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Remarks:
Amended claims in accordance with Rule 86 (2) EPC.

(54) **Method for extending the spectral bandwidth of a speech signal and system thereof**

(57)   The invention relates to a method for extending the spectral bandwidth of a bandwidth limited speech signal which comprises at least harmonics of a fundamental frequency, wherein a nonlinear function is applied to the bandwidth limited speech signal for generating the lower frequency components of the speech signal which are attenuated in the bandwidth limited speech signal.
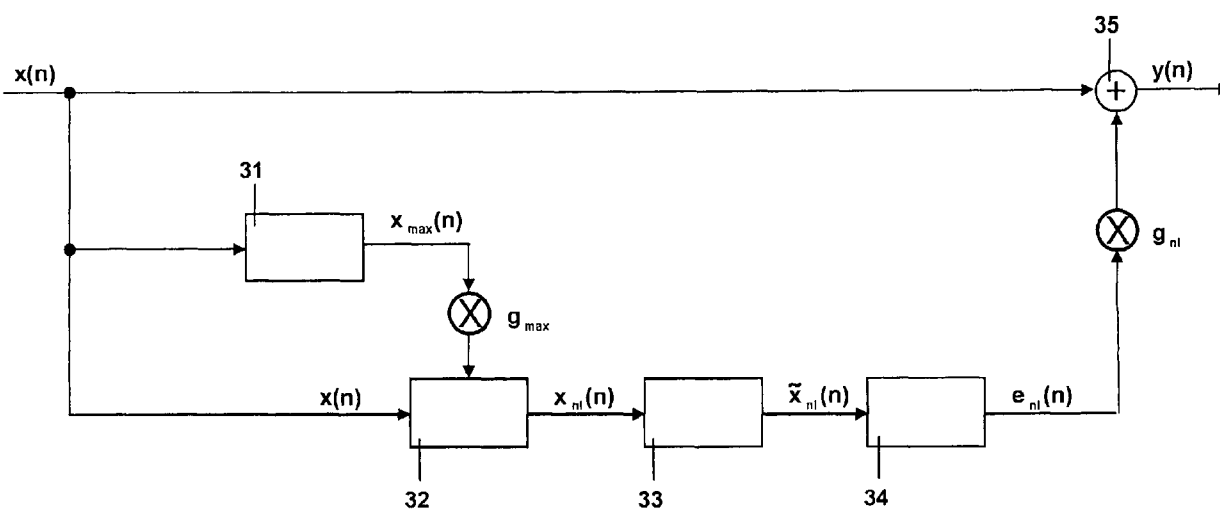
Fig. 3

EP 1 814 107 A1

Description

[0001] This invention relates to a method for extending the spectral bandwidth of a speech signal.

[0002] Speech is the most natural and convenient way of human communication. This is one reason for the great success of the telephone system since its invention in the 19th century. Today, subscribers are not always satisfied any more with the quality of the service provided by the telephone system especially when compared to other audio sources, such as radio, compact disk or DVD. The degradation of speech quality using analogue telephone systems is caused by the introduction of band limiting filters within amplifiers used to keep a certain signal level in long local loops. These filters have a passband from approximately 300 Hz up to 3400 Hz and are applied to reduce crosstalk between different channels. However, the application of such bandpass filters considerably attenuates different frequency parts of the human speech ranging from about 50 Hz up to 6000 Hz. The missing frequency components in the range between about 3400 Hz to 6000 Hz influence the perceivability of the speech, whereas the missing lower frequency components from 50 Hz to 300 Hz result in a lower speech quality.

[0003] Great efforts have been made to increase the quality of telephone speech signals in recent years. One possibility to increase the quality of a telephone speech signal is to increase the bandwidth after transmission by means of bandwidth extension. The basic idea of these enhancements is to establish the speech signal components above 3400 Hz and below 300 Hz and to complement the signal with this estimate. In this case the telephone networks can remain untouched. In the art bandwidth extension methods are known in which the spectral envelope of the speech signal is determined and an excitation signal is generated by removing the envelope. In these methods codebook pairs and neuronal networks can be used. However these methods require large memory and processing capacities.

[0004] The prior art methods further have the drawback that for determining the envelope and for removing the latter signal components have to be averaged over time, so that the signal processing leads to a delay from signal input to signal output. Especially in telecommunication networks the delay of the signal is limited to a certain value in order not to deteriorate the speech quality for the subscriber at the other end of the line.

[0005] EP 0 994 464 discloses a method for extending the spectral bandwidth of a bandwidth limited speech signal in which the telephone signal is multiplied by a constant A when the telephone signal is positive and is multiplied with -A when the telephone signal is negative.

[0006] A need exists to provide a way of further improving the speech quality in telecommunication systems.

[0007] This need is met by the features of the independent claims. In the dependent claims preferred embodiments of the invention are described.

[0008] According to a first aspect of the invention, a method is provided for extending the spectral bandwidth of a bandwidth limited speech signal, the speech signal comprising at least harmonics of a fundamental frequency. According to the invention, a non-linear function is applied to the bandwidth limited speech signal for generating the lower frequency components of the speech signal which are attenuated in the bandwidth limited speech signal. This method has several advantages over known methods. First of all, it is not necessary to calculate the spectral envelope of the speech signal. As a consequence, the processing requirements for calculating an extended bandwidth signal are lower than in systems known in the art. Furthermore, the method according to the invention has the advantage that a system working with the above-described method works delayless. Every speech signal is composed of different frequency components. Each speech signal has a fundamental frequency and the harmonics being an integer multiple of the fundamental frequency. In telecommunication systems the fundamental frequency and the first harmonics may be attenuated and filtered out by the transmission system of the telecommunication system. Accordingly, the speech system comprises most of the time only the harmonics, but not the fundamental frequency which were filtered out by the bandpass filter. In the case of such a speech signal comprising the harmonics of a fundamental frequency the lower frequency components, i.e. the harmonics, eventually also the first harmonics, can be generated by applying a non-linear function to the bandwidth limited speech signal.

[0009] According to the invention, the non-linear function is a quadratic function of the following form:

$$x_{nl}(n) = c_2(n)x^2(n) + c_1(n)x(n) + c_0(n) . \tag{1}$$

[0010] The coefficients c_0, c_1 and c_2 depend on time n. The present non-linear function, i.e. the present quadratic function, is used to generate signal components which are not contained in the bandwidth limited speech signal. The advantage of this quadratic function is that for speech signals which are an integer multiple of a fundamental frequency, larger harmonics and the fundamental frequency components are generated. A drawback of these non-linear functions is that the dynamic of the speech signal is changed.

Normally, the dynamic increases with the power of the used function. This is why in the present case the power of the function is limited to 2, meaning that a quadratic function is used.

[0011] According to the invention, the maximum $x_{max}(n)$ of the absolute value of the bandwidth limited speech signal is determined. This maximum of the bandwidth limited speech signal can be determined for each value of the sample digital speech signal, wherein the maximum at time n-1 may be used in order to adjust the maximum at time n. This maximum can be used for determining the coefficients c_0, c_1 and c_2 of the non-linear function. According to the invention, the coefficients are determined in such a way that

$$c_0(n) = -x_{mit}(n-1),$$ (2)

$$c_1(n) = K_{nl,1} - c_2(n)x_{max}(n),$$ (3)

$$c_2(n) = \frac{K_{nl,2}}{g_{max}x_{max}(n) + \varepsilon}.$$ (4)

[0012] The determination of x_{max} helps to limit the change in dynamic when a quadratic function is used which is applied to the bandwidth limited speech signal. In the coefficients the following values for the different constants have been used. According to a preferred embodiment, the constant $K_{nl,1}$ lies in a range between 0.5 and 1.5, $K_{nl,1}$ preferably being 1.2. $K_{nl,2}$ is in the range between 0.1 and 2 and is preferably 1. The constant g_{max} is preferably between 1 and 3 and is preferably 2. The constant ε has been used in order to avoid a division by 0. For ε a very small value such as 10^{-5} may be used.

