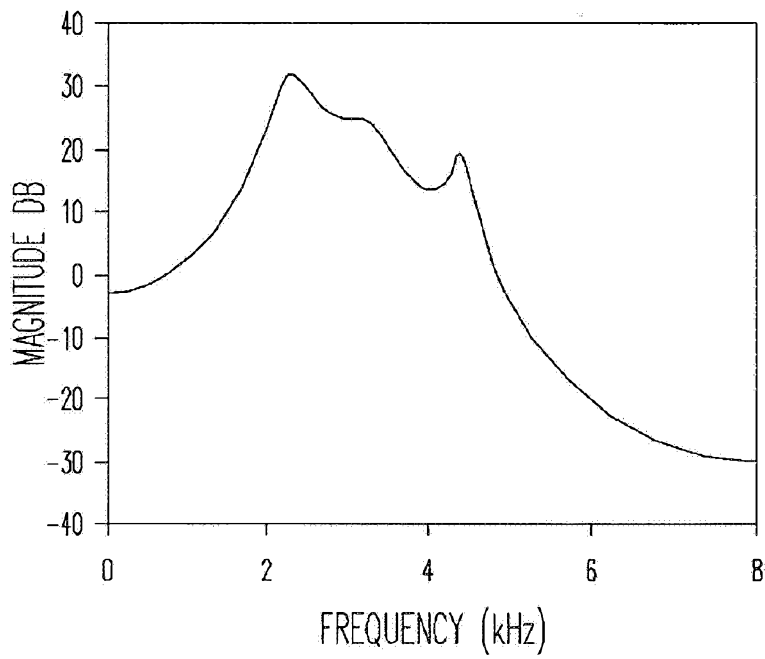
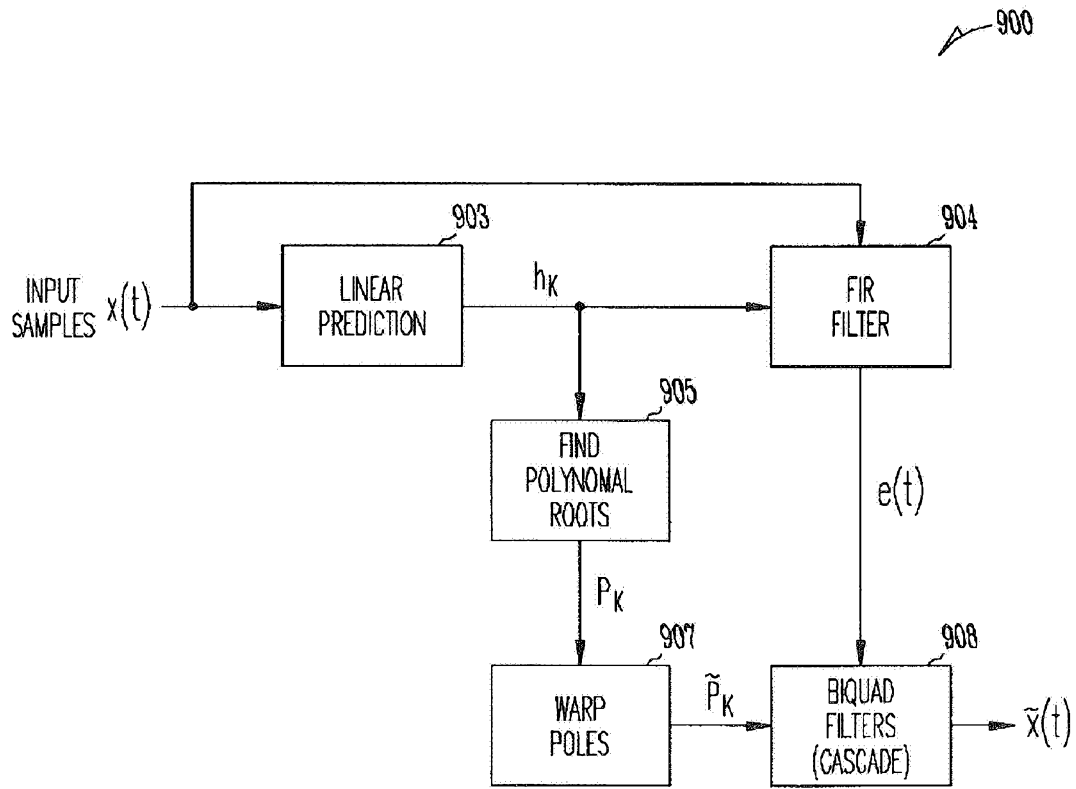


