

(19)



(11)

EP 1 944 761 A1

(12)

EUROPEAN PATENT APPLICATION

(43) Date of publication:

16.07.2008 Bulletin 2008/29

(51) Int Cl.:

G10L 21/02 ^(2006.01)**G10L 19/06** ^(2006.01)**H04M 9/08** ^(2006.01)(21) Application number: **07000716.6**(22) Date of filing: **15.01.2007**

(84) Designated Contracting States:

AT BE BG CH CY CZ DE DK EE ES FI FR GB GR HU IE IS IT LI LT LU LV MC NL PL PT RO SE SI SK TR

Designated Extension States:

AL BA HR MK RS(71) Applicant: **Siemens Networks GmbH & Co. KG****81541 München (DE)**

(72) Inventors:

- **Beaugeant, Christophe**
81735 München (DE)

• **Taddei, Herve****80336 München (DE)**• **Thepie Fapi, Emmanuel Rossignol****81373 München (DE)**(74) Representative: **Maier, Daniel Oliver et al****Siemens AG****CT IP Com E****Postfach 22 16 34****80506 München (DE)**(54) **Disturbance reduction in digital signal processing**

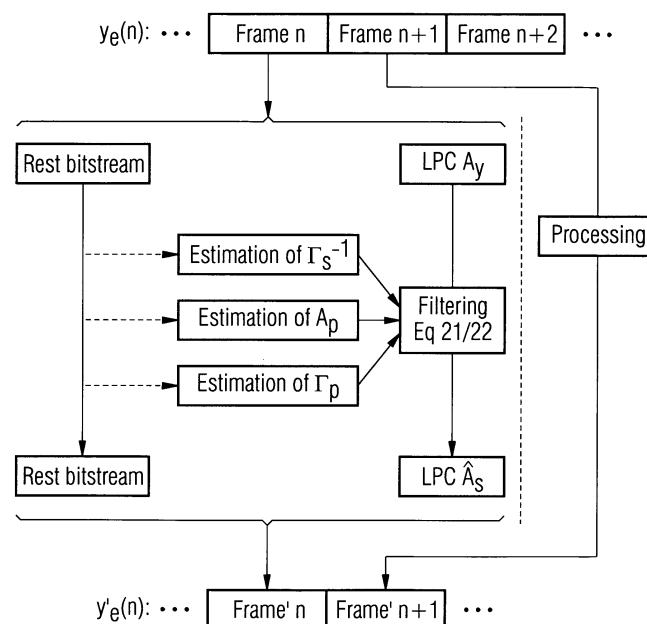
(57) A method is provided for transmitting a digital signal $y(n)$, $y(n)$ comprising a useful signal $s(n)$ and a perturbation signal $p(n)$. The method comprises the steps of:

- receiving the Linear Prediction Coefficients (LPC) A_y of the signal $y_e(n)$, $y_e(n)$ being an LPC-encoded signal of $y(n)$.

- estimating the autocorrelation matrix Γ_s of the useful signal $s(n)$, of the autocorrelation matrix Γ_p of the perturbation signal $p(n)$ and the LPC A_p of the perturbation signal $p(n)$;

- calculating a modified LPC A_s by using A_y and the estimated Γ_s , Γ_p , A_p ;

- outputting a modified data stream $y'_e(n)$ including the modified LPC A_s .

FIG 2**EP 1 944 761 A1**

Description

[0001] The invention relates to disturbance reduction in digital signal processing.

[0002] In telecommunication devices, sound pick-up has to deal with two problems: The presence of noise in the environment where the devices are and the echo phenomenon due to the coupling between the loudspeaker and the microphone. These phenomenon decrease the quality of the communication: On one hand, the speech signal of the user is corrupted by the environmental noise which leads to tiredness for the far-end speaker and in case of too loud noise to misunderstanding between the correspondents. The echo phenomenon is also really perturbing for the far-end speaker as he can hear his own voice due to the coupling. If the transmission delay of the communication network is important (more than 30 ms), he has the really bad feeling of hearing his own voice delayed. Current mobile networks and their delay of more than 150 ms make this echo phenomenon annoying (previously, fixed telephony network provided low delay for local calls and the echo phenomenon was disturbing only for international communications).

[0003] These two drawbacks are inherent to phone communication. In case of handset or headset, their effects on the far-end speaker are a little bit smoothed by the fact that the speech of the user is generally more energetic than the two perturbations (noise and echo). Nevertheless, in certain conditions (in stations, on the street), the surrounding noise can really disturb the far-end speaker. The echo can also appear in small recent devices, or in headsets. In both cases the coupling exists and the echo can be energetic enough to be disturbing.

The hands-free systems present the worse case considering noise and echo problems. Indeed, in this case the microphone is far from the talker so that the speech signal is less energetic and the perturbation gains importance. Moreover, the loudspeaker signal is also louder than in non-hands-free use cases so that the coupling between the two transducers (microphone/loudspeaker) increases.

[0004] Digital telecommunication systems include speech coding. Speech codecs are definitely perturbed by the presence of noise and echo. Indeed, they are optimized to handle single speech signals.

[0005] There exists a lot of codecs, and many of them are using Linear Prediction Coefficients (LPC) analysis. For example, CELP codecs as AMR, EFR, MELP, TCX or vocoders use such an analysis. The generic principle of the LPC analysis is to provide a linear estimation of the input speech signal $y(n)$ through an AR filter as follow:

$$\hat{y}(n) = -\sum_{k=1}^P a_y(k) \cdot y(n-k) \quad (1)$$

[0006] Where $a_y(k)$ stand for the Linear Prediction Coefficients and P stands for the prediction order. After filtering the input signal by the LPC filter, a residual signal is obtained. This signal needs to be transmitted for reconstruction of the original signal to the decoder.

[0007] Noise reduction and echo cancellation are historically built as pre-processing before coding the speech. Many solutions reducing the perturbations on PCM signals are available. A state of the art overview can be found in "Combined Noise and Echo Reduction in Hands-Free systems: A Survey" by R. Le Bouquin Jeannès, P. Scalart, G. Faucon, C. Beaugeant; IEEE Trans. On Speech and Audio Processing; vol.9; Nov 2001; pp 808-820. Such solutions are efficient when the PCM data is available, so typically if the problem are solved within the terminal itself, before the encoding of the signal.