[0013] According to another embodiment of the invention, the method comprises further the step of removing the constant component after applying the non-linear function to the bandwidth limited speech signal. When the quadratic function is multiplied to the speech signal, a constant component is generated. The coefficient $c_0(n)$ is used for removing this constant component. In the equation for determining c_0 the value $x_{mit}(n)$ is used. This value is calculated using a first order recursion with the following equation:

$$x_{mit}(n) = \beta_{mit}x_{mit}(n-1) + (1 - \beta_{mit})x_{nl}(n).$$ (5)

[0014] The time constant β_{mit} should be chosen from the range $0.95 < \beta_{mit} < 0.9995$.

[0015] When the non-linear function is applied to the bandwidth limited speech signal, the latter comprises signal components which are either already comprised in the bandwidth limited speech signal itself, or low signal components in the range between about 0 Hz to 50 Hz or 100 Hz, which do not comprise voice signal components. According to a preferred embodiment, the signal after applying the non-linear function is high-pass filtered for attenuating low frequency signal components that are lower than a predetermined value. This value may be chosen between 50 Hz and 100Hz and may depend on the fact whether the speech signal is a signal of a male or a female. This high-pass filter can be a first order Butterworth filter (an infinite impulse response filter). The output signal $\tilde{x}_{nl}(n)$ of this high-pass filter follows the following equation:

$$\tilde{x}_{nl}(n) = a_{hp}\left(x_{nl}(n-1) - x_{nl}(n)\right) + b_{hp}\tilde{x}_{nl}(n-1).$$ (6)

[0016] For the filter coefficients a_{hp} and b_{hp} the following values have proven appropriate values: $a_{hp} = 0.99$ and $b_{hp} = 0.95$. It should be understood that these filter coefficients may be chosen from a range nearby the above-described values.

[0017] The extended signal further comprises the components which are already contained in the original bandwidth limited speech signal. In order to remove these signal components the signal is low-pass filtered in such a way that the signal components comprised in the bandwidth limited speech signal are filtered out. After these two filter steps a speech signal remains having low frequency components which were attenuated in the bandwidth limited speech signal. By

way of example, the resulting filtered signal may have signal components in the range between about 50 Hz or 100 Hz to 300 Hz.

[0018] Last but not least, this low frequency speech signal is added to the bandwidth limited speech signal resulting in an improved bandwidth extended speech signal. Due to the fact that the extended speech signal also has lower frequency components, the quality of the speech signal can be improved. According to another embodiment of the invention, a lower end of the bandwidth of the frequency spectrum of the bandwidth limited speech signal may be determined, and if a predetermined frequency spectrum is not contained in the bandwidth limited speech signal, the lower frequency components are generated as described above and added to the bandwidth limited signal. When the lower end of the bandwidth of the bandwidth limited speech signal is known, the lowpass filter for filtering out the higher frequencies in the signal which were generated by the application of the non-linear function, can be adapted accordingly.

[0019] According to another embodiment of the invention, the mean fundamental frequency of the bandwidth limited speech signal can be determined. Signal components below said mean fundamental frequency do not comprise voice components, but noise. When the mean fundamental frequency of the speech signal is known, the high-pass filtering can be adapted to said mean fundamental frequency.

[0020] According to a preferred embodiment of the invention, the bandwidth limited speech signal is a speech signal which was transmitted via a telecommunication network, where the low signal components of the speech signal were filtered out. However, it is also possible that the speech signal was transmitted via any other transmission system in which the bandwidth of the speech signal is limited due to the transmission of the signal.

[0021] The invention further relates to a system for extending the spectral bandwidth as described above, the system comprising a determination unit for determining the maximum signal intensity of the bandwidth limited speech signal, a processing unit in which a non-linear function is applied to the bandwidth limited speech signal for generating the lower frequency components of the speech signal not contained in the bandwidth limited speech signal. Additionally, a high-pass filter may be provided for high-pass filtering the signal after applying the non-linear function to the speech signal. Additionally, a low-pass filter is provided for filtering the signal after applying the non-linear function to the bandwidth limited speech signal and preferably after applying the high-pass filter. Furthermore, an adder may be provided in the system which adds the original bandwidth limited speech signal to the high- and low-pass filtered signal, so that a bandwidth extended improved speech signal is obtained.

[0022] In order to know whether the speech signal should be extended a bandwidth determination unit is provided which determines the bandwidth of the speech signal and which then determines whether it is necessary to add frequency components or not.

[0023] Additionally, a fundamental frequency determination unit may be provided which determines the mean fundamental frequency of the speech signal. With this knowledge of the mean fundamental frequency the high-pass filter may be adapted accordingly. The signal component below the fundamental frequency may be filtered out.

[0024] These and other aspects of the invention will become apparent from the embodiments described hereinafter.

[0025] In the drawings

Fig. 1 shows a telecommunication system in which the bandwidth extension of the invention can be used,

Fig. 2 shows the spectra of a signal before and after a transmission over a telecommunication network,

Fig. 3 shows a system for extending the bandwidth of a speech signal,

Fig. 4 shows a flowchart comprising the different steps for carrying out the bandwidth extension,

Figs. 5a-5c show frequency analyses of a speech signal, of the speech signal after transmission, and of the extended speech signal, and

Fig. 6 shows another embodiment of a system for extending the bandwidth of a speech signal.

[0026] In Fig. 1 a telecommunication system in which the bandwidth extension according to the invention may be used is shown. A first subscriber 10 of the telecommunication system communicates with a second subscriber 11 of the telecommunication system. The speech signal from the first subscriber is transmitted via a network 15. The dashed lines indicate the locations where the transmitted speech signal undergoes the bandwidth limitations which take place depending on the routing of the call. The degradation of the speech quality using analogue telephone systems is caused by the band limiting filters within amplifiers, these filters normally having a bandwidth from around 300Hz to about 3400 Hz. One possibility to increase the speech quality for the subscriber 11 receiving the speech signal is to increase the bandwidth after the transmission by means of a bandwidth extension unit 16. The signal output from the telecommunication system is x(n). In the bandwidth extension unit 16 the bandwidth is extended before the extended speech signal

y(n) is then transmitted to the subscriber 11. In the present example the lower spectral components of the speech signal from around 50 Hz to 300 Hz are generated. In extended sound signals the sound is more natural and, as a variety of listenings indicates, the speech quality in general is increased.

[0027]   In Fig. 2 the spectra of a signal are shown before and after the transmission via a GSM network. In the present case a cellular phone was used receiving the signal. In Fig. 2, graph 21, shows the spectrum of the signal as it is emitted from the subscriber 10. Additionally, the spectrum 22 is shown as measured before the signal enters the bandwidth extension unit 16. As can be seen from the output signal of the communication system 22 the lower frequency components are highly attenuated. At 300 Hz the attenuation is already 10 dB.