Fig. 8B

*Fig. 9*

1000

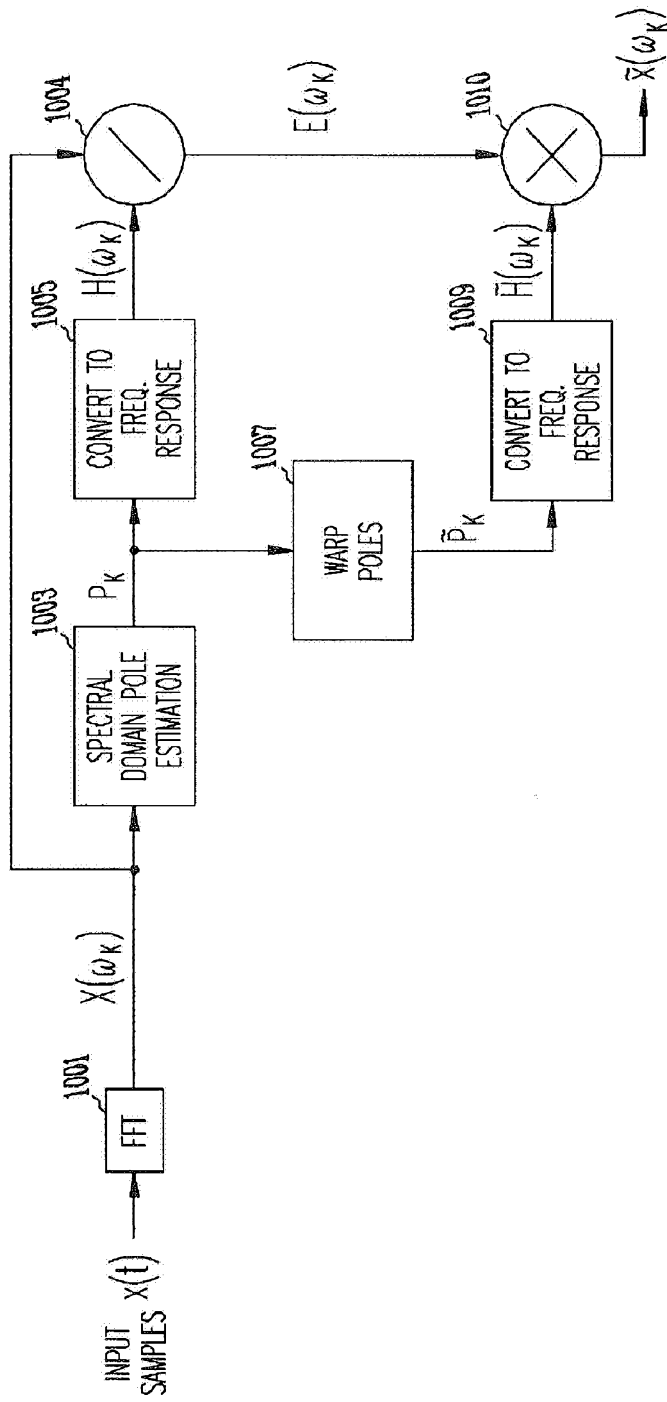
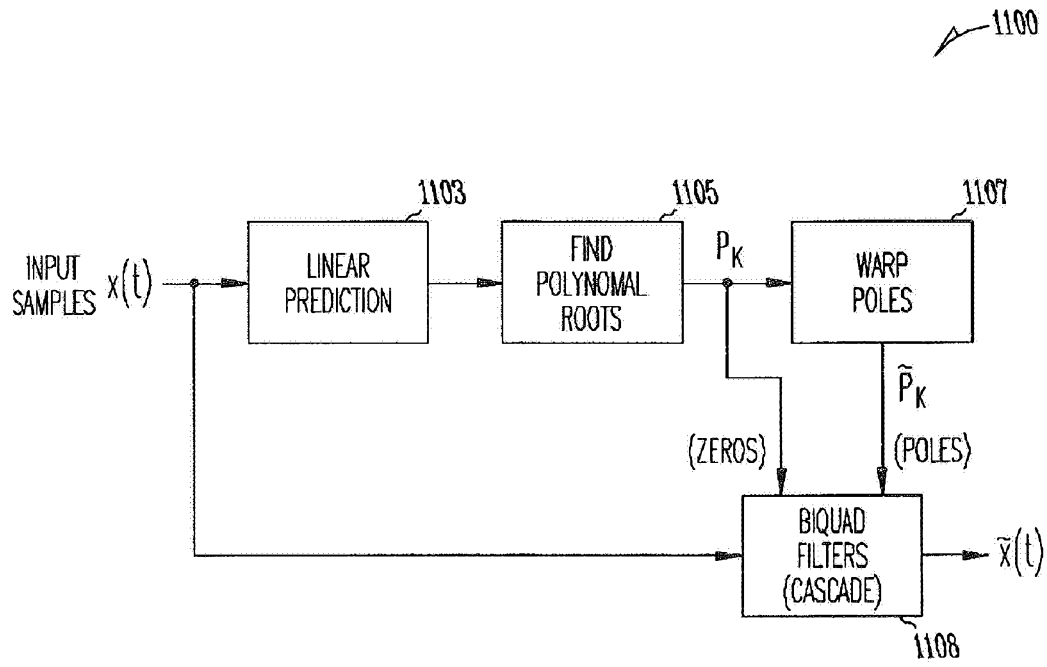