[0008] Within the communication chain, it may happen that only the coded signal is available. This is the case in any part of the network where only the codec bitstream is available. It may also be the case in terminal where an integrated chipset encodes the microphone signal and only delivers a bitstream to the processor where noise reduction or echo cancellation can be implemented.

[0009] In such scenarios, classical solutions can only be applied by decoding the signal, processing it and re-encoding. This leads to a suboptimal solution compared to any solution that would be based on the processing of the PCM signal. Indeed, decoding and re-encoding lead to high computation load, moreover, the so-called tandem effect due to decoding-re-encoding decreases the quality of the speech signal. Accordingly, the enhancement obtained by noise reduction and echo cancellation may be compensated by the artefacts introduced by the tandem effect.

[0010] More recently, the idea appeared to modify the codec parameter "fixed gain" in A-CELP coding to decrease of the signal energy when perturbation is detected. This is shown in

- "Compressed Domain Noise Reduction and Echo Suppression for Network Speech" by Chandran, Ravi and Marchok, Daniel J.; Enhancement, Proc. of the 43rd IEEE Midwest Symposium on Circuits and Systems, pp 10-13, 2000,
- "Noise reduction on speech codec parameters" by Hervé Taddei, Christophe Beaugeant, Michael de Meuleneire; ICASSP 2004
- and "Gain Loss Control based on Speech Codec Parameters" by C. Beaugeant, N. Duetsch, H. Taddei; Eusipco 2004.

[0011] Such techniques allow an efficient decrease of the perturbation energy but can not reduce all the artefacts present on the speech.

[0012] It is an object of the invention to provide a method for reducing noise on encoded digital signals being encoded with Linear Prediction Coefficients (LPC).

[0013] This object is solved by the subject-matter of the independent claims. Further enhancements are provided by the subject-matter of the dependent claims.

[0014] A method is provided for transmission of a digital signal $y(n)$. The digital signal $y(n)$ comprises a useful signal $s(n)$ and a perturbation signal $p(n)$. The perturbation signal $p(n)$ derives e.g. from noise or echo and includes everything of $y(n)$ that is not part of the useful signal $s(n)$.

[0015] The bitstream $y_e(n)$ is derived from $y(n)$ by LPC-encoding. As a first step, the Linear Prediction Coefficients (LPC) \mathbf{A}_y of a bitstream $y_e(n)$ are received. Other parameters of the bitstream $y_e(n)$ may also be received, like the fixed gain or the adaptive gain of the bitstream $y_e(n)$. As an option, the complete bitstream $y_e(n)$ is received. The autocorrelation matrix Γ_s of the useful signal $s(n)$, of the autocorrelation matrix Γ_p of the perturbation signal $p(n)$ and the LPC \mathbf{A}_p of the perturbation signal $p(n)$ are estimated.

[0016] A modified LPC \mathbf{A}_s is calculated. It is calculated from \mathbf{A}_y and the estimated Γ_s , Γ_p , \mathbf{A}_p . In an output step, a modified data stream $y_e'(n)$ including the modified LPC \mathbf{A}_s is output. This data stream can be received by a decoder which decodes the original signal $y(n)$.

[0017] Codecs for transmission of speech are optimized for speech signals. The addition of noise or of echo to the useful speech signal leads to sub-optimal behaviour of the codecs, which means additive artefacts on the decoded signal and lower quality. The use of LPC coefficients that are influenced by the noise signal makes the quality of the received speech worse. Accordingly, noise and echo are not only adding undesired information to the useful signal, they also lead to sub-optimal behaviour of speech codecs, decreasing all the quality of telecommunication.

[0018] Our solution is based on this innovative principle to modify directly the parameter computed by the speech encoders. Compared to prior art solutions, it proposes to modify the LPC coefficients. The received speech sounds more precise if the LPC coefficients are less influenced by noise.

[0019] \mathbf{A}_s is preferably calculated by $\mathbf{A}_s = \Gamma_s^{-1} \cdot [(\Gamma_s + \Gamma_p) \cdot \mathbf{A}_y - \Gamma_p \cdot \mathbf{A}_p]$. This equation only comprises multiplication and addition functions and is accordingly adapted for digital signal processors. \mathbf{A}_s also can be calculated by the equivalent equation $\mathbf{A}_s = \mathbf{A}_y + \Gamma_s^{-1} \cdot \Gamma_p [\mathbf{A}_y - \mathbf{A}_p]$.

[0020] If also the residual signal of the encoded signal $y_e(n)$ is received, the steps of estimating Γ_s , Γ_p , \mathbf{A}_p can be based on the residual signal of $y_e(n)$ and \mathbf{A}_y . The residual signal is the signal that is obtained after the LPC filtering. $y_e(n)$ comprises of the residual signal and the LPC coefficients. The estimations of Γ_s , Γ_p , \mathbf{A}_p can be done by classical methods, e.g. by frequency analysis of the encoded signal $y_e(n)$.

[0021] In an embodiment, the method also comprises a step of a noise reduction on the residual signal of the encoded signal $y_e(n)$. This gives the additional advantage that the noise is reduced in the output data stream $y_e'(n)$. A noise reduction technique on residual signals is described in the above-mentioned "Compressed Domain Noise Reduction and Echo Suppression for Network Speech". The invention described here provides a solution to achieve a reduction of perturbation, like noise and echo, by modifying the LPC coefficients computed during LPC analysis.

[0022] In another embodiment, the Linear Prediction Coefficients (LPC) \mathbf{A}_y of the signal $y_e(n)$ are not received, but calculated from the digital sample signal $y(n)$. The encoding and modifying the LPC coefficients is done only once. Therefore, the residual signal does not need to be encoded and output twice. This improves the speed for encoding and modifying the LPC.

[0023] Even if the embodiments of the invention will be described in respect to speech signals, the invention may be used with any system based on model of Eq (1) where additive perturbation disturbed the coefficients $a_y(k)$. The method is applicable in a broad range of applications in signal processing. One possible application where the LPC modification would be useful is earthquake detection.

[0024] But, the method is especially qualified for signal transmission in telecommunication. Because the different signal characteristics of voice and noise signals, the autocorrelation matrix of the perturbation signal can be estimated relatively precisely. This ensures that the cleaning of the LPC parameters is made successful.