[0028]   In Fig. 3 a system is shown which can be used for extending the bandwidth of the bandwidth limited signal 22 in the lower frequency range. The bandwidth limited speech signal x(n) received via the telecommunication system is first of all input to a maximum determination unit, where the short time maximum x_{max} depending on time n is estimated. This maximum is estimated by using a multiplicative correction of a former estimated maximum value. The maximum is determined by the following equation:

$$x_{\max}(n) = \begin{cases} \max\{K_{\max}|x(n)|, \Delta_{ink} x_{\max}(n-1)\} & \text{if } |x(n)| > x_{\max}(n-1), \quad (7) \\ \Delta_{dek} x_{\max}(n-1), & \text{else} \end{cases}$$

[0029]   For this estimation the two decrement and increment constants Δ_{dek} and Δ_{ink} are used. In this recursive formula the two constants Δ_{dek} and Δ_{ink} should meet the following condition:

$$0 < \Delta_{dek} < 1 < \Delta_{ink}. \quad (8)$$

[0030]   Additionally, the constant K_{max} is used which should be chosen from the interval

$$0.25 < K_{\max} < 4. \quad (9)$$

[0031]   The constant K_{max} is used for limiting the estimated maximum by the lower threshold K_{max}. With this formula it is determined how close the maximum value is to the actual maximum value of the speech signal. If K_{max} is at the lower threshold 0.25, this means that the minimum estimated value is at least a quarter of the actual value. The highest threshold 4 means that the estimated maximum value can become four times larger the real maximum value. The two constants Δ_{dek} and Δ_{ink} may be chosen from the interval of $1.001 < \Delta_{ink} < 2$, the constant Δ_{dek} may be chosen from the interval $0.5 < \Delta_{dek} < 0.999$. Tests have shown that the following values of K_{max} and Δ_{dek} and Δ_{ink} can be used:

$$K_{\max} = 0.8,$$

$$\Delta_{ink} = 1.05,$$

$$\Delta_{dek} = 0.995.$$

[0032]   The bandwidth limited speech signal is also fed to a processing unit 32 in which a non-linear function is applied to the bandwidth limited speech signal. As explained in the introductory part of the description, a bandwidth extension can be obtained when a speech signal containing harmonics of a fundamental frequency is multiplied with a non-linear function. In the present context the following quadratic function (1) is used:

$$x_{nl}(n) = c_2(n)x^2(n) + c_1(n)x(n) + c_0(n)$$

[0033] In speech signals the fundamental frequency depends on the person emitting the speech signal. A male voice signal can have a fundamental frequency between 50 Hz to 100 Hz, whereas the fundamental frequency of a female voice or a voice of a child can have a fundamental frequency of about 150 Hz and 200 Hz. As can be seen in Fig. 2, these fundamental frequencies are highly attenuated or even suppressed in the bandwidth limited speech signal. Also the first and eventually the second harmonic can still be highly attenuated. In the above quadratic equation the coefficients c_0, c_1 and c_2 are time-variable coefficients. These time variable coefficients are used for the following reasons:

[0034] When a quadratic function is applied on/to a signal, the signal dynamic changes considerably. In order to limit this dynamic change, time-variable coefficients are used. This means that the coefficients are adapted to the current input signal which is present at the input of the processing unit. The coefficients are calculated by the equations (2), (3), and (4) mentioned above, whereas the short time maximum $x_{max}(n)$ calculated above is used:

$$c_0(n) = -x_{mit}(n-1),$$

$$c_1(n) = K_{nl,1} - c_2(n)x_{max}(n),$$

$$c_2(n) = \frac{K_{nl,2}}{g_{max}x_{max}(n) + \varepsilon}.$$

[0035] As can be seen from the above equation, the coefficient c_2 of the quadratic term of the function has the maximum value x_{max} in the denominator in order to limit the dynamic of the signal. The other constants used for calculating the coefficients can be selected from the following ranges:

$$0.5 \le k_{nl,1} \le 1.5,$$

$$0.1 \le k_{nl,2} \le 2,$$

$$1 \le g_{max} \le 3,$$

$$10^{-4} < \varepsilon < 10^{-6}.$$

[0036] Preferably, the following values can be used:

$$K_{nl,1} = 1.2,$$

$$K_{nl,2} = 1,$$

$$g_{max} = 2,$$

$$\varepsilon = 10^{-5}.$$

[0037]   The coefficient co(n) is used for eliminating the constant component resulting from the multiplication. For the calculation of c_0, the value $x_{mit}(n)$ is used which is calculated by a first order recursion formula (5) mentioned above:

$$x_{mit}(n) = \beta_{mit} x_{mit}(n-1) + (1 - \beta_{mit}) x_{nl}(n).$$

[0038]   The time constant β_{mit} should be selected from the range

$$0.95 < \beta_{mit} < 0.99995.$$

[0039]   The resulting signal output of the processing unit 32 is the signal $x_{nl}(n)$. This extended speech signal has low frequency components in the range up to 300 Hz, but also comprises signal components of the bandwidth limited speech signal x(n) in the range between 300 Hz to 3400 Hz. In the following, unwanted signal components have to be removed. As explained above, the signal components below the fundamental speech frequency, e.g. below 100 Hz, are signal components which are not part of a voice signal. By way of example, if the first subscriber 10 is using a mobile phone in a vehicle, the surround sound of the vehicle may have low components below the fundamental speech frequency. These low signal components can be removed in a high-pass filter 33 shown in Fig. 3. In a preferred embodiment, the high-pass filter may be a first order Butterworth filter. The output signal of this Butterworth filter $\tilde{x}_{nl}(n)$ is calculated by the following equation:

$$\tilde{x}_{nl}(n) = a_{hp}\left(x_{nl}(n-1) - x_{nl}(n)\right) + b_{hp}\tilde{x}_{nl}(n-1). \tag{10}$$

[0040]   The following values of the filter coefficients a_{hp} and b_{hp} were found to be suitable:

$$a_{hp} = 0.99$$

$$b_{hp} = 0.95$$

[0041]   After having removed the low signal components in the high-pass filter 33, the signal components comprised in the original bandwidth limited speech signal x(n) are still present in signal $\tilde{x}_{nl}(n)$. These signal components transmitted by the telecommunication system and all higher signal components can be filtered out by using a low-pass filter 34. The output signal $e_{nl}(n)$ can be written by the following equation:

$$e_{nl}(n) = \sum_{i=0}^{N_{tp,ma}} a_{tp,i}\tilde{x}_{nl}(n-i) + \sum_{i=1}^{N_{tp,ar}} b_{tp,i} e_{nl}(n-i). \tag{11}$$

[0042]   In this context, Tschebyscheff low-pass filters of the order Ntp,ma = $N_{tp,ar}$ = 4 to 7 have proven suitable. After filtering out desired signal components in the low-pass filter 34, the output signal $e_{nl}(n)$ comprises the low frequency components of the speech signal which were filtered out in the telecommunication system (e.g. the signal components between 50 Hz or 100 Hz to about 300 Hz). These low signal components are added to the bandwidth limited speech

signal x(n) in an adder 35 resulting in the bandwidth extended speech signal y(n). Additionally, a weighing factor g_{nl} can be used to either attenuate or amplify the low signal components, as can be seen by the following equation:

$$y(n) = x(n) + g_{nl}e_{nl}(n).$$

(12)

[0043] The factor g_{nl} can be chosen as being 1, so that no amplification or attenuation of the lower frequency components relative to the bandwidth limited speech signal is obtained. Depending on the different embodiments, the factor g_{nl} may lie in a range between 0.001 to 4.

[0044] In Fig. 5 an analysis of the frequency over time of the speech signal is shown. In Fig. 5a the signal components of the speech signal as emitted by the first subscriber is shown. The signal was directly recorded near the mouth of the user. If this signal shown in Fig. 5a is transmitted via the telecommunication network to another cellular telephone, the received decoded signal has the frequency components shown in Fig. 5b. The missing low signal components below 300 Hz are clearly shown. After processing the signal shown in Fig. 5b as explained in connection with Fig. 3 the signal can be obtained as shown in Fig. 5c. As can be seen from Fig. 5c, the lower signal components could be reconstructed. Even if the Figs. 5a and 5c do not completely match the signal quality of the signal shown in Fig. 5c has improved over the signal quality of the signal shown in Fig. 5b.

