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(54) **Method and system for providing an acoustic signal with extended bandwidth**

Verfahren und System zur Bereitstellung eines Tonsignals mit erweiterter Bandbreite

Procédé et système fournissant un signal acoustique avec une largeur de bande étendue

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- **ISER B ET AL: "BANDWIDTH EXTENSION OF TELEPHONY SPEECH" EURASIP NEWS LETTER, XX, XX, June 2005 (2005-06), pages 1-148, XP002372006 ISSN: 1687-1421**

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Description

[0001] The invention is directed to a method and a system for providing an acoustic signal, in particular a speech signal, with extended bandwidth.

[0002] Acoustic signals transmitted via an analog or digital signal path usually suffer from the drawback that the signal path only has a restricted bandwidth such that the transmitted acoustic signal differs considerably from the original signal. For example, in the case of conventional telephone connections, a sampling rate of 8 kHz is used resulting in a maximal signal bandwidth of 4 kHz. Compared to the case of audio CD's, the speech and audio quality is significantly reduced.

[0003] Furthermore, many kinds of transmissions show additional bandwidth restrictions. In the case of an analog telephone connection, only frequencies between 300 Hz and 3.4 kHz are transmitted. As a result, only 3.1 kHz bandwidth are available.

[0004] In principle, the bandwidth of telephone connections could be increased by using broadband or wideband digital coding and decoding methods (so-called broadband codecs). In such a case, however, both the transmitter and the receiver have to support corresponding coding and decoding methods which would require the implementation of a new standard.

[0005] As an alternative, systems for bandwidth extension can be used as described, for example, in P. Jax, Enhancement of Bandlimited Speech Signals: Algorithms and Theoretical Bounds, Dissertation, Aachen, Germany, 2002 or E. Larsen, R. M. Aarts, Audio Bandwidth Extension, Wiley, Hoboken, NJ, USA, 2004. These systems are to be implemented on the receiver's side only such that existing telephone connections do not have to be changed. In these systems, the missing frequency components of an input signal with small bandwidth are estimated and added to the input signal.

[0006] An example of the structure and the corresponding signal flow in such a state of the art bandwidth extension system is illustrated in Fig. 6. In general, both the lower and the upper frequency ranges are re-synthesized.

[0007] At block 601, an incoming or received acoustic signal $x(n)$ in digitized form is processed by sub-sampling and block extraction so as to obtain signal vectors $\mathbf{x}(n)$. Here, the variable n denotes the time. In this Figure, it is assumed that the incoming signal $x(n)$ has already been converted to the desired bandwidth by increasing the sampling rate. In this conversion step, no additional frequency components are to be generated which can be achieved, for example, by using appropriate anti-aliasing or anti-imaging filter elements. In order to not amend the transmitted signal, the bandwidth extension is performed only within the missing frequency ranges. Depending on the transmission method, the extension concerns low frequency (for example from 0 to 300 Hz) and/or high frequency (for example 3400 Hz to half of the desired sampling rate) ranges.

[0008] In block 602, a narrowband spectral envelope is extracted from the narrowband signal, the narrowband signal being restricted by the bandwidth restrictions of the telephone channel. Via a non-linear mapping, a corresponding broadband envelope signal is estimated from the narrowband envelope. The mappings are based, for example, on codebook pairs (see J. Epps, W. H. Holmes, A New Technique for Wideband Enhancement of Coded Narrowband Speech, IEEE Workshop on Speech Coding, Conference proceedings, pages 174 to 176 June 1999) or on Neural Networks (see J.-M. Valin R. Lefebvre, Bandwidth Extension of Narrowband Speech for Low Bit-Rate Wideband Coding, IEEE Workshop on Speech Coding, Conference Proceedings, pages 130 to 132, September 2000). In these methods, the entries of the codebooks or the weights of the neural networks are generated using training methods requiring large processor and memory resources.

[0009] Furthermore, in block 603, a broadband or wideband excitation signal having a spectrally flat envelope is generated from the narrowband signal. This excitation signal corresponds to the signal which would be recorded directly behind the vocal cords, i.e. the excitation signal contains information about voicing and pitch, but not about form and structures or the spectral shaping in general. Thus, to retrieve a complete signal, such as a speech signal, the excitation signal has to be weighted with the spectral envelope. For the generation of excitation signals, non-linear characteristics (see U. Kornagel, Spectral Widening of the Excitation Signal for Telephone-Band Speech Enhancement, IWAENC 01, Conference Proceedings, pages 215 to 218, September 2001) such as two-ray rectifying or squaring, for example, may be used.

[0010] For bandwidth extension, the excitation signal $\mathbf{x}_{exc}(n)$ is spectrally colored using the envelope in block 604. After that, the spectral ranges used for the extension are extracted using a band stop filter in block 606 resulting in signal vectors $\mathbf{y}_{ext}(n)$. The band stop filter can be effective, for example, in the range from 200 to 3700 Hz.

[0011] The signal vectors x(n) of the received signal are passed through a complementary band pass filter in block 605. Then, the signal components $\boldsymbol{y}_{\mathrm{ext}}(\boldsymbol{n})$ and $\mathbf{y}_{tel}(\boldsymbol{n})$ are added to obtain a signal vector $\mathbf{y}(n)$ with extended bandwidth. In block 607, the different signal vectors are assembled again and an over-sampling is performed resulting in a signal $y(n)$.

[0012] A system and a method for bandwidth extension of bandlimited audio signals is known from EP 1 638 083. B. Iser et al "Bandwidth extension of telephoning speech", EURASIP NEWSLETTER, June 2005. pages 2 - 24 gives an introduction to this topic. EP 0 944 036 discloses a method and a device for detecting voice sections. A method for speech bandwidth extension is disclosed in US 2002/0138268. A frequency interpolating device and method is known from EP 1 298 643.

[0013] This problem is solved by the method according to claim 1 and the apparatus according to claim 13.

[0014] In accordance with the invention, a method for providing an acoustic signal with extended bandwidth is provided, comprising:

(a) automatically determining a current upper and a current lower bandwidth limit of received acoustic signal,

(b) automatically determining at least one complementary signal to complement the received acoustic signal between a predefined lower broadband bandwidth limit and the current lower bandwidth limit and/or between the current upper bandwidth limit and a predefined upper broadband bandwidth limit, wherein the predefined lower broadband bandwidth limit is smaller than the current bandwidth limit and the predefined upper broadband bandwidth limit is larger than the current upper bandwidth limit,

(c) automatically assembling the at least one complementary signal and the received acoustic signal to obtain an acoustic signal with extended bandwidth.

[0015] By determining current upper and lower bandwidth limits of a received acoustic signal and determining a complementary signal between the current bandwidth limits and the respective predefined broadband (or wideband) bandwidth limits, the method according to the invention allows an adaptation of the bandwidth extension to the acoustic signal actually received. For example, when the transmitter uses an ISDN telephone, a broader frequency range is used compared to the case of a mobile phone with a hands-free system. Therefore, the bandwidth of a received acoustic signal will be extended only in those ranges where it is necessary so that the quality of the resulting signal is very high.

[0016] In this way, on the one hand, no spectral gaps will occur even if the received signal covers only a very narrow frequency range. On the other hand, when receiving signals covering a relatively broad frequency range, no frequencies are cut-off when determining the complementary signal.

[0017] The received acoustic signal may be a digital signal or may be digitized. In the above method, steps (a) to (c) may be preceded by the step of converting the received acoustic signal to a predetermined sampling rate. Furthermore, steps (a) to (c) may be preceded by the step of extracting a signal vector from the acoustic signal, in particular, the converted acoustic signal. The signal vector may be obtained by sub-sampling the acoustic signal and may comprise a predefined number of entries. Then, subsequent (in time) signal vectors may overlap. The use of signal vectors simplifies further processing of the signals.

[0018] Steps (a) to (c) may be preceded by the step of determining a spectral vector of the received acoustic signal. In particular, a window function may be applied to signal vectors of the received acoustic signal. For example, a Hann or a Hamming window may be used (see K. D. Kammeyer, K. Kroschel, Digitale Signalverarbeitung, 4 th Edition, Teubner, Stuttgart, Germany 1997). Signal vectors, in particular the signal vectors weighted in this way, may be transformed into the Fourier domain using a discrete Fourier transform. The resulting vector is a short-term spectral vector. This allows for further processing in the Fourier domain.

[0019] In the above methods, step (b) comprises determining a broadband spectral envelope signal and a broadband excitation signal between the lower and upper broadband bandwidth limits such that the product of spectral envelope signal and excitation signal corresponds to the received acoustic signal according to a predetermined criterion.

[0020] Such a decomposition into an envelope signal and an excitation signal simplifies determining the current bandwidth limits and increases the accuracy when determining a complementary signal.

[0021] Step (a) comprises comparing a determined broadband spectral envelope signal and a spectrum of the received acoustic signal. It turned out that the spectrum is a suitable basis for determining current bandwidth limits of the acoustic signal.

[0022] Thus, if current bandwidth limits have been determined in step (a) in this way using a broadband spectral envelope signal of the received acoustic signal, determining a complementary signal in step (b) based on these current bandwidth limits and comprising determination of an envelope signal enables to iteratively adapt the current bandwidth limits by comparing again the (newly) determined envelope signal and a spectrum. In other words, determining current bandwidth limits in step (a) may use a spectral envelope signal determined according to step (b), particularly in a preceding step or in a preceding iteration of the method.

[0023] The comparing step may comprise selecting the minimal and maximal frequency for which the long-term power spectrum is larger than or equal to the determined broadband spectral envelope signal plus a predetermined constant.

[0024] This is a particularly simple and reliable way to determine the bandwidth limits. The predetermined constant can be chosen based on empirical or theoretical data. The predetermined constant may be negative.

[0025] In the above methods, determining a broadband spectral envelope signal may comprise selecting an envelope signal from a codebook according to a predetermined criterion.

[0026] By using codebooks, the required computing power can be reduced for determining an envelope signal. In principle, different kinds of criteria can be used when selecting an envelope signal from a codebook. In particular, using a predetermined distance criterion such as a cepstral distance can be used, particularly if the codebook entries have the form of cepstral vectors.

[0027] In particular, selecting an envelope signal may comprise equalizing the received acoustic signal and selecting an envelope signal from the codebook having minimal distance to the equalized acoustic signal according to a predetermined distance criterion, in particular, having a minimal cepstral distance.

[0028] Equalizing the acoustic signal allows to modify it such that a comparison with envelope signals from the codebook can be simplified. In particular, the received acoustic signal can be equalized in such a way that the resulting signal shows a long-term power spectrum corresponding to the long-term power spectrum of the signal used for training the codebook. Equalizing can be restricted to frequencies between the current upper and lower bandwidth limits of the received acoustic signal; outside these limits, the signal may remain unchanged. In particular, equalizing the received acoustic signal can be performed using a normalized long-term power spectrum of the signal used for training the codebooks, particularly using the normalized long-term power spectrum divided by the normalized long-term power spectrum of the received acoustic signal itself.

[0029] The codebook may comprise pairs of corresponding envelope signals, each pair comprising a broadband envelope signal between the lower and upper broadband bandwidth limits and a corresponding narrowband envelope signal between a lower narrowband bandwidth limit being larger than the lower broadband bandwidth limit and an upper narrowband bandwidth limit being smaller than the upper broadband bandwidth limit, and selecting an envelope signal may comprise determining a narrowband envelope signal having minimal distance to the equalized acoustic signal according to the predetermined distance criterion and selecting the corresponding broadband envelope signal of this pair.

[0030] In this way, a simple comparison between the received acoustic signal and the elements of the codebook can be performed as narrowband signals usually match a received acoustic signal with a narrow bandwidth more closely.

