(19) **DANMARK**

(10) **DK/EP 3709298 T3**

Patent- og
Varemærkestyrelsen

(12) Oversættelse af
europæisk patentskrift

(51) Int.Cl.: ***G 10 L 19/20 (2013.01)*** ***G 10 L 19/26 (2013.01)*** *G 10 L 19/08 (2013.01)*
G 10 L 19/22 (2013.01) *G 10 L 25/78 (2013.01)* *G 10 L 25/81 (2013.01)*
G 10 L 25/93 (2013.01)

(45) Oversættelsen bekendtgjort den: **2025-01-13**

(80) Dato for Den Europæiske Patentmyndigheds
bekendtgørelse om meddelelse af patentet: **2024-11-20**

(86) Europæisk ansøgning nr.: **20172813.6**

(86) Europæisk indleveringsdag: **2012-11-01**

(87) Den europæiske ansøgnings publiceringsdag: **2020-09-16**

(30) Prioritet: **2011-11-03 US 201161555246 P**

(62) Stamansøgningsnr: **12844916.2**

(84) Designerede stater: **AL AT BE BG CH CY CZ DE DK EE ES FI FR GB GR HR HU IE IS IT LI LT LU LV MC MK MT NL NO PL PT RO RS SE SI SK SM TR**

(73) Patenthaver: **VOICEAGE EVS LLC, 620 Newport Centre Drive , Suite 1100, Newport Beach, CA 92660, USA**

(72) Opfinder: **VAILLANCOURT, Tommy, 835, rue Farrand, Sherbrooke, Québec J1N 2K1, Canada**
JELINEK, Milan, 1355 Emile-Nelligan, Sherbrooke, Québec J1L 2W8, Canada

(74) Fuldmægtig i Danmark: **NORDIC PATENT SERVICE A/S, Bredgade 30, 1260 København K, Danmark**

(54) Benævnelse: **FORBEDRING AF IKKE-TALEINDHOLD TIL CELP-AFKODER MED LAV HASTIGHED**

(56) Fremdragne publikationer:
US-A1- 2010 114 567
US-B1- 6 240 386
US-B2- 6 704 711

DESCRIPTION

Description

Field

[0001] The present disclosure relates to a device and method for modifying a synthesis of a time-domain excitation decoded by a time-domain decoder.

Background

[0002] A state-of-the-art conversational codec can represent with a very good quality a clean speech signal with a bit rate of around 8 kbps and approach transparency at a bit rate of 16 kbps. To sustain this high speech quality even at low bit rate a multi modal coding scheme may be used. Usually the input sound signal is split among different categories reflecting its characteristics. For example, the different categories may include voiced, unvoiced and onset. The codec uses different coding modes optimized for all these categories.

[0003] However, some deployed speech codecs do not use this multi modal approach resulting in a suboptimal quality especially at low bit rates for a sound signal different from clean speech. When a codec is deployed, it is hard to modify the encoder due to the fact that the bitstream is standardized and any modification to the bitstream would break the interoperability of the codec. However modifications to the decoder can be implemented to improve the quality perceived on the receiver side.

[0004] United States Patent No. US 6 240 386 discloses a speech codec employing noise classification for noise compensation. A multi-rate speech codec supports a plurality of encoding bit rate modes by adaptively selecting encoding bit rate modes to match communication channel restrictions. In higher bit rate encoding modes, an accurate representation of speech through CELP (code excited linear prediction) and other associated modeling parameters are generated for higher quality decoding and reproduction. For each bit rate mode selected, pluralities of fixed or innovation subcodebooks are selected for use in generating innovation vectors. The speech coder distinguishes various voice signals as a function of their voice content. For example, a Voice Activity Detection (VAD) algorithm selects an appropriate coding scheme depending on whether the speech signal comprises active or inactive speech. The encoder may consider varying characteristics of the speech signal including sharpness, a delay correlation, a zero-crossing rate, and a residual energy. In another embodiment of the present invention, code excited linear prediction is used for voice active signals whereas random excitation is used for voice inactive signals; the energy level and spectral content of the voice inactive signal may also be used for noise coding. The multi-rate speech codec may employ distributed detection and compensation processing the speech signal. For high quality perceptual speech reproduction, the speech codec may perform noise detection in both an encoder and a decoder. The noise detection may be coordinated between the encoder and decoder. Similarly, noise compensation may be performed in a distributed manner among both the decoder and the encoder.