[0045] In Fig. 4 the different steps are summarized which are needed to extend the bandwidth of the bandwidth limited speech signal. After the start of the method at step 41 the maximum $x_{max}(n)$ of the speech signal is determined in the determination unit 31 (step 42). With the maximum value $x_{max}(n)$ the non-linear function of equation (1) can be determined in step 43. This non-linear function is then applied to the bandwidth limited speech signal in the processing unit 32 (step 44). The resulting signal $x_{nl}(n)$ is then high-pass filtered in high-pass filters 33 in order to remove noise components below the fundamental speech frequency (step 45). In the next step 46 the signal $\tilde{x}_{nl}(n)$ is low-pass filtered to remove the signal components already comprised in the bandwidth limited speech signal itself. Last but not least the filter signal $e_{nl}(n)$ is then added to the original bandwidth limited speech signal in step 47, resulting in an improved speech signal y(n) in which the low frequency components, the fundamental frequency and eventually the first harmonics, are contained. The bandwidth extension ends in step 48.

[0046] In Fig. 6 a further embodiment of a system for a bandwidth extension is shown. The system of Fig. 6 comprises the same components as the system shown in Fig. 3, the components having the same reference numeral working the same way as described in connection with Fig. 3. Accordingly, a detailed description of these components is omitted.

[0047] The attenuation of the speech signal can depend on the used microphone to record the signal, or on the way the signal is coded or on the signal processing in the telephone of the first subscriber or the telecommunication network, respectively. As a result, a large attenuation of the speech signal over a broad range of frequencies can occur. In other cases the attenuation of the signal can be less significant, or the signal is not attenuated in the low frequency range at all. If the low frequencies are attenuated, these low frequencies should be generated and added to the signal. If, however, the low frequencies are present in the signal, no signal components should be added to the signal. In order to be able to react on the different attenuation situations, it might be helpful to detect the frequencies present in the speech signal. This can be done in a bandwidth determination unit 61 in which the frequency components of the signals are analyzed, so that it can be determined which frequency components have been transmitted and which frequency components have been attenuated. Depending on the estimated frequency components of the speech signal x(n) the low-pass filter 34 can be controlled in accordance with the determined spectrum. To this end, a calculation unit 62 may be provided in which the filter coefficients $a_{tp,i}$ and $b_{tp,i}$ are calculated and adapted to the bandwidth of the speech signal in such a way that components which are already comprised in the signal x(n) itself are filtered out in the low-pass filter 34. The adapted filter coefficients are then supplied to the low-pass filter. If the signal comprises all signal components, the system is controlled in such a way that no low-pass filtering is carried out.

[0048] In the following, another adaptation of the system shown in Fig. 3 is described. As already mentioned above, the signal components below the fundamental frequency do not comprise speech components and should be suppressed, which is done by the high-pass filter 33. However, the fundamental frequency is not a constant value and may depend on the fact whether a male or female or a child voice is transmitted via the telecommunication system. This fundamental frequency can change between 50 Hz and 200 Hz. Accordingly, the high-pass filter 33 can be adapted to the fundamental frequency. This can be achieved by a fundamental frequency determination unit 63, in which the mean fundamental frequency of the speech signal is determined. If the determined fundamental frequency is very low (e.g. 50 Hz), the high-pass filtering may be omitted, or the high-pass filter may be adapted in such a way that only signals below 50 Hz are filtered out. In the case of the fundamental frequency being around 200 Hz the high-pass filter 33 should be adapted accordingly and should filter out the frequencies below the determined fundamental frequency. When the mean fundamental frequency is determined in unit 63, the filter coefficients for the high-pass filter can be adapted accordingly in the filter coefficient calculation unit 64, which are then fed to the high-pass filter 33.

[0049]   It should be understood that the bandwidth determination unit 61 and the corresponding filter coefficient cal-culation unit 62 can be used independently from the fundamental frequency determination unit 63. This means that either of the two units 61 and 63 or both units 61 and 63 may be used.

[0050]   Summarizing, the invention provides a method and a system for extending the lower frequency parts of a telephone band limited speech signal and can thus increase the speech quality. The advantage over other sophisticated methods is the very low computational complexity and the delaylessness of the described method. These advantages open up a broad range of possible applications. It is not necessary to calculate the envelope of the speech signal. Accordingly, the system does not generate a delay in the speech signal. In addition, the described method can be used in connection with many different frequency characteristics of the recorded speech signal and of the hardware used for the recording, or of the hardware used for the signal transmission, such as ISDN, GSM or CDMA. In addition, the system can easily handle noise components from the environment of the speaking person, e.g. when the signal is to be transmitted from a vehicle environment.

Claims

1. Method for extending the spectral bandwidth of a bandwidth limited speech signal which comprises at least harmonics of a fundamental frequency, wherein a nonlinear function is applied to the bandwidth limited speech signal for generating the lower frequency components of the speech signal which are attenuated in the bandwidth limited speech signal.

2. Method according to claim 1, **characterized in that** the nonlinear function is the following quadratic function:

$$x_{nl}(n) = c_2(n)x^2(n) + c_1(n)x(n) + c_0(n)$$

the coefficients c0, c1 c2 depending on time n, wherein the application of the nonlinear function to the bandwidth limited speech signal results in a first extended speech signal.

3. Method according to claim 1 or 2, **characterized by** further comprising the step of determining the maximum x_{max} (n) of the bandwidth limited speech signal.

4. Method according to claim 3, **characterized in that** the coefficients are determined in such a way that

$$c_0(n) = -x_{mit}(n-1),$$

$$c_1(n) = K_{nl,1} - c_2(n)x_{max}(n),$$

$$c_2(n) = \frac{K_{nl,2}}{g_{max}x_{max}(n) + \varepsilon}.$$

5. Method according to any of the preceding claims, **characterized by** further comprising the step of removing the constant component after applying the nonlinear function to the bandwidth limited speech signal

6. Method according to any of the preceding claims, **characterized by** further comprising the step of high-pass filtering the signal after applying the nonlinear function to the bandwidth limited speech signal, for attenuating low frequency signal components that are lower than a predetermined value.

7. Method according to any of the preceding claims, **characterized by** further comprising the step of low-pass filtering the signal after applying the nonlinear function to the bandwidth limited speech signal, where the signal components

comprised in the bandwidth limited speech signal are filtered out, resulting in a low frequency speech signal having frequency components which were attenuated in the bandwidth limited speech signal.

8. Method according to claim 7, **characterized by** further comprising the step of adding the low frequency speech signal to the bandwidth limited speech signal resulting in an improved bandwidth extended speech signal.

9. Method according to any of the preceding claims, **characterized by** further comprising the step of determining the lower end of the bandwidth of the frequency spectrum of the bandwidth limited speech signal and if a predetermined frequency spectrum is not contained in the bandwidth limited speech signal the lower frequency components are generated and added to the bandwidth limited speech signal.

10. Method according to claim 9, wherein the low-pass filter for filtering out the frequency components already comprised in the bandwidth limited speech signal is adjusted in accordance with the determined bandwidth of the speech signal.

11. Method according to any of the preceding claims, **characterized by** further comprising the step of determining the mean fundamental frequency of the bandwidth limited speech signal, wherein the high-pass filtering is adapted to said mean fundamental frequency.

12. Method according to any of the preceding claims, wherein the bandwidth limited speech signal is a speech signal transmitted via a telecommunication network which filters out the low signal components of the speech signal.