[0031] When using a cepstral distance to select an envelope signal, the received acoustic signal, particularly in its equalized form, has to be transformed into the cepstral domain. Thus, the step of selecting an envelope signal can further comprise the steps of determining the absolute value squared of the sub-band signals of the received acoustic signal, determining an auto-correlation in the time domain, particularly by performing an inverse discrete Fourier transform on the vector of the absolute value squared, determining prediction coefficients, particularly using the Levinson-Durbin algorithm, performing a recursion to obtain the cepstral coefficients.

[0032] In order to determine a spectral envelope from the cepstral vectors, the method may further comprise the steps of recursively transforming a cepstral vector into prediction error coefficients, augmenting the prediction error filter vector by adding a predetermined number of zeros and subsequently performing a discrete Fourier transform to obtain an inverse spectrum, determining the reciprocal of each sub-band component to obtain a spectral envelope vector.

[0033] In the above methods, the step of selecting an envelope signal may be preceded by providing adapted narrowband codebook envelope signals being adapted to the current lower and upper bandwidth limits.

[0034] Such an adaptation of the codebook entries allows for an improved selection of a corresponding envelope signal from the codebook. In particular, if the received acoustic signal shows a broader bandwidth than the original narrowband envelope signals of the codebook, the adaptation would result in envelope signals in the codebook having an extended bandwidth. In this way, particularly fricatives can be more reliably detected.

[0035] The providing step may comprise processing broadband codebook envelope signals using a long-term power spectrum of the received acoustic signal.

[0036] Due to the use of the power spectrum of the received acoustic signal, a suitable adaptation to the acoustic signal can be obtained. The long-term power spectrum may be normalized; furthermore, the long-term power spectrum of the received acoustic signal may be divided by a normalized long-term power spectrum of a broadband signal used for training of the codebook. The processing of the broadband codebook envelope signals may be performed only for frequencies outside the current bandwidth limits; within the bandwidth limits, the envelope signals may remain unchanged. Processing using the long-term power spectrum may comprise weighting broadband codebook envelope signal vectors using the long-term power spectrum of the received acoustic signal.

[0037] In particular, if the received acoustic signal has been transformed into the Fourier domain, determining a long-term power spectrum may comprise performing a first order recursive smoothing of the absolute values squared of the sub-band signals corresponding to the acoustic signal. This can be done, in particular, only if a wanted signal, such as a speech signal, has been detected in the received acoustic signal.

[0038] In addition, the long-term power spectrum may be normalized, particularly with respect to a long-term power spectrum within predetermined frequency limits.

[0039] Alternatively, the long-term power spectrum may be determined in the time domain. This can be done by determining the auto-correlation and performing an LPC analysis to obtain corresponding prediction coefficients.

[0040] In the above methods, determining a broadband excitation signal may be based on prediction error filtering and/or a non-linear characteristic. In this way, suitable excitation signals can be generated. Possible non-linear characteristics are disclosed, for example, in U. Kornagel, *Spectral Widening of the Excitation Signal for Telephone-Band*

Speech Enhancement.

[0041] In the above methods, the at least one complementary signal may be based on a product of the determined broadband spectral envelope and the determined broadband excitation signal, and step (c) may comprise summing the received acoustic signal between the current lower and upper bandwidth limits and the at least one complementary signal being restricted to the band between the lower broadband bandwidth limit and a current lower bandwidth limit and/or to the band between the current upper bandwidth limit and the upper broadband bandwidth limit.

[0042] Thus, the complementary signal is based on spectrally coloring the excitation signal using the envelope signal. By adding a complementary signal only outside the current bandwidth limits of the received acoustic signal, artifacts are avoided in the resulting signal with extended bandwidth.

[0043] Step (c) may also comprise adapting the power of the complementary signal and/or the received acoustic signal. With this step, the power of the received acoustic signal can be maintained.

[0044] In the above-described methods, at least one of the steps may be performed in the cepstral domain. Particularly if the entries of the codebook are cepstral vectors, this allows for performing the method in a simpler way.

[0045] Steps (a) to (c) of the above methods may be repeated at predetermined time intervals. Then, the repeated adaptation to the currently received acoustic signal leads to a permanent high quality of the resulting broadband signal.

[0046] Steps (a) to (c) of the above methods may be repeated only if a wanted signal component, such as speech activity, is detected in the received acoustic signal. Particularly in the case of speech signals, an extension of the bandwidth of the received acoustic signal is advantageous. Thus, restricting the method to the case of detected speech activity reduces the required computing power and avoids the presence of artifacts due to mal-adaptation.

[0047] The invention also provides a computer program product comprising one or more computer-readable media having computer-executable instructions for performing the steps of the above-described methods when run on a computer.

[0048] Furthermore, an apparatus for providing an acoustic signal with extended bandwidth is provided, comprising:

bandwidth determining means for automatically determining a current upper and a current lower bandwidth limit of a received acoustic signal,

complementary signal means for automatically determining at least one complementary signal to complement the received acoustic signal between a predefined lower broadband bandwidth limit and the current lower bandwidth limit and/or between the current upper bandwidth limit and a predefined upper broadband bandwidth limit, wherein the predefined lower broadband bandwidth limit is smaller than

the current bandwidth limit and the predefined upper broadband bandwidth limit is larger than the current upper bandwidth limit, and

assembling means for automatically assembling the at least one complementary signal and the received acoustic signal to obtain an acoustic signal with extended bandwidth.

[0049] Analogous to the above-described method, such an apparatus provides an advantageous way to extend the bandwidth of a received acoustic signal. In particular, due to the determination of current upper and lower bandwidth limits of the received acoustic signal and a corresponding determination of a complementary signal, the quality of the resulting output signal is increased compared to the case of bandwidth extension systems with fixed parameters.

[0050] The complementary signal means comprises a means for determining a broadband spectral envelope signal and a broadband excitation signal between the lower and upper broadband bandwidth limits such that the product of spectral envelope signal and excitation signal corresponds to the received acoustic signal according to a predetermined criterion.

[0051] The bandwidth determining means is configured to compare a determined broadband spectral envelope signal and a spectrum of the received acoustic signal.

[0052] The bandwidth determining means may be configured to select the minimal and maximal frequency for which the spectrum is larger than or equal to the power spectrum of the determined broadband spectral envelope signal plus a predetermined constant.

[0053] In the above-described apparatus, the means for determining a broadband spectral envelope signal may comprise a means for selecting an envelope signal from a codebook according to a predetermined criterion.

[0054] The means for selecting an envelope signal may be configured to equalize the received acoustic signal and select an envelope signal from the codebook having minimal distance to the equalized acoustic signal according to a predetermined distance criterion, in particular, having a minimal cepstral distance.

[0055] In the above-described apparatus, the codebook may comprise pairs of corresponding envelope signals, each pair comprising a broadband envelope signal between the lower and upper broadband bandwidth limits and a corresponding narrowband envelope signal between a lower narrowband bandwidth limit being larger than the lower broadband bandwidth limit and an upper narrowband bandwidth limit being smaller than the upper broadband bandwidth limit, and the means for selecting an envelope signal may be configured to determine a narrowband envelope signal having minimal distance to the equalized acoustic signal according to the predetermined distance criterion and to select the corresponding broadband envelope signal of this pair.

[0056] The means for determining a broadband spectral envelope signal may comprise a means for providing adapted narrowband codebook envelope signals being adapted to the current lower and upper bandwidth limits.

[0057] The means for providing may be configured to process the broadband codebook envelope signal using a long-term power spectrum of the received acoustic signal.

[0058] In the above-described apparatus, the means for determining a broadband excitation signal may be configured to determine the broadband excitation signal based on prediction error filtering and/or a non-linear characteristic.

[0059] The at least one complementary signal may be based on a product of the determined broadband spectral envelope and the determined broadband excitation signal, and the assembling means may be configured to sum the received acoustic signal between the current lower and upper bandwidth limits and the at least one complementary signal being restricted to the band between the lower broadband bandwidth limit and a current lower bandwidth limit and/or to the band between the current upper bandwidth limit and the upper broadband bandwidth limit.

[0060] In the above-described apparatus, at least one of the means may be configured to perform at least part of its function in the cepstral domain.

[0061] The means of the above-described apparatus may be configured to perform their respective function repeatedly at predetermined time intervals.

[0062] The apparatus may further comprise a wanted signal detector, in particular, a speech detector, and the means may be configured to perform their respective function only if a wanted signal component is detected in the received acoustic signal.

[0063] Further features and advantages of the invention will be described in the following with reference to the figures.

Fig. 1 illustrates the structure of an example of an apparatus for providing an acoustic signal with extended bandwidth;

Fig. 2 is a flow diagram of an example of a method for providing an acoustic signal with extended bandwidth;

Fig. 3 illustrates an example of a normalized long-term power spectrum for training a codebook;

Fig 4 illustrates examples of codebook entries;

Fig. 5 illustrates the determination of current bandwidth limits;

Fig. 6 illustrates the structure of a prior art system.

[0064] Fig. 1 shows the structure of the signal flow in an apparatus for providing an acoustic signal with extended bandwidth. Fig. 2 is a flow diagram illustrating an example of a method for providing an acoustic signal with extended bandwidth which could be performed by the apparatus corresponding to Fig. 1. In view of this, Fig.'s 1 and 2 will be described in the following simultaneously.

[0065] According to step 201, an acoustic signal, such as a speech signal, is received via a telephone line. Because of the restricted bandwidth of the telephone line, an extension of the bandwidth is desired to improve the signal quality. Thus, the signal is to be augmented so as to obtain a predetermined broader bandwidth. It is to be understood that the method described in the following can be used for bandwidth extension independent of the type of incoming signal and independent of the type of transmission line, i.e., it need not be a telephone line.

[0066] The acoustic signal x(n) received by block 101 has already been pre-processed by increasing the sampling rate up to the predetermined broadband or wideband bandwidth. In this way, however, no additional frequency components are generated. This can be achieved, for example, by using suitable anti-aliasing or anti-imaging filters. This kind of bandwidth extension, preferably, is performed only for the "missing" frequency ranges; in the case of an analog telephone line, these ranges may be between 0 and 300 Hz and 3400 Hz up to half of the desired sampling rate, for example, up to 3700 Hz.

[0067] From the resulting signal $x(n)$, n denoting the time variable, signal vectors $\mathbf{x}(n)$ are generated (step 202). This can be achieved by taking every r sampling values up to a certain length. Thus, a signal vector with N_{ana} elements has the form:

$$\mathbf{x}(n) = [x(nr), x(nr-1), \ldots, x(nr - N_{ana} + 1)]^T .$$

[0068] It is to be noted that an overlap may exist between neighboring signal vectors. For a desired or final sampling rate of 11.025 kHz, one may take the following values:

$$r = 64,$$

$$N_{ana} = 256.$$

[0069] After that (step 203), a windowing procedure is performed on the signal vector so as to obtain a windowed signal vector $\mathbf{x}_w(n)$:

$$\mathbf{x}_w(n) = \mathbf{F}\mathbf{x}(n).$$

[0070] The window matrix F is a diagonal matrix of the form

$$\mathbf{F} = \begin{bmatrix} h_0 & 0 & 0 & \ldots & 0 \\ 0 & h_1 & 0 & \ldots & 0 \\ 0 & 0 & h_2 & \ldots & 0 \\ \vdots & \vdots & \vdots & \ddots & \vdots \\ 0 & 0 & 0 & \ldots & h_{N_{ana}-1} \end{bmatrix}.$$

[0071] The elements of this matrix can be chosen corresponding to different kinds of windows. Typical windows are the Hann or Hamming window. The weighted signal vectors are transformed into the Fourier domain using a discrete Fourier transform:

$$\mathbf{X}_w(n) = DFT\{\mathbf{x}_w(n)\}$$

[0072] The resulting short-term spectral vector has the form:

$$\mathbf{X}_w(n) = \left[X(e^{j\Omega_0}, n), X(e^{j\Omega_1}, n), \ldots, X(e^{j\Omega_\mu}, n), \ldots, X(e^{j\Omega_{N_{DFT}-1}}, n)\right]^T ,$$

wherein Ω_μ denotes the frequency variable.