[0025] The invention also relates to a digital signal transmission apparatus that performs the inventive method. Such an apparatus comprises means for receiving the LPC coefficients \mathbf{A}_y , for the estimation of Γ_s , Γ_p , \mathbf{A}_p , for the calculation and output of \mathbf{A}_p .

[0026] Especially, the calculation of \mathbf{A}_s is preferable done by a digital signal processor (DSP) because a DSP is effective in performing multiplications and additions.

[0027] The invention deals with a method to reconstruct the LPC coefficient $\mathbf{A}_s = [a_s(k)]^T$ of the useful signal $s(n)$ knowing the LPC coefficients $\mathbf{A}_y = [a_y(k)]^T$ computed on a perturbed signal $y(n)$. It is based on the following mathematical development of the LPC analysis.

[0028] We consider that a corrupted signal $y(n)$ is the sum of a useful signal $s(n)$ with a perturbation $p(n)$. This perturbation can be additive noise, echo or more generally any signal that is not desired:

$$y(n) = s(n) + p(n) \quad (2)$$

[0029] We assume that the perturbation and the useful signal are not correlated.

[0030] We also assume that an LPC analysis of order P is applied on analysis frames of N samples (this is the case for speech codecs based on LPC), so that the signal $y(n)$ is estimated by $\hat{y}(n)$ as:

$$\hat{y}(n) = -\sum_{k=1}^P a_y(k) \cdot y(n-k) \quad (3)$$

[0031] In the same way, the useful signal $s(n)$ is modeled as:

$$\hat{s}(n) = -\sum_{k=1}^P a_s(k) \cdot s(n-k) \quad (4)$$

and, an estimation $\hat{p}(n)$ of $p(n)$ can be written with:

$$\hat{p}(n) = -\sum_{k=1}^P a_p(k) \cdot p(n-k) \quad (5)$$

[0032] Let's consider the following squared error

$$E_{ST} = \sum_{n=0}^{N-1} \left[y(n) + \sum_{k=1}^P a_y(k) \cdot y(n-k) \right]^2 \quad (6)$$

[0033] Using the fact that the signal $y(n)$ can be written as the sum of the useful signal and of the perturbation, we can write:

$$E_{ST} = \sum_{n=0}^{N-1} \left[s(n) + p(n) + \sum_{k=1}^P a_y(k) \cdot (s(n-k) + p(n-k)) \right]^2 \quad (7)$$

The error E_{ST} is minimum with respect to the LPC coefficients \mathbf{A}_y when its derivative with respect to each coefficient is zero:

$$\forall j, \frac{\partial E_{ST}}{\partial a_y(j)} = 0 \quad (8)$$

[0034] It leads to

$$\sum_{n=0}^{N-1} \left[2 \left(s(n) + p(n) + \sum_{k=1}^P a_y(k) \cdot (s(n-k) + p(n-k)) \right) (s(n-j) + p(n-j)) \right] = 0$$

(9)

[0035] This can be written equivalently

$$\begin{aligned} & \sum_{n=0}^{N-1} [s(n)s(n-j) + p(n)p(n-j) + s(n)p(n-j) + p(n)s(n-j)] + \\ & \sum_{n=0}^{N-1} \sum_{k=1}^P [a_y(k) \cdot [s(n-k)s(n-j) + p(n-k)p(n-j) + s(n-k)p(n-j) + p(n-k)s(n-j)]] = 0 \end{aligned}$$

(10)

[0036] With the hypothesis that the perturbation signal and the useful signal are not correlated, we assume that we can write

$$\sum_{n=0}^{N-1} [s(n)p(n-j) + p(n)s(n-j)] + \sum_{n=0}^{N-1} \sum_{k=1}^P [a_y(k) \cdot [s(n-k)p(n-j) + p(n-k)s(n-j)]] \approx 0$$

(11)

[0037] As a result equation Eq (10) is reduced to:

$$\sum_{n=0}^{N-1} (s(n)s(n-j) + p(n)p(n-j)) = - \sum_{n=0}^{N-1} \sum_{k=1}^P a_y(k) \cdot [s(n-k)s(n-j) + p(n-k)p(n-j)]$$

(12)

[0038] Assuming that the estimation $\hat{s}(n)$ from Eq (4) is close to the signal $s(n)$ ($s(n) \approx \hat{s}(n)$) we can write:

$$\begin{aligned} \sum_{n=0}^{N-1} s(n)s(n-j) &= - \sum_{n=0}^{N-1} \sum_{k=1}^P a_s(k) \cdot s(n-k)s(n-j) \\ &= - \sum_{k=1}^P a_s(k) \cdot \sum_{n=0}^{N-1} s(n-k)s(n-j) \end{aligned}$$

(13)

[0039] In the same way, we can write:

$$\begin{aligned} \sum_{n=0}^{N-1} p(n)p(n-j) &= - \sum_{n=0}^{N-1} \sum_{k=1}^P a_p(k) \cdot p(n-k)p(n-j) \\ &= - \sum_{k=1}^P a_p(k) \cdot \sum_{n=0}^{N-1} p(n-k)p(n-j) \end{aligned} \quad (14)$$

[0040] As a result Eq (11) leads to:

$$\begin{aligned} - \sum_{k=1}^P a_s(k) \sum_{n=0}^{N-1} s(n-k)s(n-j) - \sum_{k=1}^P a_p(k) \sum_{n=0}^{N-1} p(n-k)p(n-j) = \\ \sum_{n=0}^{N-1} \sum_{k=1}^P a_y(k) \cdot [s(n-k)s(n-j) + p(n-k)p(n-j)] \end{aligned} \quad (15)$$

[0041] Let's introduce the covariance functions of the useful signal $s(n)$ and of the perturbation $p(n)$

$$r_u(i-j) = \sum_{n=0}^{N-1} u(n-i) \cdot u(n-j), \quad u = s, p, y \quad (16)$$

as well as the autocorrelation matrix of $s(n)$, $p(n)$ and $y(n)$

$$\Gamma_u = (r_u(i-j))_{1 \leq i, j \leq P}, \quad u = s, p, y \quad (17)$$

[0042] For e.g. $P=4$, Γ_u can be written as $\Gamma_u = \begin{pmatrix} r_u(1) & r_u(2) & r_u(3) & r_u(4) \\ r_u(2) & r_u(1) & r_u(2) & r_u(3) \\ r_u(3) & r_u(2) & r_u(1) & r_u(2) \\ r_u(4) & r_u(3) & r_u(2) & r_u(1) \end{pmatrix}$