Fig. 10

*Fig. 11*

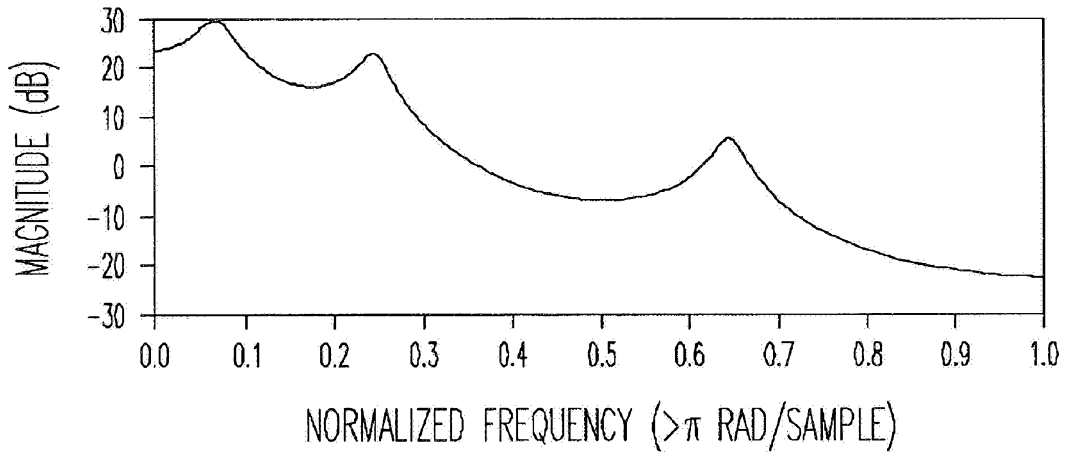


Fig. 12A

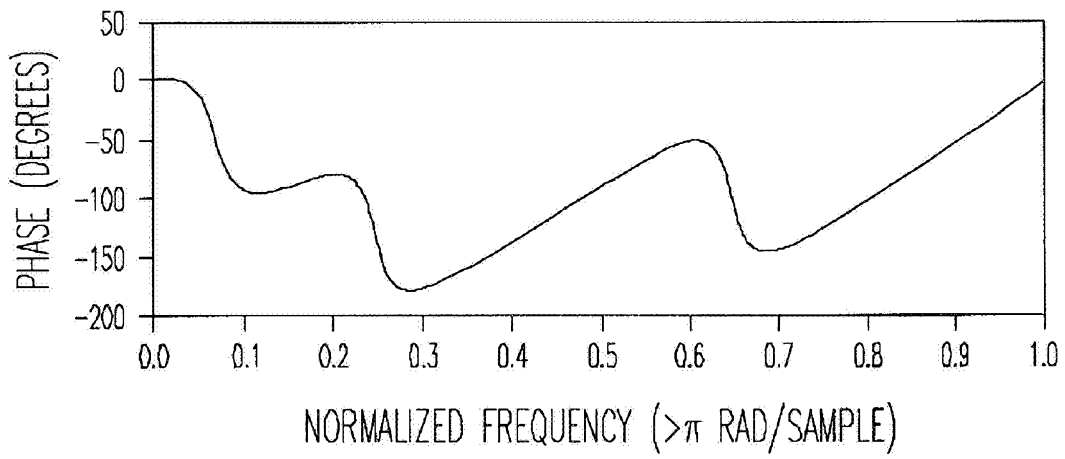