[0073] Based on the spectral vectors, a long-term power spectrum of the received acoustic signal is determined in block 102 (step 204). There are different possibilities to estimate such a long-term power spectrum. According to one alternative, a first order recursive smoothing is performed on the absolute value squared of the sub-band signals $X(e^{j\Omega_\mu}, n)$:

$$\hat{S}_{xx}(\Omega_\mu,n)=\begin{cases}\beta_{fre}\hat{S}_{xx}(\Omega_\mu,n-1)+(1-\beta_{fre})\left|X_w(e^{j\Omega_\mu},n)\right|^2, & \text{during speech activity}\\ \hat{S}_{xx}(\Omega_\mu,n-1), & \text{else.}\end{cases}$$

[0074] Preferably, the time constant β_{fre} is chosen to be close to 1 ($0 << \beta_{fre} < 1$) so as to obtain a sufficiently large averaging time.

[0075] In principle, the recursive smoothing according to the first line of the above equation may be performed continuously. However, in order to avoid any artefacts, it may be performed only if a wanted signal component is present in the received acoustic signal, for example, if speech activity is detected. For this purpose, a speech detector may be provided as described, for example, in E. Hänsler, G. Schmidt, Acoustic Echo and Noise Control - A Practical Approach, Wiley, Hoboken, NJ, USA, 2004.

[0076] In order to simplify the further processing, the long-term power spectrum may be normalized to the long-term power within a predefined frequency band:

$$\hat{S}_{xx,norm}(\Omega_\mu,n)=\frac{\hat{S}_{xx}(\Omega_\mu,n)}{\sum_{\mu=\mu_l}^{\mu_u}\hat{S}_{xx}(\Omega_\mu,n)}.$$

[0077] The band limits $\Omega_{\mu l}$ and $\Omega_{\mu u}$ denote the lower and upper limits of a predefined frequency band. For example, this frequency band may correspond to a telephone band with minimal bandwidth for which the present method is to be used, for example, the limits may be 400 Hz and 3300 Hz. Preferably, the limits correspond to a band which is smaller or at most equal to the frequency band of the narrow frequency band within which the codebook described below has been trained; these limits being denoted by Ω_l and Ω_u.

[0078] Alternatively, to determine the long-term power spectrum in the frequency domain, an estimation can be performed in the time domain as well. For this purpose, an auto-correlation is estimated for about 10 to 20 sampling cycles of offset. Afterwards, prediction coefficients can be determined using an LPC (linear predictive coding) analysis. The long-term power spectrum is obtained via a discrete Fourier transform and a division.

[0079] In block 103 (step 205), the acoustic signal is equalized. The equalizing is performed on the spectral vector determined above:

$$\mathbf{X}_{eq}(n)=\mathbf{H}_{eq}(n)\mathbf{X}_w(n).$$

[0080] The equalizing matrix $\mathbf{H}_{eq}(n)$ is a diagonal matrix of the form

$$\mathbf{H}_{eq}(n)=\begin{bmatrix}H_{eq}(e^{j\Omega_0},n) & 0 & \cdots & 0\\ 0 & H_{eq}(e^{j\Omega_1},n) & \cdots & 0\\ \vdots & \vdots & \ddots & \vdots\\ 0 & 0 & \cdots & H_{eq}(e^{j\Omega_{N_{DFT}-1}},n)\end{bmatrix}$$

with the entries

$$\tilde{H}_{eq}(e^{j\Omega_\mu},n)=\begin{cases}1 & \text{if } \left(\left(\Omega_\mu<\overline{\Omega}_l(n-1)\right)\text{or}\left(\Omega_\mu>\overline{\Omega}_u(n-1)\right)\right)\\ \dfrac{\hat{S}_{\tilde{x}\tilde{x},norm}(\Omega_\mu,n)}{\hat{S}_{xx,norm}(\Omega_\mu,n)}, & \text{else}\end{cases}$$

and

$$H_{eq}(e^{j\Omega_\mu},n)=\begin{cases}H_{eq,\max}, & \text{if } \left(\tilde{H}_{eq}(e^{j\Omega_\mu},n)>H_{eq,\max}\right)\\ H_{eq,\min}, & \text{if } \left(\tilde{H}_{eq}(e^{j\Omega_\mu},n)<H_{eq,\min}\right)\\ H_{eq}(e^{j\Omega_\mu},n), & \text{else,}\end{cases}$$

[0081] In the equations above, $\overline{\Omega}_l\ (n-1)$ and $\overline{\Omega}_u\ (n-1)$ denote the current lower and upper bandwidth limits of the received acoustic signal. Thus, for obtaining an updated equalized signal, the bandwidth limits at time (n-1) are taken as the current bandwidth limits. Furthermore, $S_{\tilde{x}\tilde{x},norm}(\Omega_\mu, n)$ denotes the normalized long-term power spectrum of the broadband signal which has been used for training the codebook. Normalizing of such a power spectrum is performed analogously to the case of the long-term power spectrum of the received acoustic signal described above. An example for such a normalized long-term power spectrum used for training a codebook is shown in Figure 3.

[0082] The equalizing is restricted to minimal and maximum values, for example, to

$$H_{eq,\min}=-12\,\text{dB},$$

$$H_{eq,\max}=12\,\text{dB}.$$

[0083] As can be seen from the above, the acoustic signal is equalized only within the current bandwidth limits one time step before. Outside these bandwidth limits, no equalizing takes place.

[0084] In the following, determining a broadband spectrum envelope will be described in more detail. An envelope signal corresponding to the received acoustic signal will be determined using a codebook. The used codebook comprises a number of pairs of corresponding narrowband and broadband envelope signals. The codebook has been obtained by training with a large database on the basis of a starting long-term power spectrum (see Y. Linde, A. Buzo, R. M. Gray, An Algorithm for Vector Quantizer Design, IEEE Trans. Comm., vol. COM-28, no. 1, pages 84 - 95, Jan. 1980).

[0085] As indicated in Figure 2, the codebook entries are adapted in step 206 (block 104). In particular, the narrowband codebook entries $\mathbf{c}_{i,s}(n)$ are adapted.

[0086] This is achieved by starting with the broadband entries of the codebook. If the broadband envelope signals are provided as cepstral vectors $\mathbf{c}_{i,b}(n)$, the corresponding spectra $C_{i,b}\ (n)$ are determined. Based on these broadband spectral envelopes, the adapted or optimized narrowband spectra are determined by a multiplication with a weighting matrix:

$$\mathbf{C}_{i,s}(n)=\mathbf{H}_{\text{mod}}(n)\mathbf{C}_{i,b}(n).$$

[0087] The weighting matrix is a diagonal matrix of the form:

$$\mathbf{H}_{\text{mod}}(n) = \begin{bmatrix} H_{mod}(e^{j\Omega_0},n) & 0 & \cdots & 0 \\ 0 & H_{mod}(e^{j\Omega_1},n) & \cdots & 0 \\ \vdots & \vdots & \ddots & \vdots \\ 0 & 0 & \cdots & H_{mod}(e^{j\Omega_{N_{DFT}-1}},n) \end{bmatrix}$$

with the entries

$$H_{\text{mod}}(e^{j\Omega_\mu},n) = \begin{cases} 1, & \text{if } \left(\overline{\Omega}_l(n-1) < \Omega_\mu < \overline{\Omega}_u(n-1)\right), \\ \dfrac{\hat{S}_{xx,norm}(\Omega_\mu,n)}{\hat{S}_{\overline{x}\overline{x},norm}(\Omega_\mu,n)}, & \text{else.} \end{cases}$$

[0088] Afterwards, cepstral vectors are determined from the resulting spectral narrowband envelopes.

[0089] The conversion from spectral vectors to cepstral vectors and vice versa will be described in the following with respect to step 207 in which broadband spectral envelopes are determined (block 105).

[0090] A broadband spectral envelope from the codebook matching the acoustic signal best is determined by comparing the narrowband codebook entries with the spectral envelope of the spectrum of the acoustic signal (after equalizing). The narrowband codebook entry is selected that has the smallness distance to the acoustic signal spectrum. In principle, different distance criteria can be used. The cepstral distance is particularly useful as the codebook entries are provided in the form of cepstral vectors.

[0091] When an optimal narrowband codebook entry has been selected, the corresponding broadband codebook entry is determined as the optimal broadband spectral envelope for the received acoustic signal. Due to the adaptation of the narrowband codebook entries as described above, an optimal narrowband envelope can be selected in a very reliable way.

[0092] Converting a spectral vector, particularly of the received acoustic signal, to a cepstral vector can be achieved by:

1. Determining the absolute value squared of each sub-band signal $X_{eq}(e^{j\Omega_\mu}, n)$.

2. Applying an inverse discrete Fourier transform on this vector results in an estimation of the auto-correlation in the time domain.

3. Using the Levinson-Durbin algorithm, prediction coefficients (with an order of about 10 to 20) can be determined from the auto-correlation.

4. By performing a recursion with respect to the order, the prediction coefficients are used to determine the cepstral coefficients. Usually, the order corresponds to one and a half of the prediction order.

[0093] The optimal cepstral vector of the broadband codebook is designated by $\mathbf{c}_{opt,b}(n)$. The resulting broadband spectral envelope has the form:

$$\mathbf{C}_{opt,b}(n) = \left[C_{opt,b}(e^{j\Omega_0},n), C_{opt,b}(e^{j\Omega_1},n), \ldots, C_{opt,b}(e^{j\Omega_{N_{DFT}-1}},n)\right]^T.$$

[0094] Conversion of cepstral vectors into spectral vectors is achieved by:

1. Converting the cepstral vectors using a recursion with respect to the order (as above) to obtain prediction error filter coefficients.

2. By augmenting the prediction error filter vector by a predetermined number of zeros and subsequent performing

of a discrete Fourier transform, an inverse spectrum is obtained.

3. By determining the reciprocal of each sub-band component, the vector $\mathbf{C}_{opt,b}(n)$ is generated. Divisions by zero have to be treated separately, for example by adding a suitable constant.

[0095] Fig. 4 illustrates an example of a codebook with four pairs of entries. In each diagram, a corresponding original narrowband envelope, and a corresponding adapted narrowband envelope are shown. The original broadband and narrowband codebook entries have been obtained on the basis of a large database for an ISDN telephone connection. As can be seen in this figure, after the adaptation, the resulting optimized entries have a higher upper limit frequency. This allows for an improved detection of fricatives.

[0096] In step 208 (block 103), an excitation signal corresponding to the received acoustic signal is generated. This broadband excitation signal shows a spectrally flat envelope. It corresponds to a signal which would be recorded directly behind the vocal cords.

[0097] For determining a broadband excitation signal, first of all, the spectral envelope of the equalized short-term spectrum $\mathbf{X}_{eq}(n)$ is estimated in the form of prediction error filter coefficients. Applying an inverse discrete Fourier transform on this spectral vector allows to determine the corresponding time signal. After that, the vector in the time domain is filtered by a prediction error filter. The corresponding filter coefficients are those that have been determined previously.

[0098] Then, a non-linear characteristic, such as a two-way rectification or squaring, is applied to the filtered time domain vector. This generates the missing low frequency and high frequency signal components. A transformation in the Fourier domain provides, then, the spectrum of the extended excitation signal $\mathbf{X}_{exc}(n)$.

[0099] Alternatively, determining an excitation signal can be performed in the time sub-band or Fourier domain as well. Examples for this alternative can be found in B. Iser, G. Schmidt, Bandwidth Extension of Telephony Speech, Eurasip Newsletter, Volume 16, Number 2, pages 2 to 24, June 2005.