[0043] Let's also introduce the LPC coefficients vector of the different signals considered ($s(n)$, $p(n)$ and $y(n)$):

$$\mathbf{A}_u = (a_u(k))_{k=1, \dots, P}^T, \quad u = s, p, y \quad (18)$$

[0044] Where T stands for the transposition operator.
With the new notation introduced Eq (14) becomes:

$$-\Gamma_s \cdot \mathbf{A}_s - \Gamma_p \cdot \mathbf{A}_p = -(\Gamma_s + \Gamma_p) \cdot \mathbf{A}_y \quad (19)$$

[0045] As a result the LPC coefficients of the useful signal can be obtained through:

$$\mathbf{A}_s = \Gamma_s^{-1} \cdot [(\Gamma_s + \Gamma_p) \cdot \mathbf{A}_y - \Gamma_p \cdot \mathbf{A}_p] \quad (20)$$

[0046] Or equivalently:

$$\mathbf{A}_s = \mathbf{A}_y + \Gamma_s^{-1} \cdot \Gamma_p [\mathbf{A}_y - \mathbf{A}_p] \quad (21)$$

[0047] The two last expressions show that the computation of the useful signal LPC is obtained when the following entities are known: the LPC of the perturbed signal (\mathbf{A}_y), the LPC of the perturbation (\mathbf{A}_p), the covariance matrix of the perturbation (Γ_p), and the inverse of the covariance matrix of the useful signal (Γ_s). One can see the formula as a filter of the perturbed LPC \mathbf{A}_y to obtain the useful LPC \mathbf{A}_s , this filter depending on \mathbf{A}_p , Γ_p and Γ_s .

[0048] Accordingly, the present invention proposes a method based on formula Eq. (20)-(21) or on any formula derived from this equation to obtain the LPC coefficients of the useful signal (\mathbf{A}_s), when the LPC coefficients of the perturbed signal (\mathbf{A}_y) are available. Eq. (20)/(21) require to know the LPC coefficients of the perturbation \mathbf{A}_p , the correlation matrix of the perturbation Γ_p and the inverse of the correlation matrix of the useful signal Γ_s^{-1} . Generally, these entities are not directly available as we place our problem in a scheme where only the perturbed coefficients \mathbf{A}_y are available. Accordingly \mathbf{A}_p , Γ_p and Γ_s^{-1} need to be estimated. It results that the invention can be seen as the generic process described below:

[0049] For each frame m, the LPC \mathbf{A}_y are available.

- Estimation of the LPC of the perturbation \mathbf{A}_p based on the LPC \mathbf{A}_y . As an alternative, if more entities than \mathbf{A}_y are available the estimation can be based on the additive information. For instance, when placing the method within a speech codec, other speech codec parameters can be used to obtain the estimation.
- Estimation of the correlation matrix of the perturbation Γ_p based on the LPC \mathbf{A}_y . As an alternative, if more entities than \mathbf{A}_y are available the estimation can be based on the additive information. For instance, when placing the method within a speech codec, other speech codec parameters can be used to obtain the estimation.
- Estimation of the correlation matrix of the useful signal Γ_s based on the LPC \mathbf{A}_y . As an alternative, if more entities than \mathbf{A}_y are available the estimation can be based on the additive information. For instance, when placing the method within a speech codec, other speech codec parameters can be used to obtain the estimation.

Applying the filter defined in Eq (20)/(21) to get \mathbf{A}_s .

[0050] This process can be applied on speech codec bitstream by applying the following steps:

- For each frame m, extracting the LPC coefficients \mathbf{A}_y from the speech codec bitstream.
- Applying the processed as described previously to get the useful signal LPC \mathbf{A}_s .
- Exchanging the coefficients \mathbf{A}_y with the useful one \mathbf{A}_s

[0051] The solution has following advantages:

- The method makes it possible to obtain clean LPC coefficients when knowing perturbed ones. Any system based on linear prediction, computing coefficients \mathbf{A}_y can use this solution to obtain the clean coefficients \mathbf{A}_s .
- A typical application of the method is processing noise reduction, echo reduction or reduction of any other perturbation on the LPC on speech.
- When PCM samples are not available, but only codec parameters, the methods permits to re-construct the useful LPC parameters without the need to decode the bitstream to get the PCM data and apply classical noise reduction or echo cancellation. This is an alternative solution to existing prior art solutions.
- In the LPC analysis step (speech encoding), such method can be applied in parallel to the LPC analysis. It can be seen as a kind of postfilter after the LPC analysis.
- The mathematical expression obtained in Equation (21) is relatively easy to implement. It requires of course estimations of certain entities like cross-correlation functions or LPC or the noisy signal, but such estimations are quite classical. The method is accordingly quite classical from a signal processing point of view, and possible to implement in real time applications.

[0052] The invention is demonstrated with help of the drawings.

Figure 1 shows a signal transmission from a sender to a receiver in a telecommunication system.

Figure 2 is a flow chart for the modification of LPC coefficients according to a first embodiment.

5 Figure 3 shows a comparison of transfer function with non-modified LPC versus modified LPC coefficients.

Figure 4 shows a second embodiment for the modification of LPC coefficients.

10 **[0053]** Figure 1 shows an embodiment of a telecommunication system 1 in a signal transmission with modified LPC coefficients. The sender 2 generates the useful signal $s(t)$ by talking. Perturbations generate a perturbation signal $p(t)$ which is added to the useful signal resulting in the signal $y(t)$. The signal $y(t)$ is digitalized in the Analog-Digital-Converter (AD-Converter) 3 which generates a digital signal $y(n)$. The digital signal $y(n)$ is encoded in the encoder to the signal $y_e(n)$.

[0054] The encoding is done with the help of an LPC analysis. The encoded signal $y_e(n)$ is transmitted via the transmission block 5 to the decoder 6. The decoder 6 receives the signal $y_e'(n)$ from the transmission block and decodes $y_e'(n)$ to a digital signal $y_d(n)$. Depending on how the transmission block 5 is implemented, $y_e'(n)$ is either equal or unequal to $y_e(n)$. The transmission block 5 is e.g. a telephone switch, a router or a simple wire.