Summary

[0005] According to a first aspect, the present invention relates to device for modifying a synthesis of a time-domain excitation decoded by a time-domain decoder according to claim 1.

[0006] According to another aspect, the present invention relates to device for decoding a sound signal encoded by encoding parameters, comprising: a decoder of a time-domain excitation in response to the sound signal encoding parameters; a synthesis filter responsive to the decoded time-domain excitation to produce a synthesis of said time-domain excitation; and the above described device for modifying the synthesis of the time-domain excitation.

[0007] According to a third aspect, the present invention relates to a method for modifying a synthesis of a time-domain excitation decoded by a time-domain decoder according to claim 8.

[0008] According to a further aspect, the present invention is concerned with a method for decoding a sound signal encoded by encoding parameters, comprising: decoding a time-domain excitation in response to the sound signal encoding parameters; synthesizing the decoded time-domain excitation to produce a synthesis of said time-domain excitation; and the above described method for modifying the synthesis of the time-domain excitation.

[0009] The foregoing and other features of the device and method for modifying the synthesis of a time-domain excitation will become more apparent upon reading of the following non restrictive description, given by way of non limitative example with reference to the accompanying drawings.

Brief description of the drawings

[0010] In the appended drawings:

Figure 1 is a simplified schematic diagram showing modification of a CELP decoder for inactive and active unvoiced frames improvement;

Figure 2 is a detailed schematic diagram showing the CELP decoder modification for inactive and active unvoiced frames improvement;

Figure 3 is a simplified schematic diagram showing modification of a CELP decoder for generic audio frames improvement; and

Figure 4 is a detailed schematic diagram showing the CELP decoder modification for generic audio frames improvement.

Description

[0011] The present disclosure relates to an approach to implement on the decoder side a multimodal decoding such that interoperability is maintained and the perceived quality is increased. In the disclosure, although AMR-WB as described in reference [3GPP TS 26.190, "Adaptive Multi-Rate - Wideband (AMR-WB) speech codec; Transcoding functions] is used as illustrative example, it should be kept in mind that this approach can be applied to other types of low bit rate speech decoders as well.

[0012] Referring to Figure 1, to achieve this multimodal decoding, a time-domain excitation decoder 102 first decodes entirely the received bitstream 101, for example the AMR-WB bitstream, to get a complete time-domain Code-Excited Linear Prediction (CELP) decoded excitation. The decoded time-domain excitation is processed through a Linear Prediction (LP) synthesis filter 103 to obtain a speech/sound signal time-domain synthesis at the inner sampling frequency of the decoder. For AMR-WB, this inner sampling frequency is 12.8 kHz, but for another codec it could be different.

[0013] The time-domain synthesis of the current frame from the LP synthesis filter 103 is processed through a classifier 104-105-106-301 (Figures 1, 2 and 3) supplied with voice activity detection (VAD) information 109 from the bitstream 101. The classifier 104-105-106-301 analyses and categorizes the time-domain synthesis either as inactive speech, active voiced speech, active unvoiced speech, or generic audio. Inactive speech (detected at 1051) includes all background noises between speech burst, active voiced speech (detected at 1061) represents a frame during an active speech burst having voiced characteristics, active unvoiced speech (detected at 1062) represents a frame during a speech burst having unvoiced characteristics, and generic audio (detected at 3010) represents music or reverberant speech. Other categories can be added or derived from the above categories. The disclosed approach aims at improving in particular, but not exclusively, the perceived quality of the inactive speech, the active unvoiced speech and the generic audio.