13. System for extending the spectral bandwidth of a bandwidth limited speech signal, comprising:

    - a determination unit (31) for determining the maximum signal intensity of the bandwidth limited speech signal,
    - a processing unit (32) in which a nonlinear function is applied to the bandwidth limited speech signal for generating the lower frequency components of the speech signal which are lower than a predetermined signal component,
    - a high-pass filter (33) for high-pass filtering the signal after applying the nonlinear function to the bandwidth limited speech signal,
    - a low-pass filter (34) filtering the signal after applying the nonlinear function to the bandwidth limited speech signal,
    - an adder (35) in which the high and low-pass filtered signal is added to the original bandwidth limited speech signal.

14. System according to claim 13, further comprising a bandwidth determination unit (61) determining the bandwidth of the bandwidth limited speech signal.

15. System according to claim 13 or 14, further comprising a fundamental frequency determination unit (63) determining the mean fundamental frequency of the bandwidth limited speech signal.

Amended claims in accordance with Rule 86(2) EPC.

1. Method for extending the spectral bandwidth of a bandwidth limited speech signal which comprises at least harmonics of a fundamental frequency, wherein a nonlinear function is applied to the bandwidth limited speech signal for generating the lower frequency components of the speech signal which are attenuated in the bandwidth limited speech signal, **characterized in that** the nonlinear function is the following quadratic function:

$$x_{nl}(n) = c_2(n)x^2(n) + c_1(n)x(n) + c_0(n)$$

the coefficients c0, c1 c2 depending on time n, wherein the application of the nonlinear function to the bandwidth limited speech signal results in a first extended speech signal,
the coefficients being determined in such a way that

$$c_0(n) = -x_{mit}(n-1),$$

$$c_1(n) = K_{nl,1} - c_2(n)x_{max}(n),$$

$$c_2(n) = \frac{K_{nl,2}}{g_{max}x_{max}(n) + \varepsilon},$$

$K_{nl,1}$, $K_{nl,2}$, g_{max}, E being predetermined constants,
$X_{max}(n)$ being the short time maximum of the absolute value of the bandwidth limited speech signal,
$X_{mit}(n)$ being the short time mean value of the output of the nonlinear function.

2. Method according to claim 1, **characterized by** further comprising the step of removing the constant component after applying the nonlinear function to the bandwidth limited speech signal.

3. Method according to any of the preceding claims, **characterized by** further comprising the step of high-pass filtering the signal after applying the nonlinear function to the bandwidth limited speech signal, for attenuating low frequency signal components that are lower than a predetermined value.

4. Method according to any of the preceding claims, **characterized by** further comprising the step of low-pass filtering the signal after applying the nonlinear function to the bandwidth limited speech signal, where the signal components comprised in the bandwidth limited speech signal are filtered out, resulting in a low frequency speech signal having frequency components which were attenuated in the bandwidth limited speech signal.

5. Method according to claim 4, **characterized by** further comprising the step of adding the low frequency speech signal to the bandwidth limited speech signal resulting in an improved bandwidth extended speech signal.

6. Method according to any of the preceding claims, **characterized by** further comprising the step of determining the lower end of the bandwidth of the frequency spectrum of the bandwidth limited speech signal and if a predetermined frequency spectrum is not contained in the bandwidth limited speech signal the lower frequency components are generated and added to the bandwidth limited speech signal.

7. Method according to claim 6, wherein the low-pass filter for filtering out the frequency components already comprised in the bandwidth limited speech signal is adjusted in accordance with the determined bandwidth of the speech signal.

8. Method according to any of the preceding claims, **characterized by** further comprising the step of determining the mean fundamental frequency of the bandwidth limited speech signal, wherein the high-pass filtering is adapted to said mean fundamental frequency.

9. Method according to any of the preceding claims, wherein the bandwidth limited speech signal is a speech signal transmitted via a telecommunication network which filters out the low signal components of the speech signal.

10. System for extending the spectral bandwidth of a bandwidth limited speech signal, comprising:

- a determination unit (31) for determining the maximum signal intensity of the bandwidth limited speech signal,
- a processing unit (32) in which a nonlinear function is applied to the bandwidth limited speech signal for generating the lower frequency components of the speech signal which are lower than a predetermined signal component, the nonlinear function being the following quadratic function:

$$x_{nl}(n) = c_2(n)x^2(n) + c_1(n)x(n) + c_0(n)$$

the coefficients c0, c1 c2 depending on time n, wherein the application of the nonlinear function to the bandwidth limited speech signal results in a first extended speech signal,
the coefficients being determined in such a way that

$$c_0(n) = -x_{mit}(n-1),$$

$$c_1(n) = K_{nl,1} - c_2(n)x_{max}(n),$$

$$c_2(n) = \frac{K_{nl,2}}{g_{max}x_{max}(n) + \varepsilon},$$

$K_{nl,1}$, $K_{nl,2}$, g_{max}, ε being predetermined constants,
$X_{max}(n)$ being the short time maximum of the absolute value of the bandwidth limited speech signal,
$X_{mit}(n)$ being the short time mean value of the output of the nonlinear function comprising
- a high-pass filter (33) for high-pass filtering the signal after applying the nonlinear function to the bandwidth limited speech signal,
- a low-pass filter (34) filtering the signal after applying the nonlinear function to the bandwidth limited speech signal,
- an adder (35) in which the high and low-pass filtered signal is added to the original bandwidth limited speech signal.

11. System according to claim 10, further comprising a bandwidth determination unit (61) determining the bandwidth of the bandwidth limited speech signal.