[0055] $y_d(n)$ is finally DA-converted by the DA-converter to $y_a(t)$ which is received as an analog signal by the receiver 8.

[0056] In the embodiment of Figure 2, the modification of the LPC parameters is done in the transmission block 5, whereas in the embodiment of Figure 4, the encoder 4 directly modifies the LPC parameters.

20 **[0057]** Figure 2 is a flow chart for the modification of LPC coefficients within the transmission block 5. $y_e(n)$ is a bitstream including LPC coefficients. If the encoder uses the AMR codec, the LPC coefficients are transmitted as Line Spectral Pair (LSP). The frames of $y_e(n)$ also comprise the parameters pitch delay, fixed codebook index, fixed gain and adaptive gain. The bitstream is computed by the analysis of successive frames, each, each comprising a defined number of samples (generally 160). If the signal $y_e(t)$ is sampled at a frequency of 8 MHz, in the so-called narrow band, the number of LPC coefficients is chosen to 8 or 10 in current standardized codecs (AMR, EFR, FR). In other words, the codec uses 8th respectively a 10th order linear prediction filter. In Eq. (1) k runs from 1 to 9 respectively from 1 to 11.

[0058] In the case of the so-called wide band, the sampling frequency is 16 kHz and the number of coefficients is preferably chosen to 16 in current standardized codecs (AMR-WB).

30 **[0059]** The Figure 2 shows the flow chart where the LPC coefficients are extracted from the bitstream $y_e(n)$. The bitstream is divided in the LPC coefficients and the rest of the bitstream, including the information needed to decode the residual waveform. Then the estimations of \mathbf{A}_p , Γ_p and Γ_s^{-1} are applied, taking into account the LPC coefficient as well as eventually additive information from the bitstream.

[0060] In an embodiment, \mathbf{A}_p is generated by the help of a Voice Activity Detection (VAD). Voice Activity Detection is known in the art. Here, the output of the VAD generates zero if no voice signal is detected in the bitstream, else the VAD outputs a one. $\mathbf{A}_p(m)$ is calculated by the following algorithm, whereby m is the index of the frame or subframe.

35 if VAD = 0,

$$\mathbf{A}_p(m) = \alpha \mathbf{A}_y(m) + (1 - \alpha) \mathbf{A}_p(m-1)$$

if VAD = 1,

$$\mathbf{A}_p(m) = \mathbf{A}_p(m-1)$$

α is a fixed parameter being heuristically chosen, whereby $0 < \alpha < 1$.

50 **[0061]** To estimate Γ_p , the perturbation is assumed to be white noise. Accordingly, the autocorrelation matrix Γ_p has the following form:

$$\Gamma_p = \begin{pmatrix} E_p(m) & 0 & 0 & 0 \\ 0 & E_p(m) & 0 & 0 \\ 0 & 0 & E_p(m) & 0 \\ 0 & 0 & 0 & E_p(m) \end{pmatrix}$$

[0062] Depending on the output of VAD, $E_p(m)$ is calculated by the equations if VAD = 0,

$$E_p(m) = \beta E_y(m) + (1 - \beta) E_p(m-1)$$

if VAD = 1,

$$E_p(m) = E_p(m-1) ,$$

wherein by $E_y(m)$ is the energy of the signal $y(n)$, m indicates a frame or subframe and β is a fixed parameter being heuristically chosen, $0 < \beta < 1$.

[0063] In this embodiment, Γ_s is estimated by the equation:

$$\Gamma_s = \Gamma_y - \Gamma_p .$$

Γ_y may be calculated with the help of Eq. 15 if the data stream is decoded.

[0064] In this embodiment, the bitstream of the encoded signal has to be decoded to make the estimation of Γ_p , Γ_s . Alternatively, the estimation of this matrices and vectors can also be done on the basis of the codec parameters of the signal $y_e(n)$ by interpreting the fixed gain and the adaptive gain.

[0065] After the estimation of \mathbf{A}_s , Γ_p , Γ_s , the clean LPC coefficients \mathbf{A}_s are generated by one of the equations 21 or 22. It should be noticed that the calculated clean LCP \mathbf{A}_s are an estimation of a LPC of the useful signal $s(n)$. Accordingly, the calculated LPC \mathbf{A}_s are as good as the estimations for \mathbf{A}_s , Γ_p , Γ_s are.

[0066] The filter on the LPC \mathbf{A}_y coefficients is applied to get the clean LPC \mathbf{A}_s and finally the LPC are replaced in the bitstream by changing each frame by the use of the clean LPC parameters \mathbf{A}_s . The frames are modified sequentially and sent to the decoder as signals $y_e'(n)$.

[0067] This method of improving the speech signal quality can be done anywhere in the path between the encoder and decoder. For example, in telecommunication systems, the method can be applied in the terminal of the sender, in the terminal of the receiver or in one of the routers telephone switches or gateways between different networks.

[0068] The use of the modified LPC coefficients improves the quality of the received signal which is demonstrated with the help of Figure 3.

[0069] Figure 3 shows a comparison of transfer functions with non-modified LPC versus modified LPC coefficients. The synthesis LPC filter function can be described by the filter transfer function $H(f)$ in the frequency domain. The graph of Fig. 3 shows a functions $H(f)$ dependent on the frequency f for a non-noisy LPC function and, with the dashed line, for a noisy LPC filter. The transfer function of the noisy LPC filter, in our case a non-modified LPC, has more energy but is smoother. Using a LPC that was generated on the basis of a noisy signal worsens the quality of the speech. Hence, the modification of the LPC to a clean LPC make it easier for the receiver to understand the received speech and enhance the clarity of the speech.

[0070] If only the LPC's are modified, the received speech still includes noise. Therefore, as an option, the flow chart of Figure 2 may be extended by an additional step which reduces noise on the rest bitstream. This noise reduction is performed after the estimation of \mathbf{A}_p , Γ_p and Γ_s^{-1} and before the generation the new frames of the bitstream of the signal.

[0071] One exemplary noise reduction technique for the rest bitstream is the method for reducing noise on the codec parameters pitch gain and codebook gain described in the above-mentioned "Compressed Domain Noise Reduction

and Echo Suppression for Network Speech".

[0072] Figure 4 shows a second embodiment of the modification of LPC coefficients. In this case, the function is included in the encoder 4.