Fig. 12B

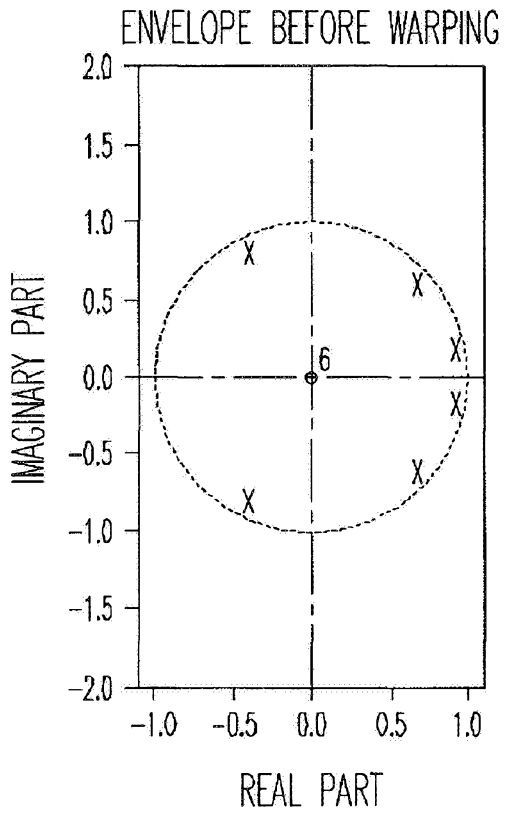


Fig. 13A

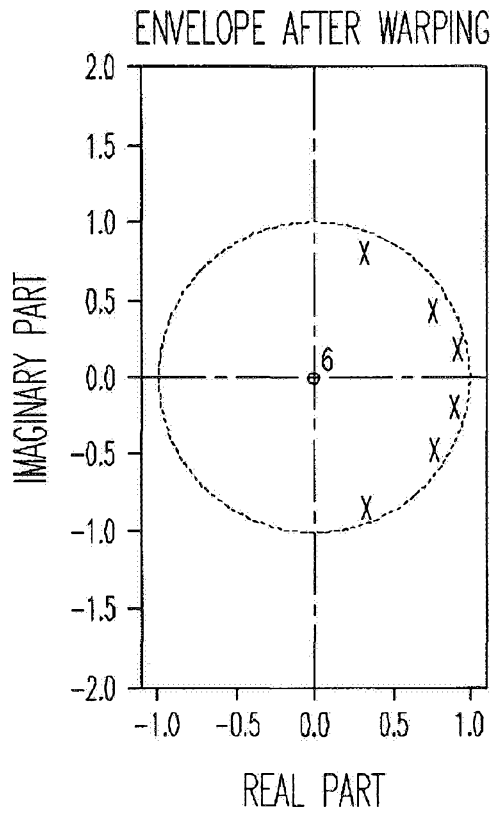


Fig. 13B