12. System according to claim 10 or 11, further comprising a fundamental frequency determination unit (63) determining the mean fundamental frequency of the bandwidth limited speech signal.
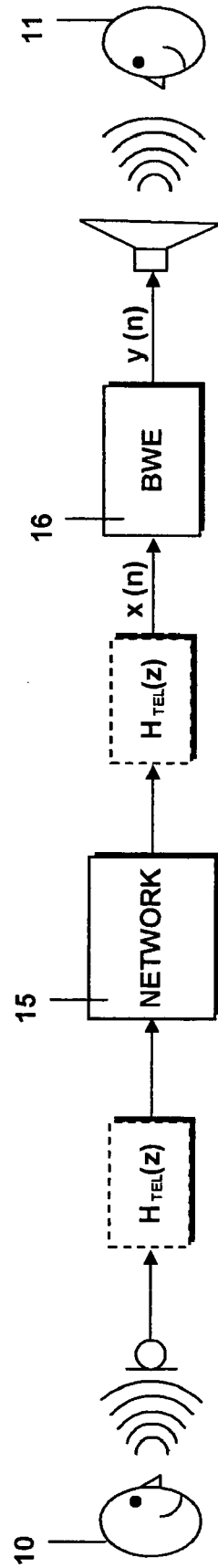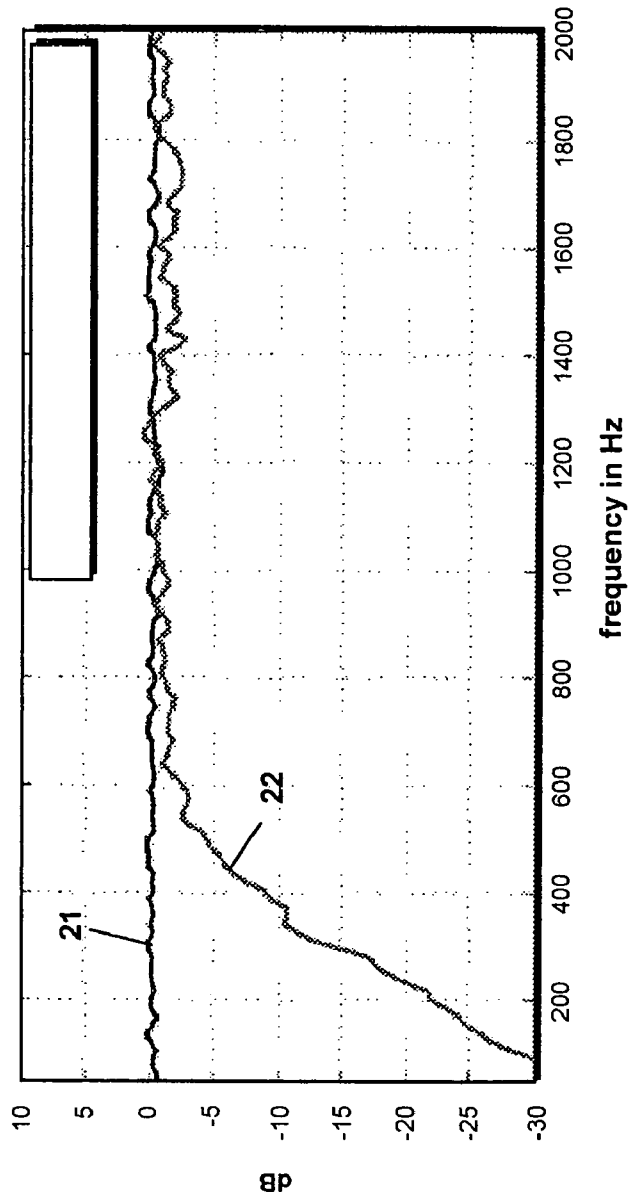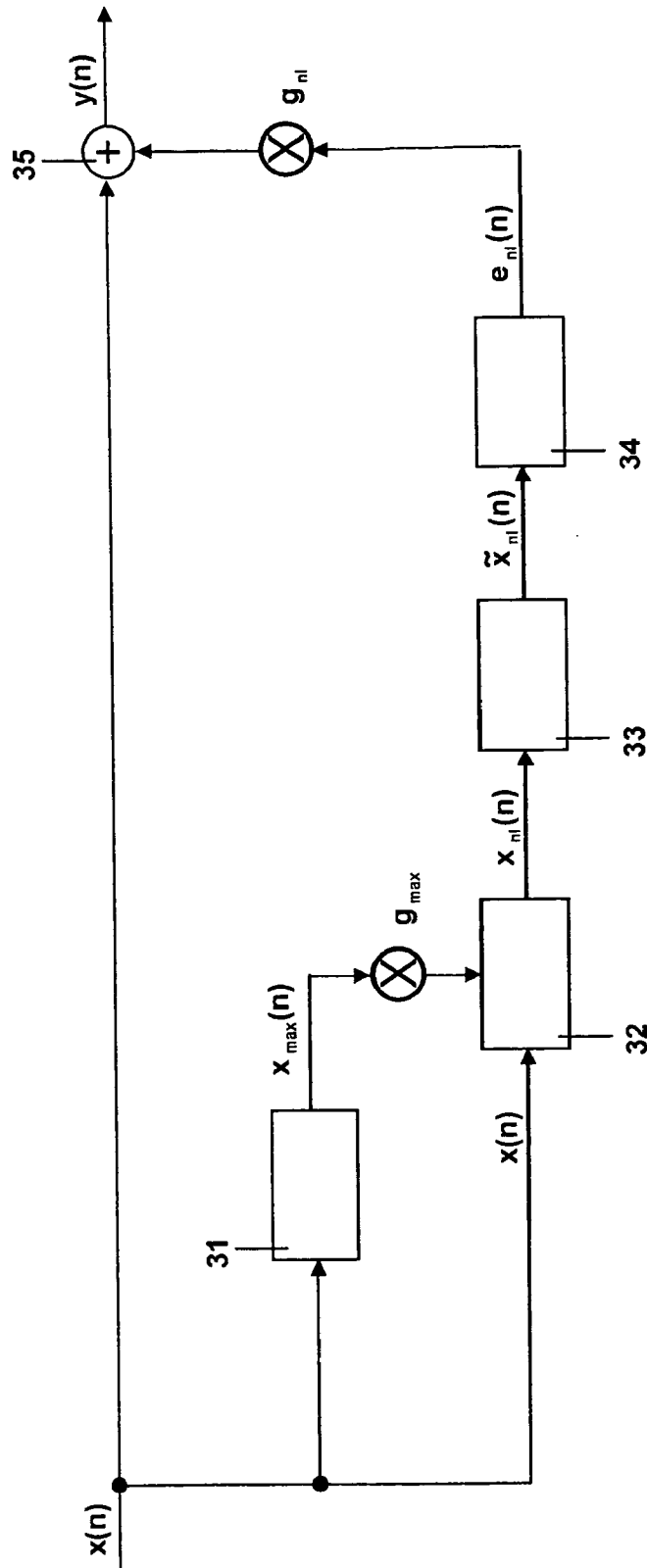
Fig. 1