[0073] In speech encoding, like typically AMR, AMR-WB, G723, G729, the LPC coefficient are computed by an analysis of successive weighted frames. Typically, the Levinson-Durbin algorithm permits to get the LPC coefficient from the sample $y(n)$ of the analysis frame. Our method maybe placed as a postfilter of the computation blocks of the LPC analysis. In this scenario, it enhances the LPC coefficient by reducing the influence of the noise. The needed estimations (\mathbf{A}_p , Γ_p and Γ_s^{-1}) may be done by using the LPC coefficients but also some additive information of sample $y(n)$ as depicted in Fig 4. Finally, the filter of Eq (20)/(21) is applied on the perturbed coefficients to get the enhanced ones.

[0074] In the embodiment of Figure 4, the method of improving the speech quality is performed within the encoder. The encoder receives the samples from the A/D-converter. The samples are organized as frames. After windowing a frame, the LPC analysis outputs the LCP coefficients \mathbf{A}_y . The parameters \mathbf{A}_p , Γ_p and Γ_s^{-1} are estimated like one of the embodiments described above. LPC coefficients \mathbf{A}_s are calculated by one of the equations (21) and (22). The encoding of the frame is done with \mathbf{A}_s .

Reference number list

[0075]

- 1 telecommunication system
- 2 sender
- 3 A/D-converter
- 4 encoder
- 5 transmission block
- 6 decoder
- 7 D/A-converter
- 8 receiver

Claims

1. Method for transmitting a digital signal $y(n)$, $y(n)$ comprising a useful signal $s(n)$ and a perturbation signal $p(n)$, the method comprising the steps of:

- receiving the Linear Prediction Coefficients (LPC) \mathbf{A}_y of a bitstream $y_e(n)$, $y_e(n)$ being an LPC-encoded signal of $y(n)$;
- estimating the autocorrelation matrix Γ_s of the useful signal $s(n)$, of the autocorrelation matrix Γ_p of the perturbation signal $p(n)$ and the LPC \mathbf{A}_p of the perturbation signal $p(n)$;
- calculating a modified LPC \mathbf{A}_s by using \mathbf{A}_y and the estimated Γ_s , Γ_p , \mathbf{A}_p ;
- outputting a modified data stream $y_e'(n)$ including the modified LPC \mathbf{A}_s .

2. Method according to claim 1, wherein \mathbf{A}_s is calculated by $\mathbf{A}_s = \Gamma_s^{-1} \cdot [(\Gamma_s + \Gamma_p) \cdot \mathbf{A}_y - \Gamma_p \cdot \mathbf{A}_p]$.

3. Method according to claim 1, further comprising the step:

- receiving the residual signal of the encoded signal $y_e(n)$

and wherein in the step of

- estimating Γ_s , Γ_p , \mathbf{A}_y

the estimated values for at least one of the parameters Γ_s , Γ_p , \mathbf{A}_y are calculated from the residual signal of $y_e(n)$.

4. Method according to one of the preceding claims, further comprising the steps:

- performing a noise reduction on the residual signal of the encoded signal $y_e(n)$.

5. Method according to one or the claims 1 to 4,
in which instead of

- receiving the Linear Prediction Coefficients (LPC) A_y of the signal $y_e(n)$,

the method comprises the steps of

- receiving the digital signal $y(n)$ and
- calculating Linear Prediction Coefficients (LPC) A_y by encoding the signal $y(n)$.

6. Method according to one of the preceding claims
characterized in that
the method is performed in a telecommunication system.