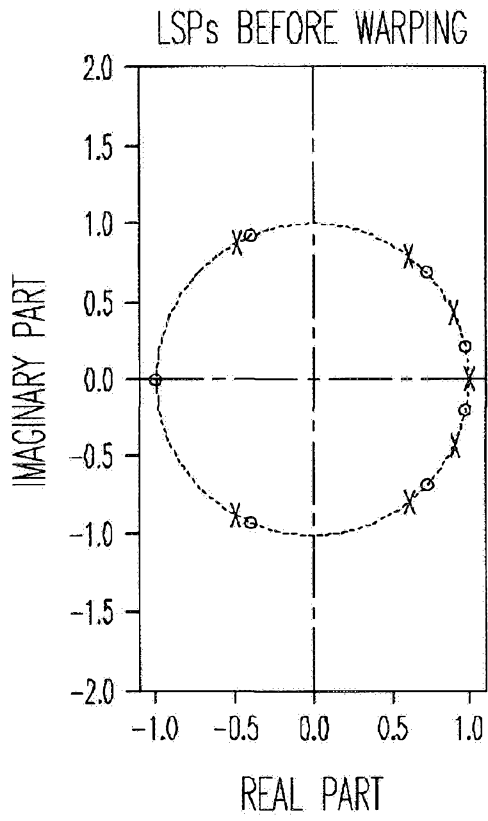


Fig. 14A

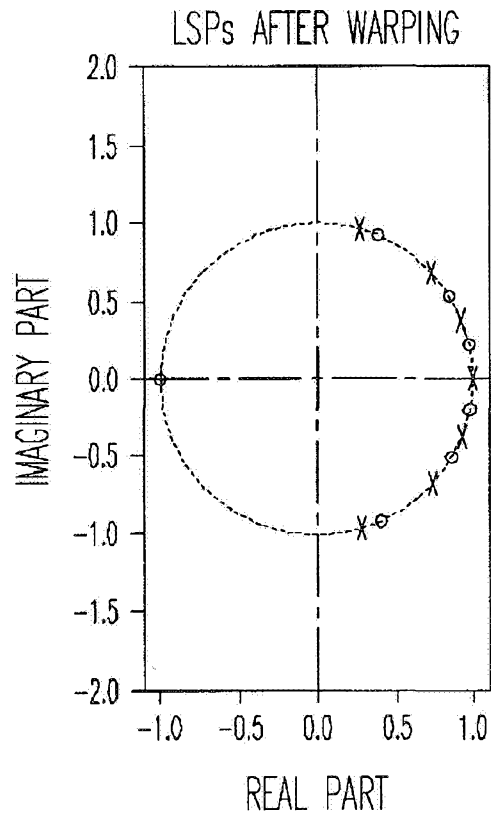


Fig. 14B