Fig. 2

Fig. 3

41 — START

42 — DETERMINE MAXIMUM $x_{max}(n)$ OF SPEECH SIGNAL

43 — DETERMINE NONLINEAR FUNCTION

44 — APPLY NONLINEAR FUNCTION TO SPEECH SIGNAL

45 — HIGH PASS FILTERING OF SIGNAL

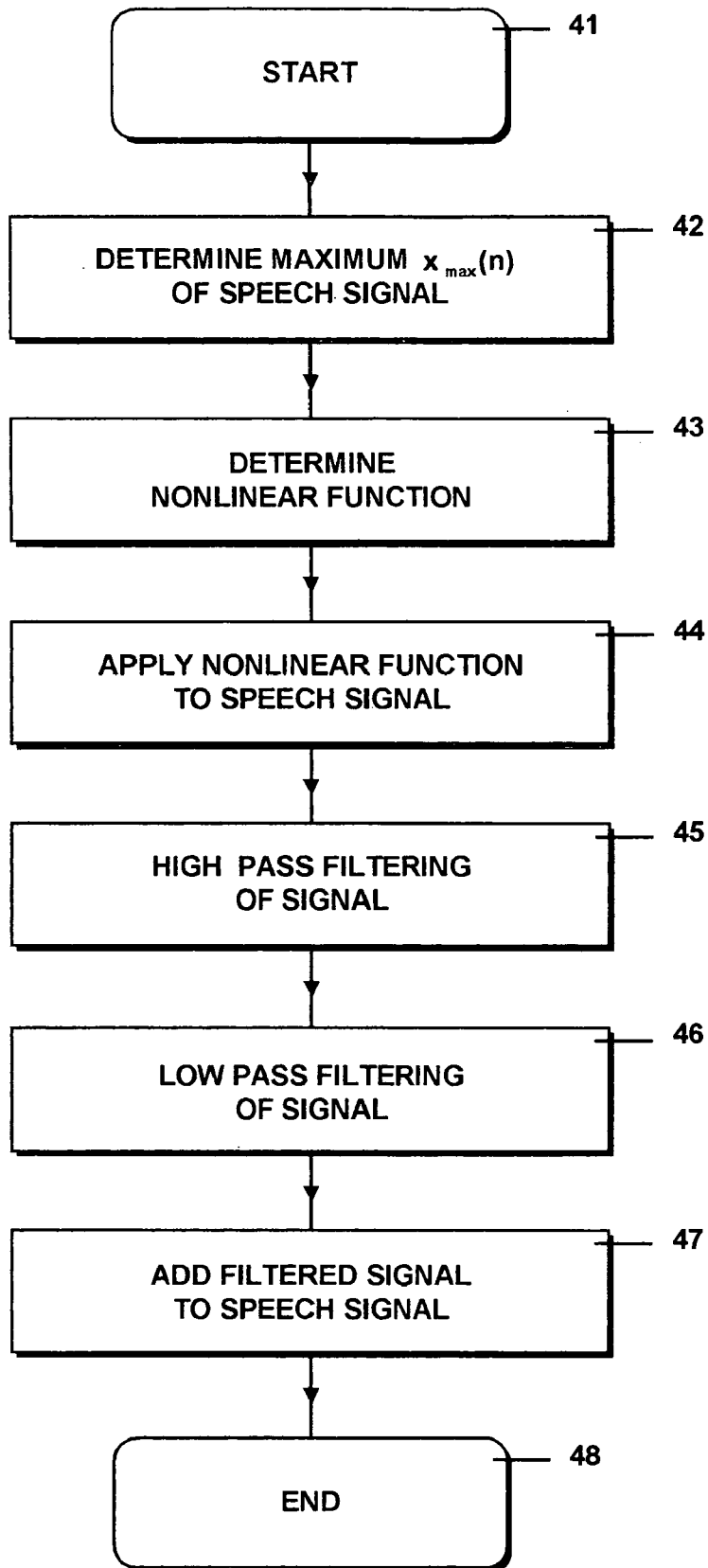46 — LOW PASS FILTERING OF SIGNAL

47 — ADD FILTERED SIGNAL TO SPEECH SIGNAL

48 — END

FIG. 4

Fig. 5 A

Fig. 5 B

Fig. 5 C

Fig. 6

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Application Number

EP 06 00 1984