[0100] In the following step 209 (block 107), the broadband spectral envelope and the excitation signal are used for spectrally coloring the excitation signal. This can be achieved by multiplication in the sub-band or Fourier domain:

$$\tilde{\mathbf{Y}}_{ext}(n) = \operatorname{diag}\{\mathbf{C}_{opt,b}(n)\}\mathbf{X}_{exc}(n).$$

[0101] The diagonal matrix diag $\{\mathbf{C}_{opt,b}(n)\}$ has the form:

$$\operatorname{diag}\{\mathbf{C}_{opt,b}(n)\} = \begin{bmatrix} C_{opt,b}(e^{j\Omega_0},n) & 0 & \cdots & 0 \\ 0 & C_{opt,b}(e^{j\Omega_1},n) & \cdots & 0 \\ \vdots & \vdots & \ddots & \vdots \\ 0 & 0 & \cdots & C_{opt,b}(e^{j\Omega_{N_{DFT}-1}},n) \end{bmatrix}.$$

[0102] Because of the non-linearity or the prediction error filtering when generating the excitation signal, the power of the acoustic signal need not be maintained. Therefore, a power adaptation may be performed:

$$\mathbf{Y}_{ext}(n) = K(n)\tilde{\mathbf{Y}}_{ext}(n).$$

[0103] The correction factor K can be chosen to be

$$K(n) = \frac{\sum_{\mu=\mu_l}^{\mu_u} \left| X_w\left(e^{j\Omega},n\right)\right|^2}{\sum_{\mu=\mu_l}^{\mu_u} \tilde{Y}_{erw}\left(e^{j\Omega_\mu},n\right)^2}$$

wherein $\Omega_{\mu l}$ and $\Omega_{\mu u}$ denote the same bandwidth limits as in the estimation of the long-term power spectrum above.

[0104] The current bandwidth limits are adapted in step 210 (block 108). According to one possibility, the bandwidth limits are determined starting with a comparison of the spectrum of the received acoustic signal and the broadband spectral envelope being reduced by a predefined constant:

$$\tilde{\Omega}_l(n) = \min\left\{\Omega_\mu \in \left\{\left|X_w\left(e^{j\Omega_\mu,n}\right)\right|^2 \geq \left|C_{opt,b}\left(e^{j\Omega_\mu,n}\right)^2\right| + K_C\right\}\right\},$$

$$\tilde{\Omega}_u(n) = \max\left\{\Omega_\mu \in \left\{\left|X_w\left(e^{j\Omega_\mu,n}\right)\right|^2 \geq \left|C_{opt,b}\left(e^{j\Omega_\mu,n}\right)^2\right| + K_C\right\}\right\}.$$

[0105] The parameter K_C can have the value:

$$K_C = -12\,\text{dB}.$$

[0106] In Fig. 5, an example for determining the bandwidth limits is illustrated. The above, intermediate limit values are given by the points of intersection between the lowered broadband spectral envelope and the spectrum of the received acoustic signal.

[0107] These intermediate limit values may be recursively smoothed to eliminate temporary mal-estimations. In this case, preferably, smoothing is performed only if speech activity is detected in the current signal frame.

$$\overline{\Omega}_l(n) = \begin{cases} \beta_{bandl}\overline{\Omega}_l(n-1) + (1-\beta_{bandl})\tilde{\Omega}_l(n), & \text{during speech activity,} \\ \overline{\Omega}_l(n-1), & \text{else,} \end{cases}$$

$$\overline{\Omega}_u(n) = \begin{cases} \beta_{bandl}\overline{\Omega}_u(n-1) + (1-\beta_{bandl})\tilde{\Omega}_u(n), & \text{during speech activity,} \\ \overline{\Omega}_u(n-1), & \text{else.} \end{cases}$$

[0108] Then, the received acoustic signal is passed through an adaptive band pass filter to retain only components within the current bandwidth limits (block 109) to obtain a spectral vector $\mathbf{Y}_{tel}(n)$. Similarly, the spectrally colored excitation signal is passed through a complementary adaptive band stop filter (block 110) so as to obtain a vector $\mathbf{Y}_{ext}(n)$.

[0109] An output signal with a standard bandwidth is generated (step 211) by starting with summing these two spectral vectors:

$$\mathbf{Y}(n) = \mathbf{Y}_{tel}(n) + \mathbf{Y}_{ext}(n).$$

[0110] The components of these vectors are generated as:

$$\mathbf{Y}_{tel}(n) = \mathbf{G}_{tel}(n)\mathbf{X}_{w}(n),$$

$$\mathbf{Y}_{ext}(n) = \mathbf{G}_{ext}(n)\mathbf{X}_{ext}(n),$$

wherein the weighting matrices $\mathbf{G}_{tel}(n)$ and $\mathbf{G}_{ext}(n)$ are diagonal matrices:

$$\mathbf{G}_{tel}(n) = \begin{bmatrix} G_{tel}(e^{j\Omega_0}, n) & 0 & \cdots & 0 \\ 0 & G_{tel}(e^{j\Omega_1}, n) & \cdots & 0 \\ \vdots & \vdots & \ddots & \vdots \\ 0 & 0 & \cdots & G_{tel}(e^{j\Omega_{N_{DFT}-1}}, n) \end{bmatrix},$$

$$\mathbf{G}_{ext}(n) = \begin{bmatrix} G_{ext}(e^{j\Omega_0}, n) & 0 & \cdots & 0 \\ 0 & G_{ext}(e^{j\Omega_1}, n) & \cdots & 0 \\ \vdots & \vdots & \ddots & \vdots \\ 0 & 0 & \cdots & G_{ext}(e^{j\Omega_{N_{DFT}-1}}, n) \end{bmatrix}.$$

[0111] The elements of the matrix $\mathbf{G}_{tel}(n)$ are determined as:

$$G_{tel}(e^{j\Omega_\mu}, n) = \begin{cases} 1, & \text{if } \overline{\Omega}_l(n) \le \Omega_\mu \le \overline{\Omega}_u(n), \\ 0, & \text{else.} \end{cases}$$

[0112] The weights of the complementary weighting matrix are determined so as to yield the unity matrix when summed:

$$G_{ext}(e^{j\Omega_\mu}, n) = 1 - G_{tel}(e^{j\Omega_\mu}, n).$$

[0113] Alternatively, the transitions at the bandwidth limits can be realized in a smoother way.

[0114] The resulting output spectrum $\mathbf{Y}(n)$, then, is transformed into the time domain via an inverse Fourier transform:

$$\mathbf{y}(n) = \mathrm{IDFT}\{\mathbf{Y}(n)\},$$

followed by windowing the resulting vector. Particularly when using the above-indicated values for $\mathbf{N}_{ana}$ and r and a Hann window, this window function can be used again to obtain windowed time domain vectors:

$$\mathbf{y}_w(n) = \mathbf{F}\mathbf{y}(n).$$

[0115] The resulting time domain vectors are, then, assembled using an overlap add method (as described in K. D. Kammeyer, K. Kroschel, *Digitale Signalverarbeitung*) to obtain the final output signal $\mathbf{y}(n)$.

[0116] In the above-described steps of the method, more complex filter bank systems may be used instead of the conventional discrete Fourier transform and inverse discrete Fourier transform (see, for example, P. P. Vaidyanathan, Multirate Systems and Filter Banks, Prentice Hall, Englewood Cliffs, NJ, USA, 1992).

[0117] Further alternatives to the above-described variants are possible as well. For example, the steps performed in the Fourier domain may also be performed in the time domain. Furthermore, equalizing the acoustic signal may be performed when adapting the narrowband codebook entries. Also, the above-described equalizing step may be augmented. For example, if an amplification or an attenuation is detected at particular frequencies, it may be adjusted within the bandwidth limits as well. In this case, the output vector $\mathbf{Y}_{tel}(n)$ is modified with the weighting matrix $\mathbf{H}_{\text{mod}}(n)$.

[0118] In addition to the above-described codebook analysis for estimating the broadband spectral envelopes, a so-called linear mapping (see B. Iser, G. Schmidt, *Bandwidth Extension of Telephony Speech*) may be used additionally.

[0119] Further modifications and variations of the present invention will be apparent to those skilled in the art in view of this description. Accordingly, the description is to be construed as illustrated only and is for the purpose of teaching those skilled in the art the general manner of carrying out the present invention. It is to be understood that the forms of the invention shown and described herein are to be taken as the presently preferred embodiments.

Claims

1. Method for providing an acoustic signal with extended bandwidth, comprising:

   (a) automatically determining a current upper and a current lower bandwidth limit of a received acoustic signal.
   (b) automatically determining at least one complementary signal to complement the received acoustic signal between a predefined lower broadband bandwidth limit and the current lower bandwidth limit and/or between the current upper bandwidth limit and a predefined upper broadband bandwidth limit, wherein the predefined lower broadband bandwidth limit is smaller than the current lower bandwidth limit and the predefined upper broadband bandwidth limit is larger than the current upper bandwidth limit,
   (c) automatically assembling the at least one complementary signal and the received acoustic signal to obtain an acoustic signal with extended bandwidth,

   wherein step (b) comprises determining a broadband spectral envelope signal and a broadband excitation signal between the lower and upper broadband bandwidth limits such that the product of spectral envelope signal and excitation signal corresponds to the received acoustic signal according to a predetermined criterion, and wherein step (a) comprises comparing the determined broadband spectral envelope signal and a spectrum of the received acoustic signal such that the current upper and the current lower bandwidth limit are iteratively adapted, wherein the comparing step comprises selecting the minimal and maximal frequency for which the spectrum is larger than or equal to the determined broadband spectral envelope signal plus a predetermined constant.

2. Method according to claim 1, wherein determining a broadband spectral envelope signal comprises selecting an envelope signal from a codebook according to a predetermined criterion.

3. Method according to claim 2, wherein selecting an envelope signal comprises equalizing the received acoustic signal and selecting an envelope signal from the codebook having minimal distance to the equalized acoustic signal according to a predetermined distance criterion, in particular, having a minimal cepstral distance.

4. Method according to claim 3, wherein the codebook comprises pairs of corresponding envelope signals, each pair comprising a broadband envelope signal between the lower and upper broadband bandwidth limits and a corresponding narrowband envelope signal between a lower narrowband bandwidth limit being larger than the lower broadband bandwidth limit and an upper narrowband bandwidth limit being smaller than the upper broadband bandwidth limit, and selecting an envelope signal comprises determining a narrowband envelope signal having

minimal distance to the equalized acoustic signal according to the predetermined distance criterion and selecting the corresponding broadband envelope signal of this pair.

5. Method according to claim 4, wherein the step of selecting an envelope signal is preceded by providing adapted narrowband codebook envelope signals being adapted to the current lower and upper bandwidth limits.

6. Method according to claim 5, wherein the providing step comprise* processing broadband codebook envelope signals using a tong-term power spectrum of the received acoustic signal.

7. Method according to one of the claims 1 - 6, wherein determining a broadband excitation signal is based on prediction error filtering and/or a nonlinear characteristic.

8. Method according to one of claims 1 - 7, wherein the at least one complementary signal is based on a product of the determined broadband spectral envelope and the determined broadband excitation signal, and step (c) comprises summing the received acoustic signal between the current tower and upper bandwidth limits and the at least one complementary signal being restricted to the band between the lower broadband bandwidth limit and a current lower bandwidth limit and/or to the band between the current upper bandwidth limit and the upper broadband bandwidth limit.

9. Method according to one of the preceding claims, wherein at least one of the steps is performed in the cepstral domain.

10. Method according to one of the preceding claims, wherein steps (a) to (c) are repeated at predetermined time intervals.

11. Method according to one of the preceding claims, wherein steps (a) to (c) are repeated only if a wanted signal component, In particular, speech activity, is detected In the received acoustic signal.