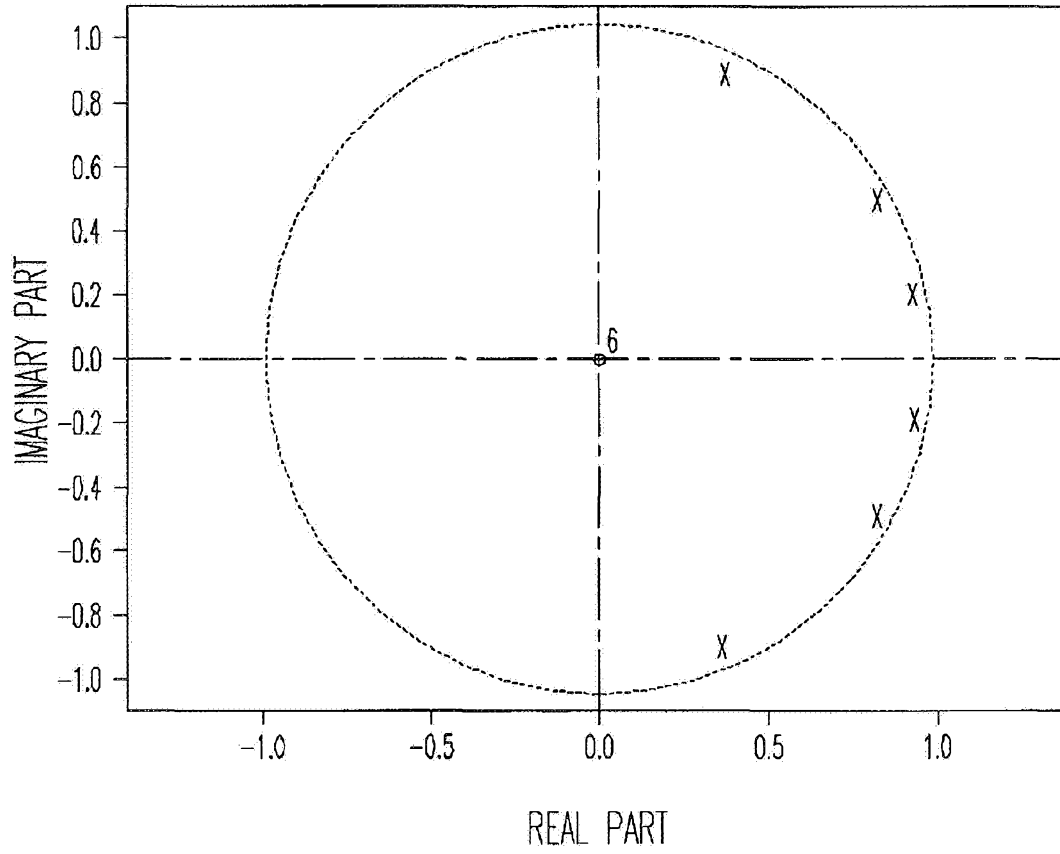


Fig. 15

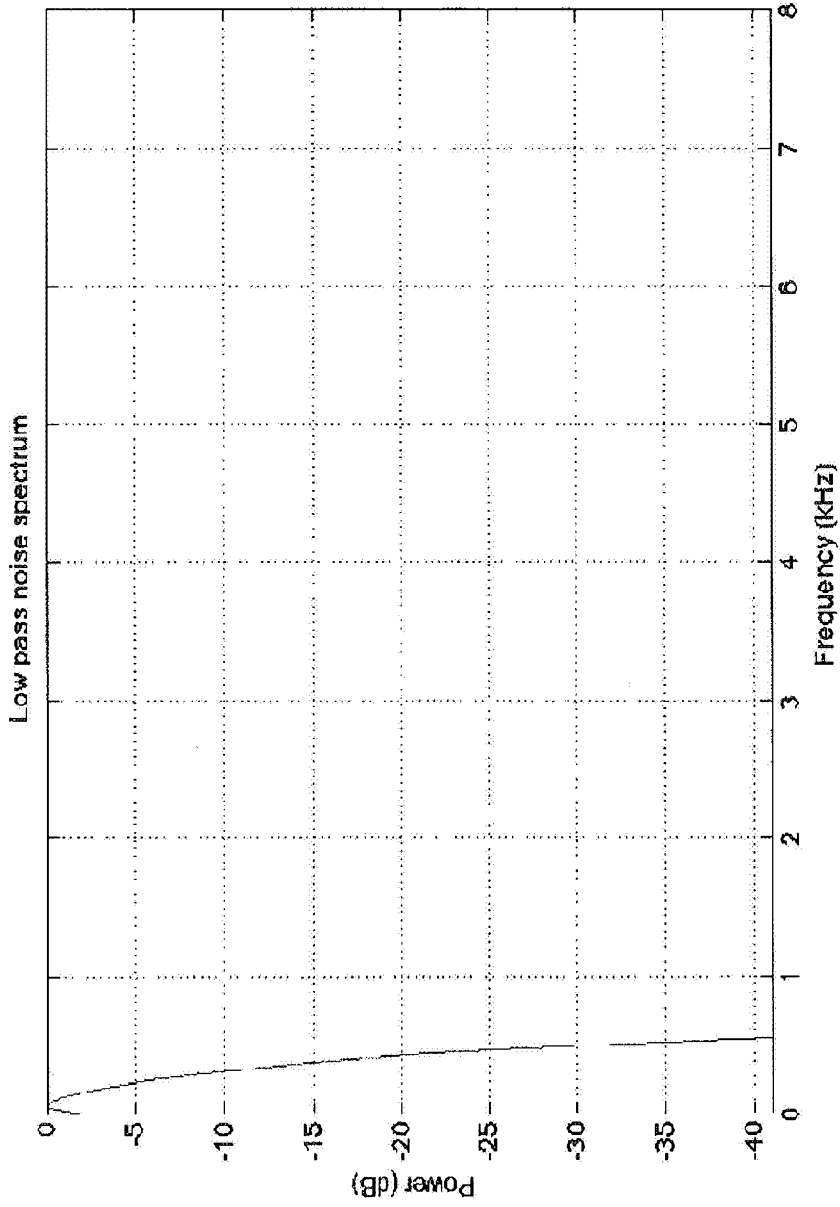


Fig. 16

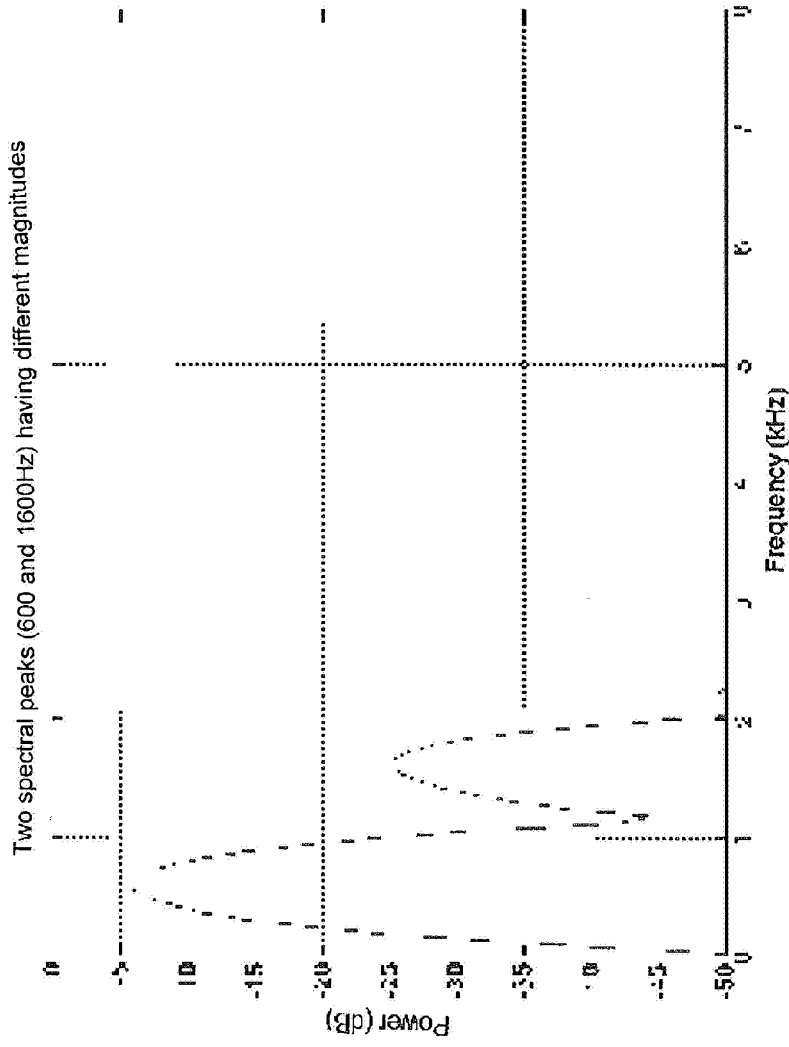