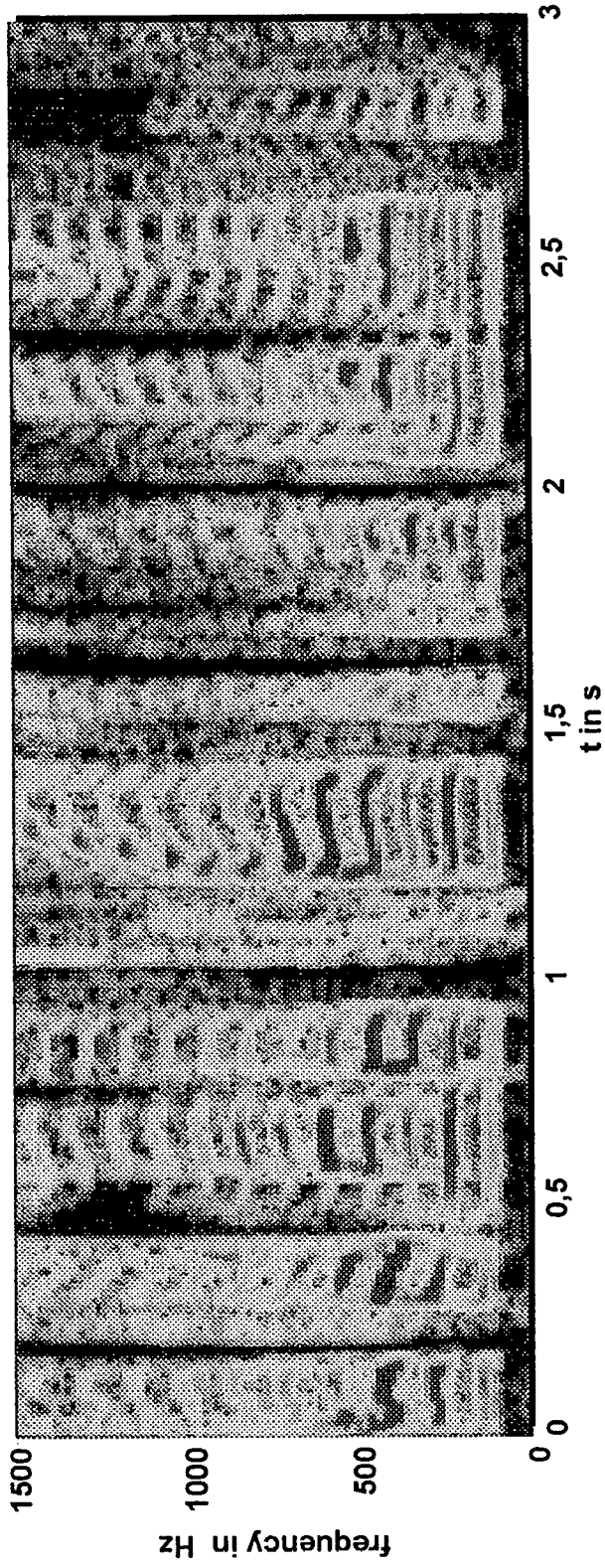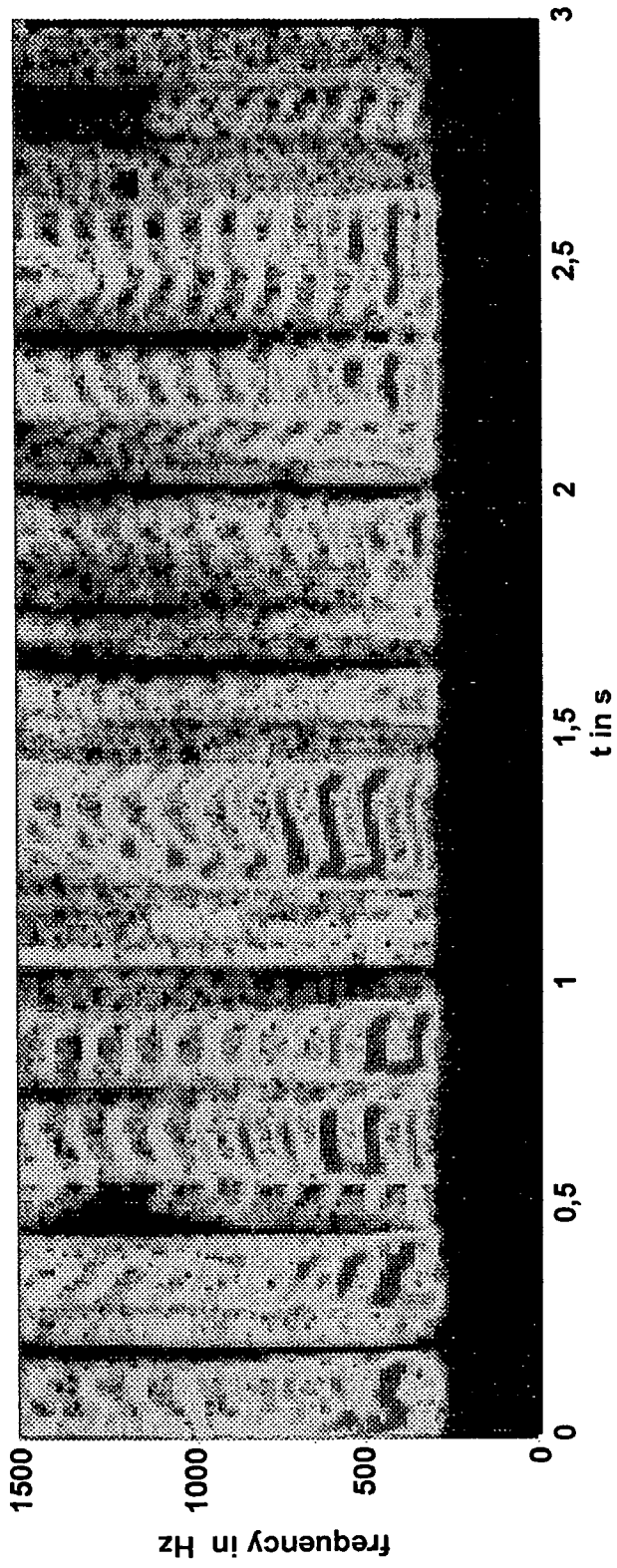
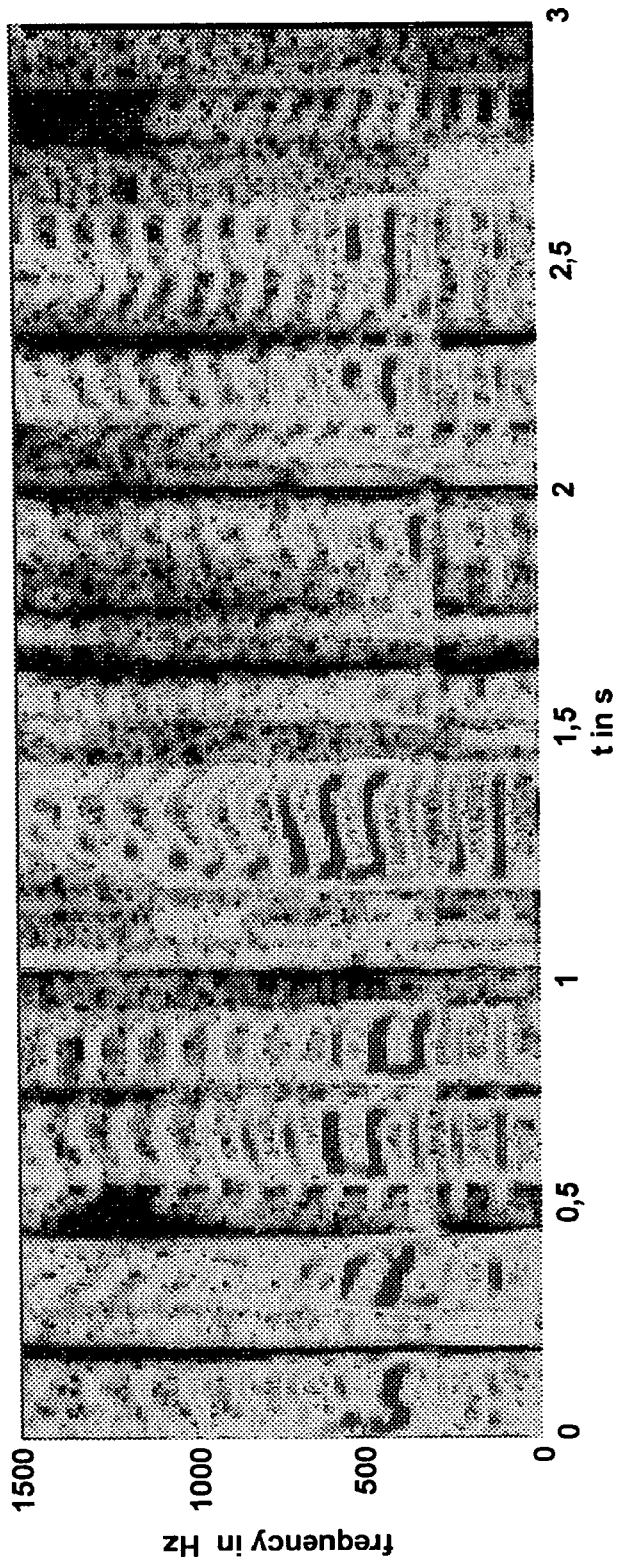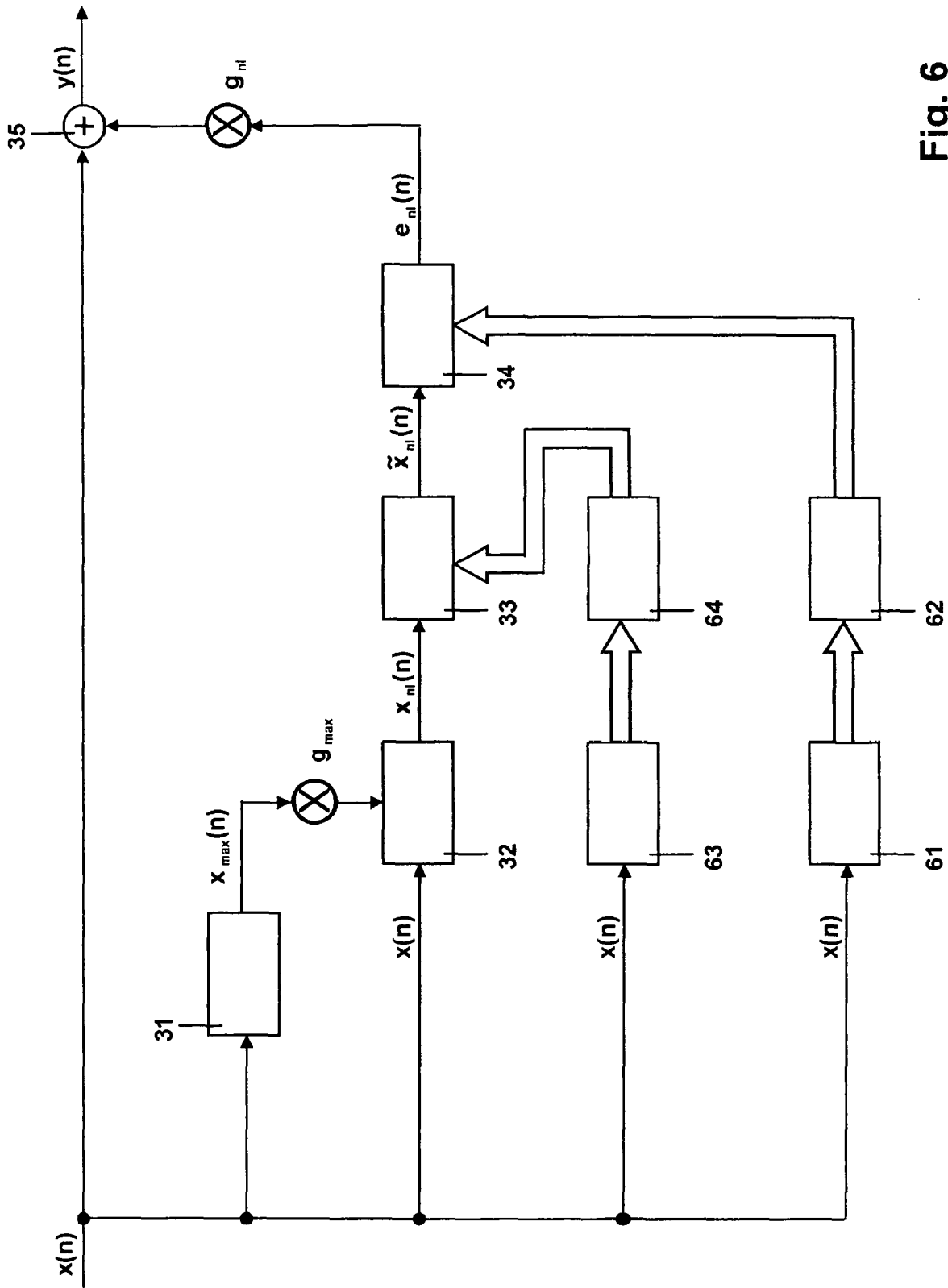## DOCUMENTS CONSIDERED TO BE RELEVANT

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The present search report has been drawn up for all claims

Place of search	Date of completion of the search	Examiner
The Hague	9 May 2006	Burchett, S

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