[0014] Once the category of the time-domain synthesis is determined, a converter/modifier 107 converts the decoded excitation from the time-domain excitation decoder 102 into frequency domain using a non-overlap frequency transform. An overlap transform can be used as well, but it implies an increase of the end-to-end delay which is not desirable in most cases. The frequency representation of the excitation is then split into different frequency bands in the converter/modifier 107. The frequency bands can have fixed size, can rely on critical bands [J. D. Johnston, "Transform coding of audio signal using perceptual noise criteria," IEEE J. Select. Areas Commun., vol. 6, pp. 314-323, Feb. 1988], or any other combinations. Then the energy per band is computed and kept in memory in the converter/modifier 107 for use after the reshaping process to ensure the modification does not alter the global frame energy level.

[0015] The modification of the excitation in the frequency domain as performed by the converter/modifier 107 may differ with the classification of the synthesis. For inactive speech and active unvoiced speech, the reshaping may consist of a normalization of the low frequencies with an addition of noise and replacement of the high frequency content with noise only. A cut-off frequency of the decoded time-domain synthesis, the limit between low and high frequency, can be fixed at a value around 1 to 1.2 kHz. Some of the low frequency content of the decoded time-domain synthesis is kept to prevent artifact when switching between a non-modified frame and a modified frame. It is also possible to make the cut-off frequency variable from frame to frame by choosing a frequency bin as a function of the decoded pitch from the time-domain excitation decoder 102. The modification process has as effect of removing the kind of electrical noise associated with the low bit rate speech codec. After the modification process, a gain matching per frequency band is applied to get back the initial energy level per frequency band with a slight increase of the energy for the frequencies above 6 kHz to compensate for an LP filter gain drop at those frequencies.

[0016] For a frame categorized as generic audio, the processing in the converter/modifier 107 is different. First the normalization is performed per frequency band for all the bands. In the normalization operation, all the bins inside a frequency band that are below a fraction of the maximum frequency value within the band are set to zero. For higher frequency bands, more bins are zeroed per band. This simulates a frequency quantification scheme with a high bit budget, but having more bits allocated to the lower frequencies. After the normalization process, a noise fill can be applied to replace the zeroed bins with random noise but, depending on the bit rate, the noise fill is not always used. After the modification process, a gain matching per frequency band is applied to get back the initial energy level per frequency band and a tilt correction depending on the bit rate is applied along the frequency band to compensate for the systematic under estimation of the LP filter in case of generic audio input. Another differentiation for the generic audio path comes from the fact that the gain matching is not applied over all frequency bins. Because the spectrum of generic audio is usually more peaky than speech, the perceived quality is improved when it is possible to identify spectral pulses and to put some emphasis thereon. To do so, full gain matching with tilt correction is applied only to the highest energy bins inside a frequency band. For the lowest energy bins, only a fraction of the gain matching is applied to those bins. This results in increasing the spectral dynamic.

[0017] After the excitation frequency reshaping and gain matching, the converter/modifier 107 applies an inverse frequency transform to obtain the modified time-domain excitation. This modified excitation is processed through the LP synthesis filter 108 to obtain a modified time-domain synthesis. An overwriter 110 simply overwrites the time-domain decoded synthesis from LP synthesis filter 103 with the modified time-domain synthesis from the LP synthesis filter 108 depending on the classification of the time-domain decoded synthesis before final de-emphasis and resampling to 16 kHz (for the example of AMR-WB) in a deemphasizing filter and resampler 112.

[0018] In case of inactive speech, the only difference compared to active unvoiced speech modification is the use of a smoother 111 for smoothing the LP synthesis filter 108 to give smoother noise variation. The remaining modifications are the same as for the active unvoiced path. In the following text a more detailed example of implementation of the disclosed approach is described with reference to Figure 2.