Fig. 17

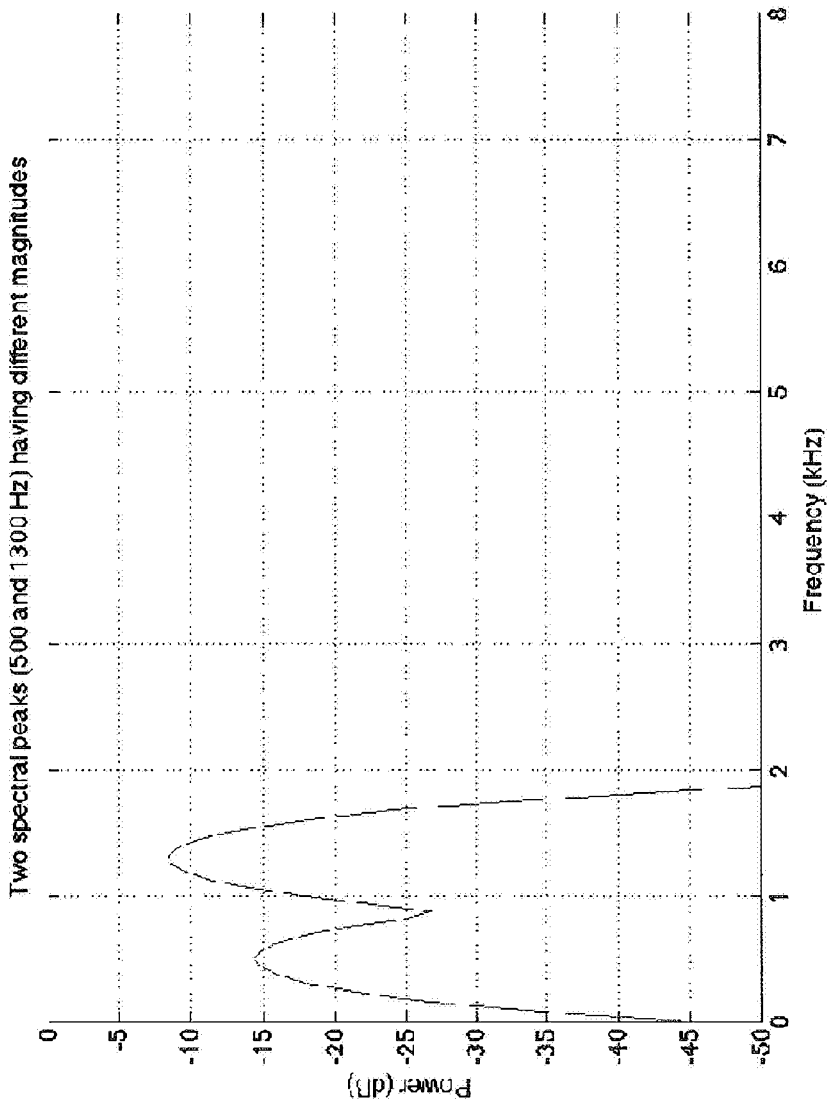


Fig. 18

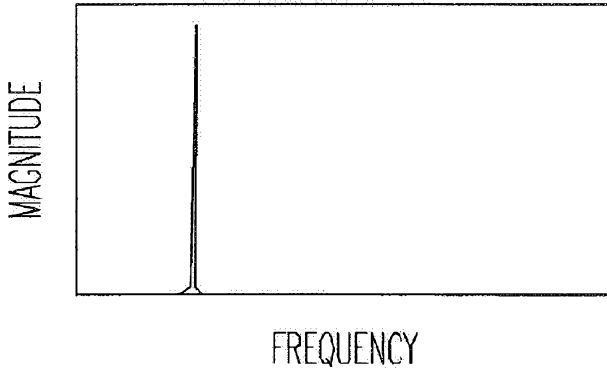