7. Digital signal transmission apparatus
which performs a method according to one of the claims one to 6.

8. Digital signal transmission apparatus according to claim 7, comprising a Digital Signal Processor (DSP).

9. Digital signal transmission apparatus according to claim 7 being a telecommunication terminal.

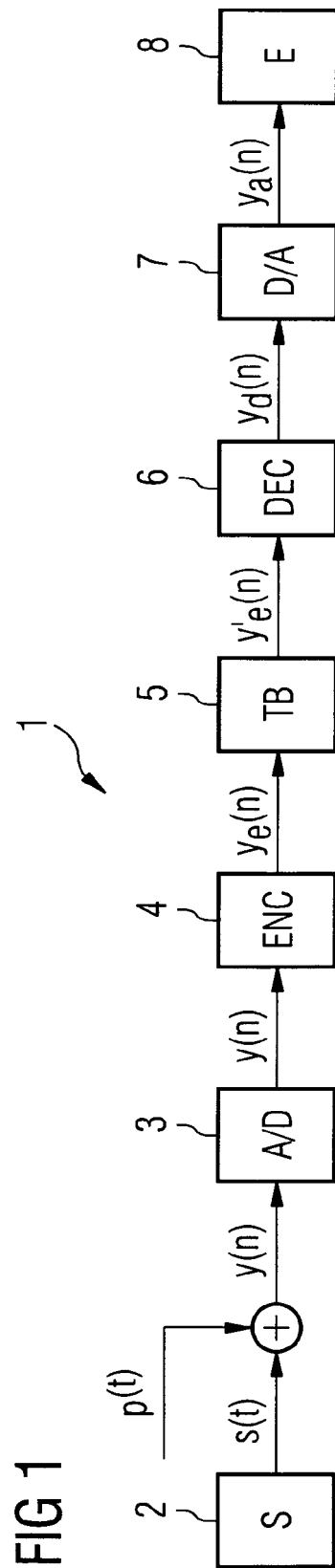


FIG 2

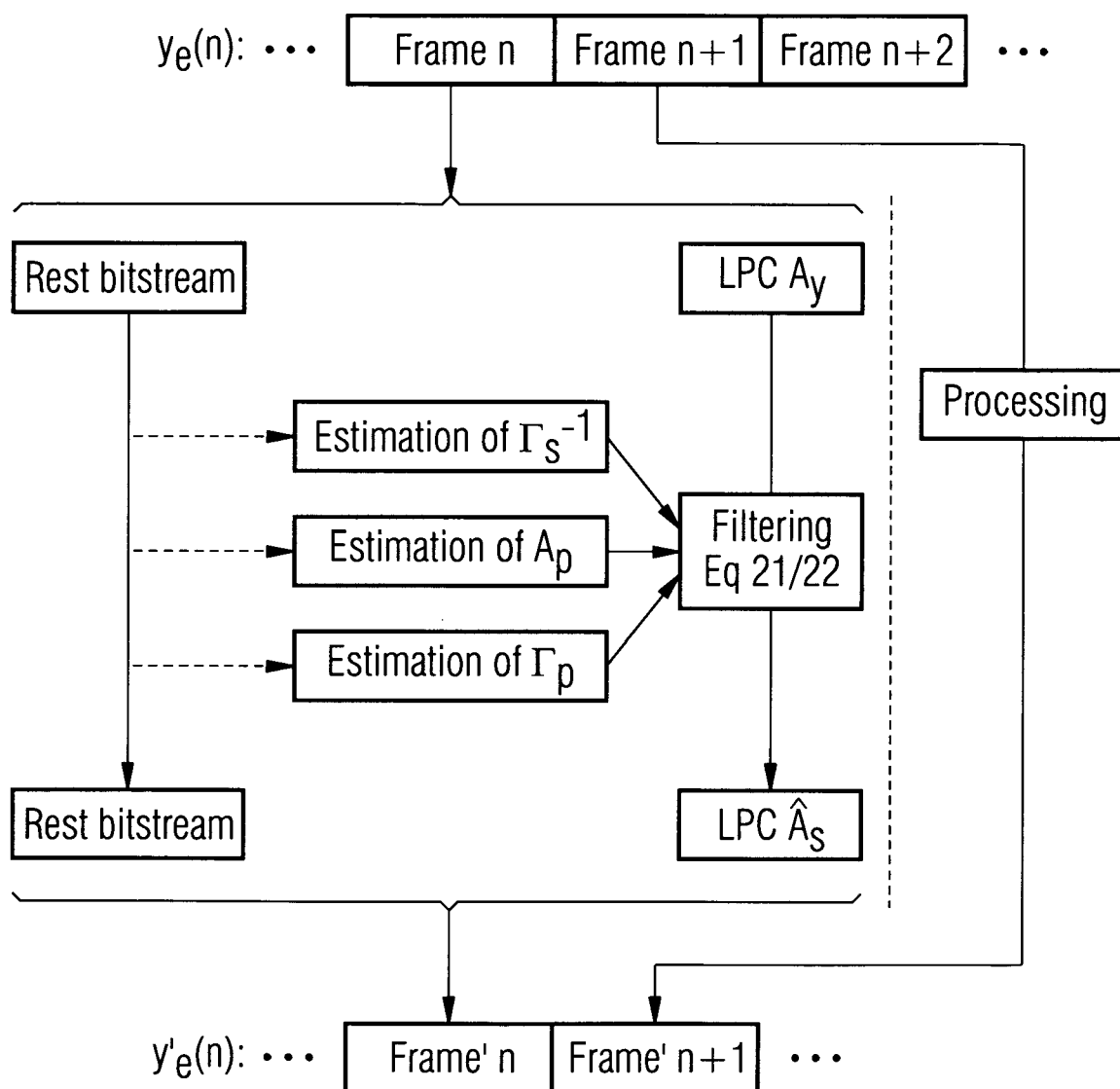


FIG 3

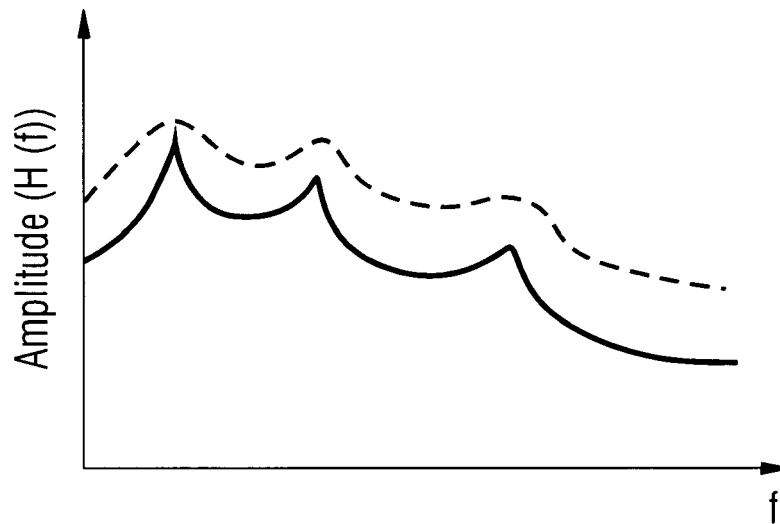
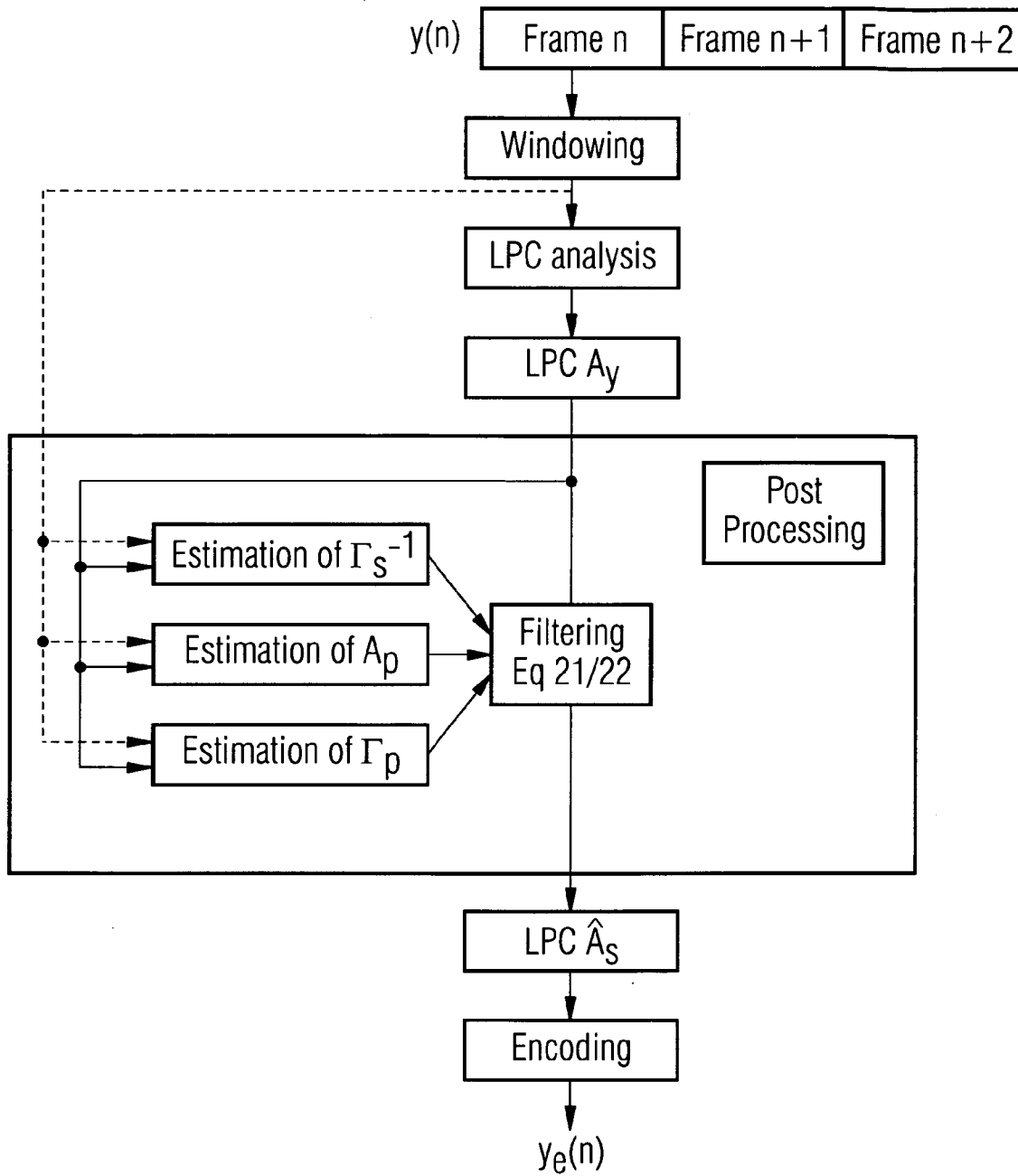