12. Computer program product comprising one or more computer readable media having computer-executable Instructions for performing the steps of the method of one of the preceding claims when run on a computer.

13. Apparatus for providing an acoustic signal with extended bandwidth, comprising:

bandwidth determining means for automatically determining a current upper and a current lower bandwidth limit of a received acoustic signal,
complementary signal means for automatically determining at least one complementary signal to complement the received acoustic signal between a predefined lower broadband bandwidth limit and the current lower bandwidth limit and/or between the current upper bandwidth limit and a predefined upper broadband bandwidth limit wherein the predefined tower broadband bandwidth limit is smaller than the current lower bandwidth limit and the predefined upper broadband bandwidth limit is larger than the current upper bandwidth limit, and
assembling means for automatically assembling the at least one complementary signal and the received acoustic signal to obtain an acoustic signal with extended bandwidth,

wherein the complementary signal means comprises a means for determining a broadband spectral envelope signal and a broadband excitation signal between the lower and upper broadband bandwidth limits such that the product of spectral envelope signal and excitation signal corresponds to the received acoustic signal according to a predetermined criterion,
wherein the bandwidth determining means is configured to compare the determined broadband spectral envelope signal and a spectrum of the received acoustic signal such that the current upper and the current lower bandwidth limit are iteratively adapted, and wherein the bandwidth determining means is configured to select the minimal and maximal frequency for which the spectrum is larger than or equal to of the determined broadband spectral envelope signal plus a predetermined constant.

14. Apparatus according to claims 13 wherein the means for determining a broadband spectral envelope signal comprises a means for selecting an envelope signal from a codebook according to a predetermined criterion.

15. Apparatus according to claim 14, wherein the means for selecting an envelope signal is configured to equalize the received acoustic signal and select an envelope signal from the codebook having minimal distance to the equalized acoustic signal according to a predetermined distance criterion, in particular, having a minimal cepstral distance.

16. Apparatus according to claim 15, wherein the codebook comprises pairs of corresponding envelope signals, each pair comprising a broadband envelope signal between the lower and upper broadband bandwidth limits and a corresponding narrowband envelope signal between a lower narrowband bandwidth limit being larger than the lower broadband bandwidth limit and an upper narrowband bandwidth limit being smaller than the upper broadband bandwidth limit, and the means for selecting an envelope signal is configured to determine a narrowband envelope signal having minimal distance to the equalized acoustic signal according to the predetermined distance criterion and to select the corresponding broadband envelope signal of this pair.

17. Apparatus according to claim 16, wherein the means for determining a broadband spectral envelope signal comprises a means for providing adapted narrowband codebook envelope signals being adapted to the current lower and upper bandwidth limits.

18. Apparatus according to claim 17, wherein the means for providing is configured to process the broadband codebook envelope signal using a long-term power spectrum of the received acoustic signal.

19. Apparatus according to one of the claims 13 - 18, wherein the means for determining a broadband excitation signal is configured to determine the broadband excitation signal based on prediction error tittering and/or a nonlinear characteristic.

20. Apparatus according to one of claims 13 - 19, wherein the at least one complementary signal is based on a product of the determined broadband spectral envelope and the determined broadband excitation signal, and the assembling means is configured to sum the received acoustic signal between the current lower and upper bandwidth limits and the at least one complementary signal being restricted to the band between the lower broadband bandwidth limit and a current lower bandwidth limit and/or to the band between the current upper bandwidth limit and the upper broadband bandwidth limit.

21. Apparatus according to one of the claims 13-20, wherein at least one of the means is configured to perform at least part of its function in the cepstral domain.

22. Apparatus according to one of the claims 13-21, wherein the means are configured to perform their respective function repeatedly at predetermined time intervals.

23. Apparatus according to one of the claims 13-22, further comprising a wanted signal detector, In particular, a speech detector, and wherein the means are configured to perform their respective function only if a wanted signal component is detected in the received acoustic signal.

Patentansprüche

1. Verfahren zum Bereitstellen eines akustischen Signals mit erweiterter Bandbreite, das umfasst:

   (a) automatisches Bestimmen einer aktuellen oberen und aktuellen unteren Bandbreitengrenze eines empfangenen akustischen Signals,
   (b) automatisches Bestimmen wenigstens eines Komplementärsignals, das das empfangene akustische Signal ergänzt, zwischen einer vordefinierten unteren Breitband-Bandbreitengrenze und der aktuellen unteren Bandbreitengrenze und/oder zwischen der aktuellen oberen Bandbreiten-Grenze und einer vordefinierten oberen Breitband-Bandbreitengrenze, wobei die vordefinierte untere Breitband-Bandbreitengrenze niedriger ist als die aktuelle untere Bandbreitengrenze, und die vordefinierte obere Breitband-Bandbreitengrenze höher ist als die aktuelle obere Bandbreitengrenze,
   (c) automatisches Zusammensetzen des wenigstens einen Komplementärsignals und des empfangenen akustischen Signals, um ein akustisches Signal mit erweiterter Bandbreite zu gewinnen,

   wobei Schritt (b) umfasst, dass ein Breitband-Spektral-Hüllkurvensignal und ein Breitband-Erregungssignal zwischen der unteren und der oberen Breitband-Bandbreitengrenze so bestimmt werden, dass das Produkt des Spektral-Hüllkurvensignals und des Erregungssignals dem empfangenen akustischen Signal gemäß einem vorgegebenen Kriterium entspricht; und
   wobei Schritt (a) Vergleichen des bestimmten Breitband-Spektral-Hüllkurvensignals und eines Spektrums des empfangenen akustischen Signals umfasst, so dass die aktuelle obere und die aktuelle untere Bandbreitengrenze iterativ

angepasst werden, wobei der Vergleichsschritt Auswählen der minimalen und maximalen Frequenz umfasst, für die das Spektrum größer ist als oder genauso groß wie das bestimmte Breitband-Spektral-Hüllkurvensignal zuzüglich einer vorgegebenen Konstante.

2. Verfahren nach Anspruch 1, wobei Bestimmen eines Breitband-Spektral-Hüllkurvensignals Auswählen eines Hüllkurvensignals aus einem Codebuch gemäß einem vorgegebenen Kriterium umfasst.

3. Verfahren nach Anspruch 2, wobei Auswählen eines Hüllkurvensignals Entzerren des empfangenen akustischen Signals und Auswählen eines Hüllkurvensignals aus dem Codebuch umfasst, das gemäß einem vorgegebenen Distanz-Kriterium minimale Distanz zu dem entzerrten akustischen Signal, insbesondere eine minimale cepstrale Distanz, hat.

4. Verfahren nach Anspruch 3, wobei das Codebuch Paare entsprechender Hüllkurvensignale umfasst, wobei jedes Paar ein Breitband-Hüllkurvensignal zwischen der unteren und der oberen Breitband-Bandbreitengrenze sowie ein entsprechendes Schmalband-Hüllkurvensignal zwischen einer unteren Schmalband-Bandbreitengrenze umfasst, die höher ist als die untere Breitband-Bandbreitengrenze, und einer oberen Schmalband-Bandbreitengrenze umfasst, die niedriger ist als die obere Breitband-Bandbreitengrenze, und Auswählen eines Hüllkurvensignals Bestimmen eines Schmalband-Hüllkurvensignals, das gemäß dem vorgegebenen Distanz-Kriterium minimale Distanz zu dem entzerrten akustischen Signal hat, sowie Auswählen des entsprechenden Breitband-Hüllkurvensignals dieses Paars umfasst.

5. Verfahren nach Anspruch 4, wobei dem Schritt des Auswählens eines Hüllkurvensignals Bereitstellen angepasster Schmalband-Codebuch-Hüllkurvensignale vorangeht, die an die aktuelle untere und obere Bandbreitengrenze angepasst sind.

6. Verfahren nach Anspruch 5, wobei der Schritt des Bereitstellens Verarbeiten von Breitband-Codebuch-Hüllkurvensignalen unter Verwendung eines Langzeit-Leistungsspektrums des empfangenen akustischen Signals umfasst.

7. Verfahren nach einem der Ansprüche 1-6, wobei Bestimmen eines Breitband-Erregungssignals auf Prädiktions-Fehlerfilterung und/oder einer nicht linearen Charakteristik basiert.

8. Verfahren nach einem der Ansprüche 1-7, wobei das wenigstens eine Komplementärsignal auf einem Produkt der bestimmten Breitband-Spektral-Hüllkurve und des bestimmten Breitband-Erregungssignals basiert und Schritt (c) Summieren des empfangenen akustischen Signals zwischen der aktuellen unteren und oberen Bandbreiten-Grenze und des wenigstens einen Komplementärsignals umfasst, das auf das Band zwischen der unteren Breitband-Bandbreitengrenze und einer aktuellen unteren Bandbreitengrenze und/oder auf das Band zwischen der aktuellen oberen Bandbreitengrenze und der oberen Breitband-Bandbreitengrenze beschränkt ist.

9. Verfahren nach einem der vorangehenden Ansprüche, wobei wenigstens einer der Schritte in der cepstralen Domäne durchgeführt wird.

10. Verfahren nach einem der vorangehenden Ansprüche, wobei die Schritte (a) bis (c) in vorgegebenen Zeitintervallen wiederholt werden.

11. Verfahren nach einem der vorangehenden Ansprüche, wobei die Schritte (a) bis (c) nur wiederholt werden, wenn eine erwünschte Signalkomponente, insbesondere Sprachaktivität, in dem empfangenen akustischen Signal erfasst wird.

12. Computerprogrammerzeugnis, das ein oder mehrere computerlesbare Medium/Medien umfasst, das/die durch Computer ausführbare Befehle zum Durchführen der Schritte des Verfahrens nach einem der vorangehenden Ansprüche bei Ausführung auf einem Computer umfasst/umfassen.

13. Vorrichtung zum Bereitstellen eines akustischen Signals mit erweiterter Bandbreite, die umfasst:

eine Bandbreiten-Bestimmungseinrichtung zum automatischen Bestimmen einer aktuellen oberen und einer aktuellen unteren Bandbreitengrenze eines empfangenen akustischen Signals,
eine Komplementärsignal-Einrichtung zum automatischen Bestimmen wenigstens eines Komplementärsignals, das das akustische Signal ergänzt, zwischen einer vordefinierten unteren Breitband-Bandbreitengrenze und

der aktuellen unteren Bandbreitengrenze und/oder zwischen der aktuellen oberen Bandbreitengrenze und einer vordefinierten oberen Breitband-Bandbreitengrenze, wobei die vordefinierte untere Breitband-Bandbreitengrenze niedriger ist als die aktuelle untere Bandbreitengrenze und die vordefinierte obere Breitband-Bandbreitengrenze höher ist als die aktuelle obere Bandbreitengrenze, und
eine Zusammensetzeinrichtung zum automatischen Zusammensetzen des wenigstens einen Komplementärsignals und des empfangenen akustischen Signals, um ein akustisches Signal mit erweiteter Bandbreite zu gewinnen,

wobei die Komplementärsignal-Einrichtung eine Einrichtung umfasst, mit der ein Breitband-Spektral-Hüllkurvensignal und ein Breitband-Erregungssignal zwischen der unteren und der oberen Breitband-Bandbreitengrenze so bestimmt werden, dass das Produkt des Spektral-Hüllkurvensignals und des Erregungssignals dem empfangenen akustischen Signal gemäß einem vorgegebenen Kriterium entspricht,
wobei die Bandbreiten-Bestimmungseinrichtung so konfiguriert ist, dass sie das bestimmte Breitband-Spektral-Hüllkurvensignal und ein Spektrum des empfangenen akustischen Signals vergleicht, so dass die aktuelle obere und die aktuelle untere Bandbreitengrenze iterativ angepasst werden, und wobei die Bandbreiten-Bestimmungseinrichtung so konfiguriert ist, dass sie die minimale und maximale Frequenz auswählt, für die das Spektrum größer ist als oder genauso groß wie das bestimmte Breitband-Spektral-Hüllkurvensignal zuzüglich einer vorgegebenen Konstante.