Fig. 19A

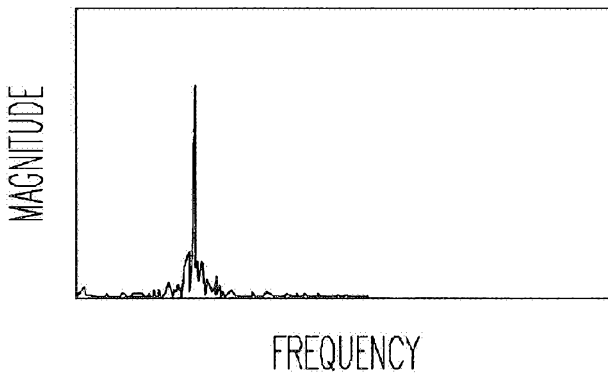


Fig. 19B

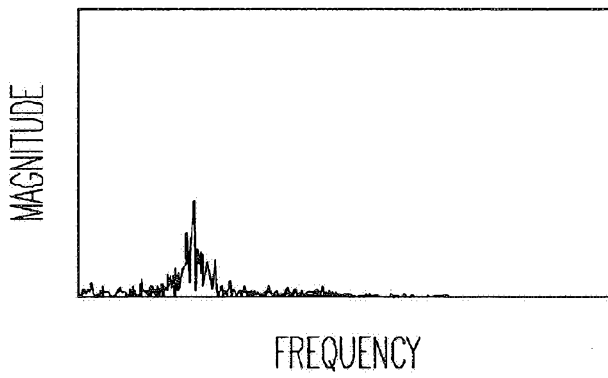


Fig. 19C