FIG 4





European Patent
Office

EUROPEAN SEARCH REPORT

Application Number
EP 07 00 0716

DOCUMENTS CONSIDERED TO BE RELEVANT			
Category	Citation of document with indication, where appropriate, of relevant passages	Relevant to claim	CLASSIFICATION OF THE APPLICATION (IPC)
A	WO 02/080149 A1 (ERICSSON TELEFON AB L M [SE]) 10 October 2002 (2002-10-10) * abstract * * page 1 - page 2, line 11 * * page 4, line 6 - page 8 * * figures 4-8,10 *	1,2,5-9	INV. G10L21/02 G10L19/06 H04M9/08
D,A	CHANDRAN R ET AL: "Compressed domain noise reduction and echo suppression for network speech enhancement" CIRCUITS AND SYSTEMS, 2000. PROCEEDINGS OF THE 43RD IEEE MIDWEST SYMPOSIUM ON AUGUST 8-11, 2000, PISCATAWAY, NJ, USA, IEEE, vol. 1, 8 August 2000 (2000-08-08), pages 10-13, XP010558066 ISBN: 0-7803-6475-9 * the whole document *	1,3-9	
A	WO 02/054744 A (NOKIA CORP [FI]; KOISTINEN TOMMI [FI]; KIRLA OLLI [FI]) 11 July 2002 (2002-07-11) * abstract * * page 2 * * page 4, line 25 - page 7, line 20 * * page 10 - page 13, line 7 * * page 16, line 24 - page 36 * * figures 1-8 *	1,5-9	TECHNICAL FIELDS SEARCHED (IPC) G10L H04M
The present search report has been drawn up for all claims			
Place of search Munich		Date of completion of the search 23 May 2007	Examiner Tilp, Jan
CATEGORY OF CITED DOCUMENTS X : particularly relevant if taken alone Y : particularly relevant if combined with another document of the same category A : technological background O : non-written disclosure P : intermediate document T : theory or principle underlying the invention E : earlier patent document, but published on, or after the filing date D : document cited in the application L : document cited for other reasons & : member of the same patent family, corresponding document			

3
EPO FORM 1503 03.82 (P04C01)

**ANNEX TO THE EUROPEAN SEARCH REPORT
ON EUROPEAN PATENT APPLICATION NO.**

EP 07 00 0716

This annex lists the patent family members relating to the patent documents cited in the above-mentioned European search report.
The members are as contained in the European Patent Office EDP file on
The European Patent Office is in no way liable for these particulars which are merely given for the purpose of information.

23-05-2007

Patent document cited in search report	Publication date	Patent family member(s)	Publication date
WO 02080149 A1	10-10-2002	CN 1500261 A	26-05-2004
		DE 10296562 T5	22-04-2004
		GB 2390790 A	14-01-2004
		SE 521693 C2	25-11-2003
		SE 0102519 A	01-10-2002
		US 2002184010 A1	05-12-2002

WO 02054744 A	11-07-2002	CN 1504042 A	09-06-2004
		EP 1346553 A1	24-09-2003
		US 2004076271 A1	22-04-2004

REFERENCES CITED IN THE DESCRIPTION

This list of references cited by the applicant is for the reader's convenience only. It does not form part of the European patent document. Even though great care has been taken in compiling the references, errors or omissions cannot be excluded and the EPO disclaims all liability in this regard.

Non-patent literature cited in the description

- **R. LE BOUQUIN JEANNÈS ; P. SCALART ; G. FAUCON ; C. BEAUGEANT.** Combined Noise and Echo Reduction in Hands-Free systems: A Survey. *IEEE Trans. On Speech and Audio Processing*, November 2001, vol. 9, 808-820 [0007]
- **CHANDRAN, RAVI ; MARCHOK, DANIEL J.** Compressed Domain Noise Reduction and Echo Suppression for Network Speech. *Enhancement, Proc. of the 43rd IEEE Midwest Symposium on Circuits and Systems*, 2000, 10-13 [0010]
- **HERVÉ TADDEI ; CHRISTOPHE BEAUGEANT ; MICHAEL DE MEULENEIRE.** Noise reduction on speech codec parameters. *ICASSP*, 2004 [0010]
- **C. BEAUGEANT ; N. DUETSCH ; H. TADDEI.** Gain Loss Control based on Speech Codec Parameters. *Eusipco*, 2004 [0010]