14. Vorrichtung nach Anspruch 13, wobei die Einrichtung, mit der ein Breitband-Spektral-Hüllkurvensignal bestimmt wird, eine Einrichtung zum Auswählen eines Hüllkurvensignals aus einem Codebuch gemäß einem vorgegebenen Kriterium umfasst.

15. Vorrichtung nach Anspruch 14, wobei die Einrichtung zum Auswählen eines Hüllkurvensignals so konfiguriert ist, dass sie das empfangene akustische Signal entzerrt und ein Hüllkurvensignal aus dem Codebuch auswählt, das gemäß einem vorgegebenen Distanz-Kriterium minimale Distanz zu dem entzerrten akustischen Signal, insbesondere eine minimale cepstrale Distanz, hat.

16. Vorrichtung nach Anspruch 15, wobei das Codebuch Paare entsprechender Hüllkurvensignale umfasst und jedes Paar ein Breitband-Hüllkurvensignal zwischen der unteren und der oberen Bandbreitengrenze sowie ein entsprechendes Schmalband-Hüllkurvensignal zwischen einer unteren Schmalband-Bandbreitengrenze, die höher ist als die untere Breitband-Bandbreitengrenze, und einer oberen Schmalband-Bandbreitengrenze umfasst, die niedriger ist als die obere Breitband-Bandbreitengrenze, und die Einrichtung zum Auswählen eines Hüllkurvensignals so konfiguriert ist, dass sie ein Schmalband-Hüllkurvensignal bestimmt, das gemäß dem vorgegebenen Distanz-Kriterium minimale Distanz zu dem entzerrten akustischen Signal hat, und das entsprechende Breitband-Hüllkurvensignal dieses Paars auswählt.

17. Vorrichtung nach Anspruch 16, wobei die Einrichtung zum Bestimmen eines Breitband-Spektral-Hüllkurvensignals eine Einrichtung zum Bereitstellen angepasster Schmalband-Codebuch-Hüllkurvensignale umfasst, die an die aktuelle untere und obere Bandbreitengrenze angepasst sind.

18. Vorrichtung nach Anspruch 17, wobei die Einrichtung zum Erzeugen so konfiguriert ist, dass sie das Breitband-Codebuch-Hüllkurvensignal unter Verwendung eines Langzeit-Leistungsspektrums des empfangenen akustischen Signals verarbeitet.

19. Vorrichtung nach einem der Ansprüche 13-18, wobei die Einrichtung zum Bestimmen eines Breitband-Erregungssignals so konfiguriert ist, dass sie das Breitband-Erregungssignal auf Basis von Prädiktions-Fehlerfilterung und/oder einer nicht linearen Charakteristik bestimmt.

20. Vorrichtung nach einem der Ansprüche 13-19, wobei das wenigstens eine Komplementärsignal auf einem Produkt der bestimmten Breitband-Spektral-Hüllkurve und des bestimmten Breitband-Erregungssignals basiert und die Zusammensetzeinrichtung so konfiguriert ist, dass sie das empfangene akustische Signal zwischen der aktuellen unteren und oberen Bandbreitengrenze und das wenigstens eine Komplementärsignal summiert, das auf das Band zwischen der unteren Breitband-Bandbreitengrenze und einer aktuellen unteren Bandbreitengrenze und/oder das Band zwischen der aktuellen oberen Bandbreitengrenze und der oberen Breitband-Bandbreitengrenze beschränkt ist.

21. Vorrichtung nach einem der Ansprüche 13-20, wobei wenigstens eine der Einrichtungen so konfiguriert ist, dass sie wenigstens einen Teil ihrer Funktion in der cepstralen Domäne erfüllt.

22. Vorrichtung nach einem der Ansprüche 13-21, wobei die Einrichtungen so konfiguriert sind, dass sie ihre Funktion wiederholt in vorgegebenen Zeitintervallen erfüllen.

23. Vorrichtung nach einem der Ansprüche 13-22, die des Weiteren einen Detektor für ein erwünschtes Signal, insbesondere einen Sprachdetektor, umfasst, und wobei die Einrichtungen so konfiguriert sind, dass sie ihre jeweilige Funktion nur erfüllen, wenn eine erwünschte Signalkomponente in dem empfangenen akustischen Signal erfasst wird.

Revendications

1. Procédé pour mettre à disposition un signal acoustique avec une largeur de bande étendue, comprenant :

a) la détermination automatique d'une limite de largeur de bande courante plus basse et plus haute d'un signal acoustique reçu,
b) la détermination automatique d'au moins un signal complémentaire pour compléter le signal acoustique reçu entre une limite prédéterminée de largeur de bande à bande large plus basse et la limite de largeur de bande courante plus basse et/ou entre la limite de largeur de bande courante plus haute et une limite prédéterminée de largeur de bande à bande large plus haute, la limite prédéterminée de largeur de bande à bande large plus basse étant plus petite que la limite de largeur de bande courante basse et la limite prédéterminée de largeur de bande à bande large plus haute étant plus grande que la limite de largeur de bande actuelle plus haute,
c) l'assemblage automatique du, au moins un, signal complémentaire et du signal acoustique reçu pour obtenir un signal acoustique avec une largeur de bande étendue,

dans lequel l'étape (b) comprend la détermination d'un signal d'enveloppe spectrale à bande large et un signal d'excitation à bande large entre les limites de largeur de bande plus haute et plus basse tel que le produit du signal d'enveloppe spectrale et le signal d'excitation correspond au signal acoustique reçu selon un critère prédéterminé, et dans lequel l'étape (a) comprend la comparaison du signal d'enveloppe spectrale à bande large déterminé et d'un spectre du signal acoustique reçu telle que la limite de largeur de bande courante plus haute et la limite de largeur de bande courante plus basse soient itérativement adaptées, et où l'étape de comparaison comprend la sélection de la fréquence minimale et maximale pour lesquelles le spectre est plus grand ou égal au signal d'enveloppe spectrale à bande large déterminé ajouté à une constante prédéterminée.

2. Procédé selon la revendication 1, dans lequel la détermination d'un signal d'enveloppe spectrale à bande large comprend la sélection d'un signal d'enveloppe d'un livre code selon un critère prédéterminé.

3. Procédé selon la revendication 2, où la sélection d'un signal d'enveloppe comprend l'égalisation du signal acoustique reçu et la sélection d'un signal d'enveloppe à partir du livre code ayant une distance minimale du signal acoustique égalisé selon un critère distance prédéterminé, en particulier ayant une distance cepstral minimale.

4. Procédé selon la revendication 3, où le livre code comprend des paires de signaux d'enveloppe correspondants, chaque paire comprenant un signal d'enveloppe à bande large entre les limites de largeur de bande à bande large plus basse et plus haute et un signal d'enveloppe à bande étroite correspondant entre une limite de largeur de bande à bande basse plus basse étant plus large que la limite de largeur de bande à bande large basse et une limite de largeur de bande à bande base plus haute étant plus petite que la limite de largeur de bande à bande large plus haute, et la sélection d'un signal d'enveloppe comprend la détermination d'un signal d'enveloppe à bande basse ayant une distance minimale du signal acoustique égalisé selon le critère de distance prédéterminée et la sélection du signal d'enveloppe à bande large correspondant de cette paire.

5. Procédé selon la revendication 4, dans lequel l'étape de sélection d'un signal d'enveloppe est anticipée par la mise à disposition de signaux d'enveloppe de livre code à bande étroite adaptés étant adaptés aux limites de largeur de bande plus basse et plus haute courantes.

6. Procédé selon la revendication 5, dans lequel l'étape de mise à disposition comprend le traitement des signaux d'enveloppe de livre code à bande large en utilisant un spectre de puissance à long terme du signal acoustique reçu.

7. Procédé selon une des revendications 1 à 6, dans lequel la détermination d'un signal d'excitation à bande large est basé sur un filtrage d'erreur de prédiction et/ou une caractéristique non linéaire.

8. Procédé selon une des revendications 1 à 7, dans lequel le, au moins un, signal complémentaire est basé sur le produit de l'enveloppe spectrale à bande large déterminée et le signal d'excitation à bande large déterminé, et où l'étape (c) comprend la sommation du signal acoustique reçu entre les limites de largeur de bande plus basse et plus haute courantes et le, au moins un, signal complémentaire étant restreint à la bande entre la limite de largeur de bande à bande large plus basse et la limite de largeur de bande plus basse courante et/ou à la bande entre la limite de largeur de bande plus haute courante et la limite de largeur de bande à bande large plus haute.

9. Procédé selon une des revendications précédentes, dans lequel au moins une des étapes est réalisée dans le domaine cepstral.

10. Procédé selon une des revendications précédentes, dans lequel les étapes (a) à (c) sont répétées à des intervalles temporels prédéterminés.

11. Procédé selon une des revendications précédentes, dans lequel les étapes (a) à (c) sont répétées seulement si un composant d'un signal désiré, en particulier activité de parole, est détecté dans le signal acoustique reçu.

12. Produit de programme d'ordinateur comprenant un ou plusieurs médias lisible par ordinateur ayant des instructions pouvant être exécutées sur ordinateur pour réaliser les étapes du procédé selon une des revendications précédentes lorsqu'elles sont réalisées sur ordinateur.

13. Appareil pour mettre à disposition un signal acoustique avec une largeur de bande étendue, comprenant :

un moyen de détermination de largeur de bande pour déterminer automatiquement une limite de largeur de bande actuelle plus haute et plus basse d'un signal acoustique reçu,
un moyen de signal complémentaire pour déterminer automatiquement au moins un signal complémentaire pour compléter le signal acoustique reçu entre une limite prédéterminée de largeur de bande à bande large plus basse et la limite de largeur de bande courante plus basse et/ou entre la limite de largeur de bande courante plus haute et une limite prédéterminée de largeur de bande à bande large plus haute, où la limite prédéterminée de largeur de bande à bande large plus basse est plus petite que la limite de largeur de bande courante plus basse et la limite prédéterminée de largeur de bande à bande large plus haute est plus grande que la limite de largeur de bande courante plus haute, et
un moyen d'assemblage pour l'assemblage automatique du, au moins un, signal complémentaire et du signal acoustique reçu pour obtenir un signal acoustique avec une largeur de bande étendue,

dans lequel le moyen de signal complémentaire comprend un moyen pour déterminer un signal d'enveloppe spectrale à bande large et un signal d'excitation à bande large entre les limites de largeur de bande plus haute et plus basse telle que le produit du signal d'enveloppe spectrale et le signal d'excitation correspond au signal acoustique reçu selon un critère prédéterminé, et
dans lequel le le moyen de détermination de largeur de bande est configuré pour comparer le signal d'enveloppe spectrale à bande large déterminé et un spectre du signal acoustique reçu telle que la limite de largeur de bande courante plus haute et la limite de largeur de bande courante basse soient itérativement adaptées, et où le moyen de détermination de largeur de bande est configuré pour sélectionner la fréquence minimale et maximale pour lesquelles le spectre est plus grand ou égal au signal d'enveloppe spectrale à bande large déterminé plus une constante prédéterminée.

14. Appareil selon la revendication 13, dans lequel le moyen de détermination d'un signal d'enveloppe spectrale à bande large comprend un moyen pour sélectionner un signal d'enveloppe d'un livre code selon un critère prédéterminé.

15. Appareil selon la revendication 14, dans lequel le moyen pour sélectionner un signal d'enveloppe est configuré pour égaliser le signal acoustique reçu et pour sélectionner un signal d'enveloppe à partir du livre code ayant une distance minimale du signal acoustique égalisé selon un critère distance prédéterminé, en particulier ayant une distance cepstrale minimale.