1) Signal classification

[0019] Referring to Figure 2, the classifier 104-105-106-301 performs at the decoder a classification of the time-domain synthesis 1021 of the speech/sound signal as described hereinabove for the bit rates where the modification is applied. For the purpose of simplification of the drawings, the LP synthesis filter 103 is not shown in Figure 2. Classification at the decoder is similar to that as described in references [Milan Jelinek and Philippe Gournay; PCT Patent application WO03102921A1, "A method and device for efficient frame erasure concealment in linear predictive based speech codecs"] and [T.Vaillancourt et al., PCT Patent application WO2007073604A1, "Method and device for efficient frame erasure concealment in speech codecs"], plus some adaption for the generic audio detection. The following parameters are used for the classification of the frames at the decoder: a normalized correlation r_X, a spectral tilt measure e_t, a pitch stability counter *pc*, a relative frame energy of the sound signal at the end of the current frame E_S, and a zero-crossing counter *zc*. The computation of these parameters which are used to classify the signal is explained below.

[0020] The normalized correlation r_X is computed at the end of the frame based on the speech/sound signal time-domain synthesis $s_{out}(n)$. The pitch lag of the last sub-frame from the time-domain excitation decoder 102 is used. More specifically, the normalized correlation r_X is computed pitch synchronously as follows:

$$r_x = \frac{\sum_{i=0}^{T-1} x(t+i)x(t+i-T)}{\sqrt{\sum_{i=0}^{T-1} x^2(t+i)\sum_{i=0}^{T-1} x^2(t+i-T)}}, \qquad (1)$$

where $x(n)=s_{out}(n)$, T is the pitch lag of the last sub-frame, $t=L-T$, and L is the frame size. If the pitch lag of the last sub-frame is larger than 3N12 (N being the sub-frame size), T is set to the average pitch lag of the last two sub-frames.

[0021] Therefore, the normalized correlation r_X is computed using the speech/sound signal time-domain synthesis $s_{out}(n)$. For pitch lags lower than the sub-frame size (64 samples) the normalized correlation is computed twice at instants $t=L-T$ and $t=L-2T$, and the normalized correlation r_x is given as the average of these two computations.

[0022] The spectral tilt parameter e_t contains the information about the frequency distribution of energy. As a non limitative example, the spectral tilt at the decoder is estimated as the first normalized autocorrelation coefficient of the time-domain synthesis. It is computed based on the last 3 sub-frames as:

$$e_t = \frac{\sum_{i=N}^{L-1} x(i)x(i-1)}{\sum_{i=N}^{L-1} x^2(i)} \qquad (2)$$

where $x(n) = \boldsymbol{s}_{out}(n)$ is the time-domain synthesis signal, N is the sub-frame size, and L is the frame size (N=64 and L=256 in the example of AMR-WB).

[0023] The pitch stability counter pc assesses the variation of the pitch period. It is computed at the decoder as follows:

$$pc = |p_3 + p_2 - p_1 - p_0| \qquad (3)$$

[0024] The values p_0, p_1, p_2 and p_3 correspond to the closed-loop pitch lag from the 4 sub-frames of the current frame (in the example of AMR-WB).

[0025] The relative frame energy E_s is computed as a difference between the current frame energy E_f in dB and its long-term average E_{lt}

$$E_s - E_f - E_{lt} \qquad (4)$$

where the current frame energy E_f is the energy of the time-domain synthesis $\boldsymbol{s}_{out}(n)$ in dB computed pitch synchronously at the end of the frame as

$$E_f = 10log_{10}\left(\frac{1}{T}\sum_{i-0}^{T-1} s_{out}^2(i+L-T)\right) \qquad (5)$$

where L=256 (in the example of AMR-WB) is the frame length and T is the average pitch lag of the last two sub-frames. If T is less than the sub-frame size then T is set to 2T (the energy computed using two pitch periods for short pitch lags).