16. Appareil selon la revendication 15, où le livre code comprend des paires de signaux d'enveloppe correspondants,

chaque paire comprenant un signal d'enveloppe à bande large entre les limites de largeur de bande à bande largue plus basse et plus haute et un signal d'enveloppe à bande étroite correspondant entre une limite de largeur de bande à bande basse plus basse étant plus large que la limite de largeur de bande à bande large basse et une limite de largeur de bande à bande base plus haute étant plus petite que la limite de largeur de bande à bande large plus haute, et le moyen de sélection d'un signal d'enveloppe est configuré pour déterminer un signal d'enveloppe à bande basse ayant une distance minimale du signal acoustique égalisé selon le critère de distance prédéterminé et pour sélectionner le signal d'enveloppe à bande large correspondant de cette paire.

17. Appareil selon la revendication 16, dans lequel le moyen pour déterminer un signal d'enveloppe à bande large comprend un moyen pour mettre à disposition des signaux d'enveloppe de livre code à bande étroite adaptés étant adaptés aux limites de largeur de bande plus basse et plus haute actuelles.

18. Appareil selon la revendication 17, dans lequel le moyen de mise à disposition est configuré pour traiter le signal d'enveloppe de livre code à bande large en utilisant un spectre de puissance à long terme du signal acoustique reçu.

19. Appareil selon une des revendications 13 à 18, dans lequel le moyen pour déterminer un signal d'excitation à bande large est configuré pour déterminer le signal d'excitation à bande large basé sur un filtrage d'erreur de prédiction et/ou une caractéristique non linéaire.

20. Appareil selon une des revendications 13 à 19, dans lequel le, au moins un, signal complémentaire est basé sur le produit de l'enveloppe spectrale à bande large déterminée et le signal d'excitation à bande large déterminé, et où le moyen d'assemblage est configuré pour sommer le signal acoustique reçu entre les limites de largeur de bande plus basse et plus haute courantes et le, au moins un, signal complémentaire étant restreint à la bande entre la limite de largeur de bande à bande large plus basse et la limite de largeur de bande plus basse actuelle et/ou à la bande entre la limite de largeur de bande plus haute actuelle et la limite de largeur de bande à bande large plus haute.

21. Appareil selon une des revendications 13 à 20, dans lequel au moins un des moyens est configuré pour réaliser au moins une partie de sa fonction dans le domaine cepstral.

22. Appareil selon une des revendications 13 à 21, où les moyens sont configurés pour réaliser leur fonction respective d'une façon répétitive à des intervalles temporels prédéterminés.

23. Appareil selon une des revendications 13 à 22, comprenant en outre un capteur de signal désiré, en particulier un détecteur de parole, et où les moyens sont configurés pour réaliser leur fonction respective seulement si un composant d'un signal désiré est détecté dans le signal acoustique reçu.

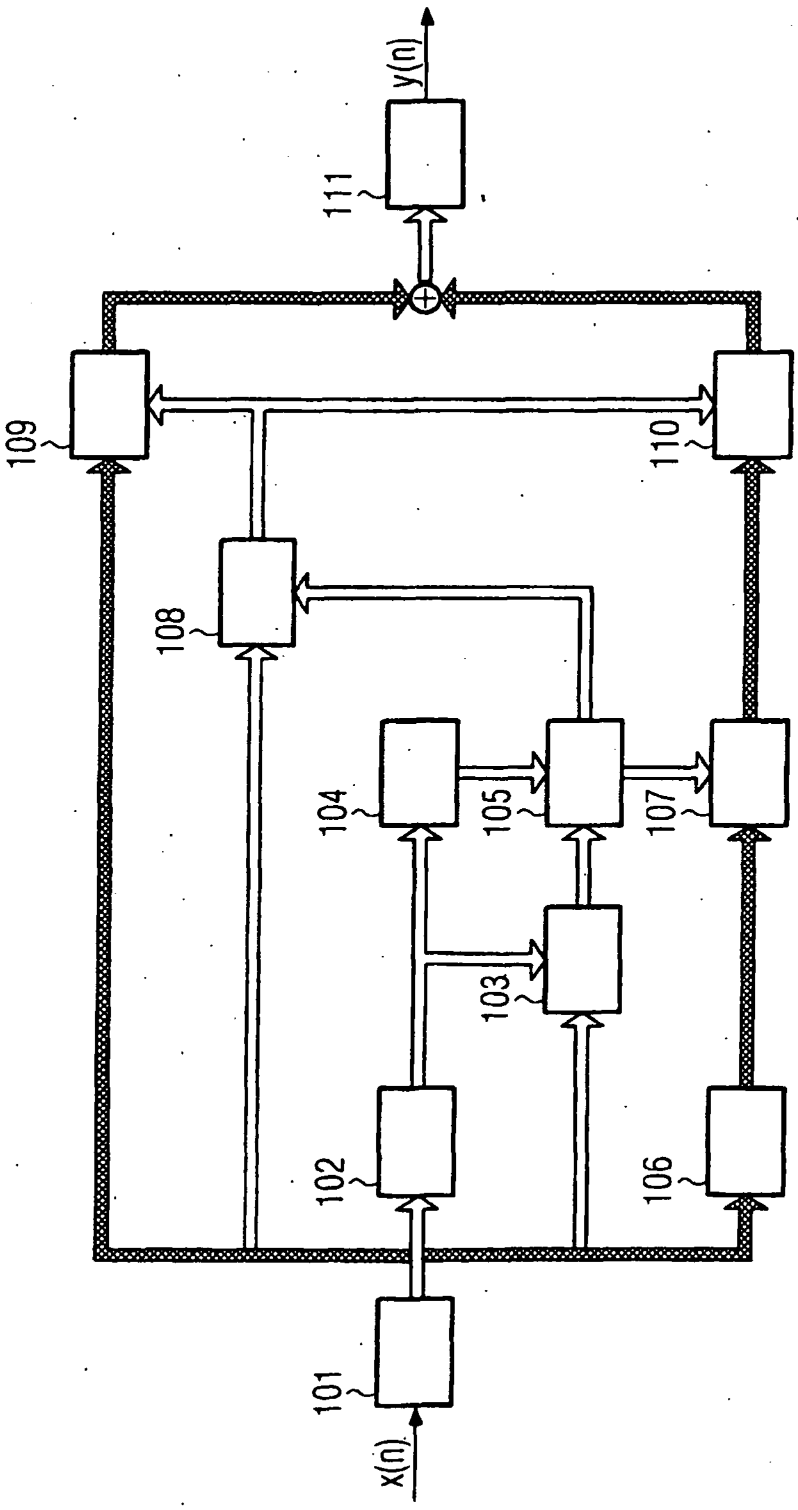


FIG. 1

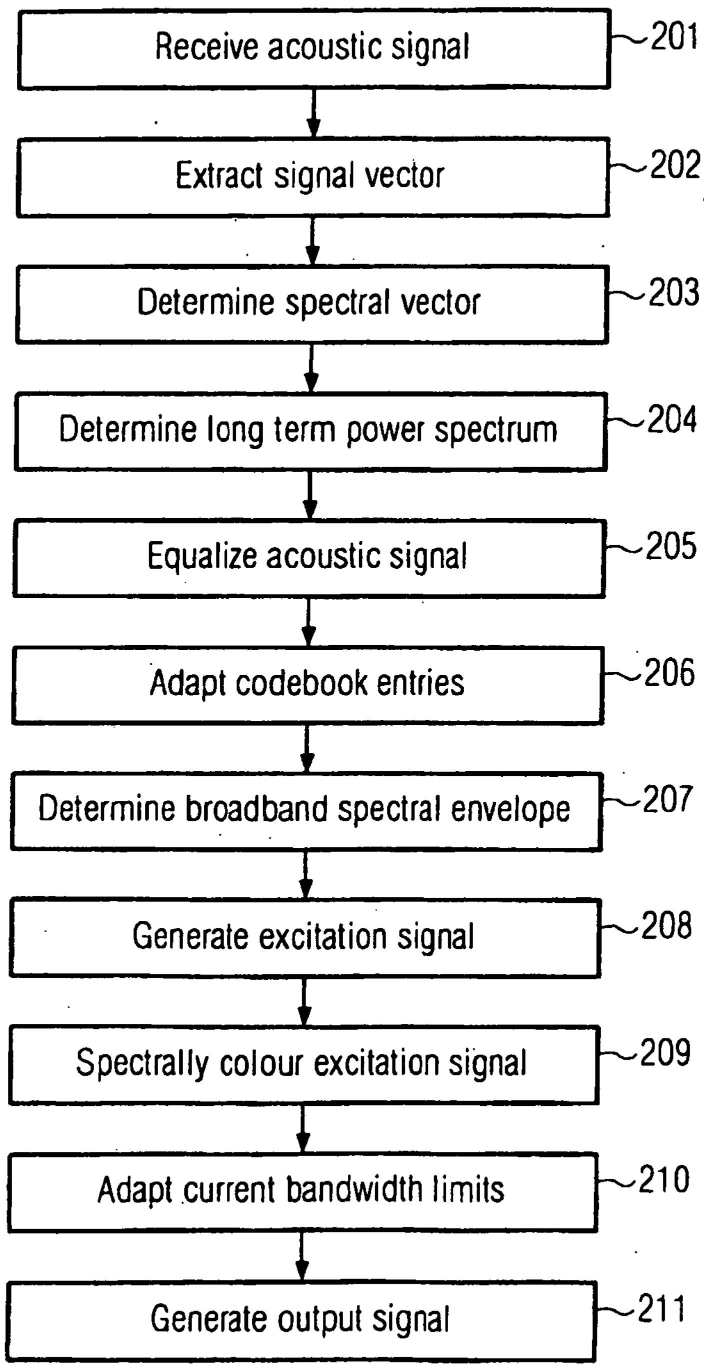

Receive acoustic signal
201
Extract signal vector
202
Determine spectral vector
203
Determine long term power spectrum
204
Equalize acoustic signal
205
Adapt codebook entries
206
Determine broadband spectral envelope
207
Generate excitation signal
208
Spectrally colour excitation signal
209
Adapt current bandwidth limits
210
Generate output signal
211

FIG. 2

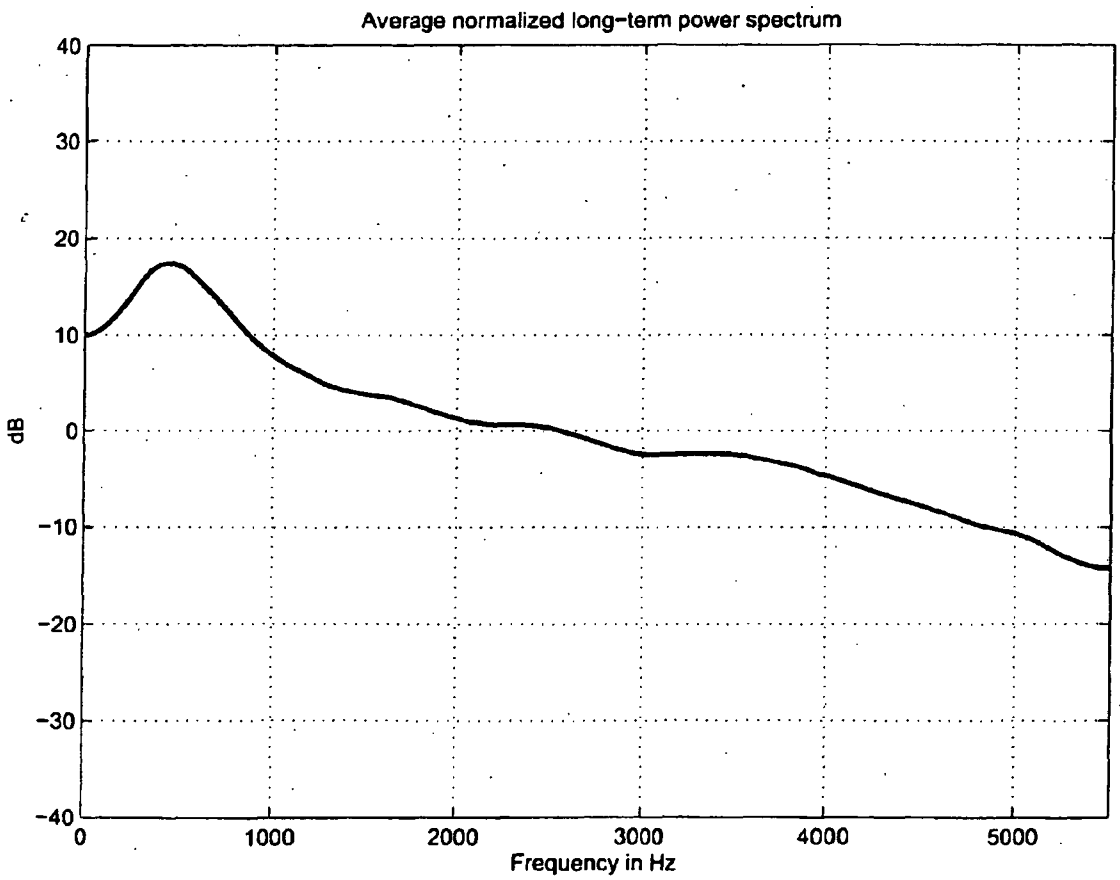

Average normalized long-term power spectrum
40
30
20
10
0
-10
-20
-30
-40
dB
0
1000
2000
3000
4000
5000
Frequency in Hz

Fig. 3

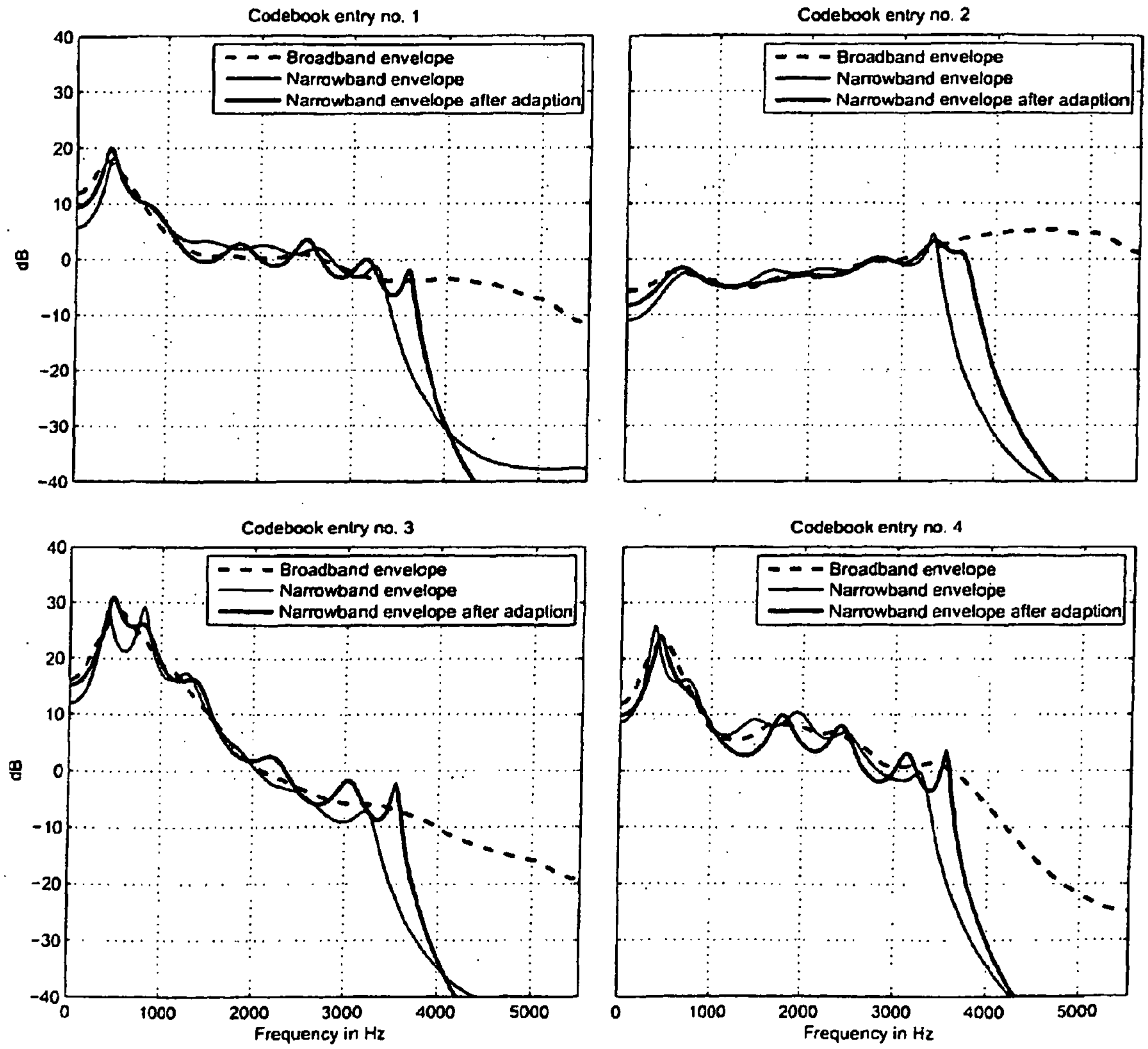
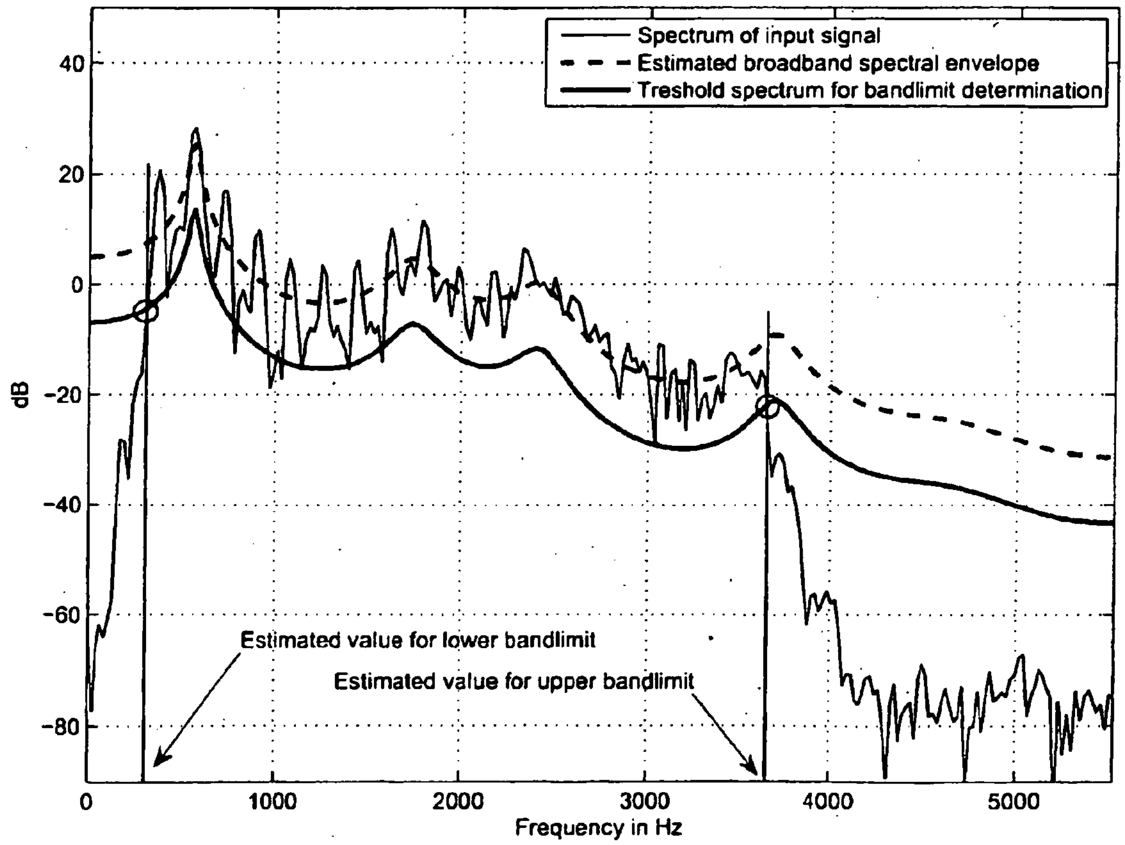

Codebook entry no. 1
Codebook entry no. 2
Codebook entry no. 3
Codebook entry no. 4
Broadband envelope
Narrowband envelope
Narrowband envelope after adaption
dB
Frequency in Hz
40
30
20
10
0
-10
-20
-30
-40
0
1000
2000
3000
4000
5000

Fig. 4



Spectrum of input signal
Estimated broadband spectral envelope
Treshold spectrum for bandlimit determination
40
20
0
-20
-40
-60
-80
dB
Estimated value for lower bandlimit
Estimated value for upper bandlimit
0
1000
2000
3000
4000
5000
Frequency in Hz

Fig. 5

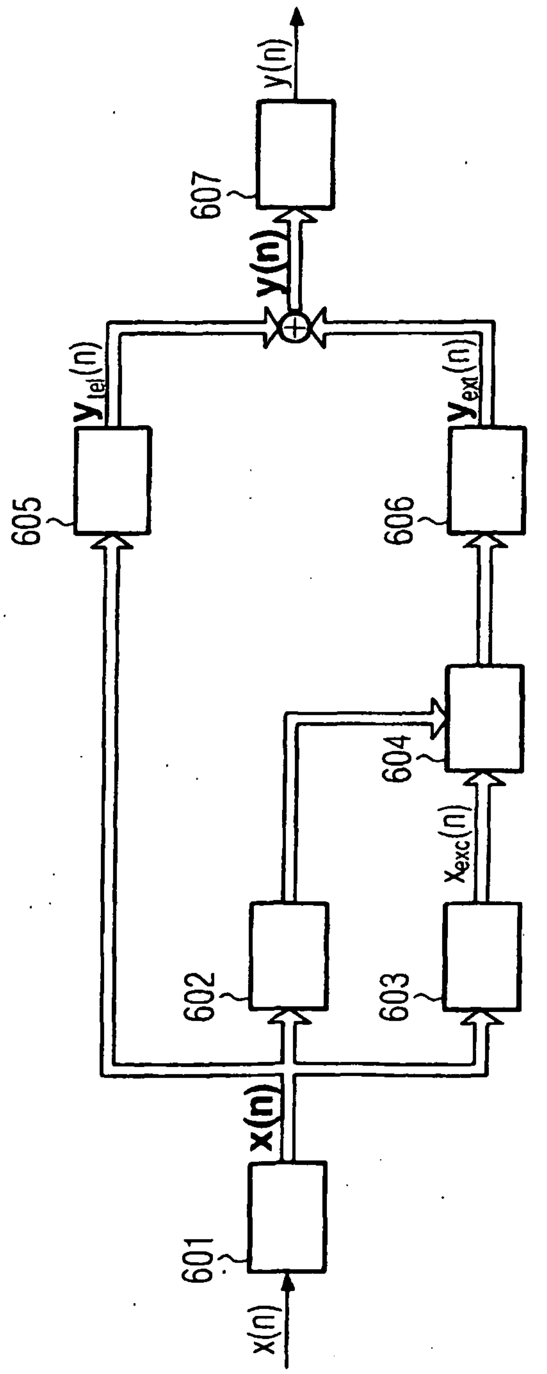


FIG. 6
(prior art)

REFERENCES CITED IN THE DESCRIPTION

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