[0026] The long-term averaged energy is updated on active speech frames using the following relation:

$$E_{lt} = 0.99E_{lt} + 0.01E_f \qquad (6)$$

[0027] The last parameter is the zero-crossing counter zc computed on one frame of the time-domain synthesis $\boldsymbol{s}_{out}(n)$. As a non limitative example, the zero-crossing counter zc counts the number of times the sign of the time-domain synthesis changes from positive to negative during that interval.

[0028] To make the classification more robust, the classification parameters are considered together forming a function of merit fm. For that purpose, the classification parameters are first scaled using a linear function. Let us consider a parameter p_x, its scaled version is obtained using:

$$p^s = k_p \cdot p_x + c_p \qquad (7)$$

[0029] The scaled pitch stability counter *pc* is clipped between 0 and 1. The function coefficients k_p and c_p have been found experimentally for each of the parameters. The values used in this example of implementation are summarized in Table 1:

Table 1. Frame Classification Parameters at the decoder and the coefficients of their respective scaling functions

Parameter	Meaning	k_p	c_p
r_x	Normalized Correlation	0.8547	0.2479
e_t	Spectral Tilt	0.8333	0.2917
pc	Pitch Stability counter	-0.0357	1.6074
E_s	Relative Frame Energy	0.04	0.56
zc	Zero Crossing Counter	-0.04	2.52

[0030] The function of merit is defined as:

$$f_m = \frac{1}{6}(2 \cdot r_x^s + e_t^s + pc^s + E_s^s + zc^s) \qquad (8)$$

where the superscript s indicates the scaled version of the parameters.

[0031] The classification of the frames is then done using the function of merit f_m and following the rules summarized in Table 2:

Table 2: Signal Classification Rules at the decoder

Previous Frame Class	Rule	Current Frame Class
ONSET	$f_m \geq 0.63$	VOICED
VOICED		
VOICED TRANSITION		
ARTIFICIAL ONSET		
GENERIC AUDIO SOUND		
	$0.39 \leq f_m < 0.63$	VOICED TRANSITION
	$f_m < 0.39$	UNVOICED
UNVOICED TRANSITION	$f_m > 0.56$	ONSET
UNVOICED		

Previous Frame Class	Rule	Current Frame Class
	$0.56 \geq f_m > 0.45$	UNVOICED TRANSITION
	$f_m \leq 0.45$	UNVOICED
Current frame VAD information		
	VAD = 0	UNVOICED

[0032] In addition to this classification, the information 109 on the voice activity detection (VAD) by the encoder can be transmitted into the bitstream 101 (Figure 1) as it is the case with the example of AMR-WB. Thus, one bit is sent into the bitstream 101 to specify whether or not the encoder considers the current frame as active content (VAD = 1) or inactive content (background noise, VAD = 0). When the VAD information indicates that the content is inactive, the classifier portion 104, 105, 106 and 301 then overwrites the classification as UNVOICED.

[0033] The classification scheme also includes a generic audio detection (see classifier portion 301 of Figure 3). The generic audio category includes music, reverberant speech and can also include background music. A second step of classification allows the classifier 104-105-106-301 to determine with good confidence that the current frame can be categorized as generic audio. Two parameters are used to realize this second classification step. One of the parameters is the total frame energy E_f as formulated in Equation (5).

[0034] First, a mean of the past forty (40) total frame energy variations E_{df} is calculated using the following relation:

$$\overline{E}_{df} = \frac{\sum_{t=-40}^{t=-1} \Delta_E^t}{40}; \qquad where\ \ \Delta_E^t = E_f^t - E_f^{(t-1)} \qquad (9)$$

[0035] Then, a statistical deviation of the energy variation history σ_E over the last fifteen (15) frames is determined using the following relation:

$$\sigma_E = 0.7745967 \cdot \sqrt{\sum_{t=-15}^{t=-1} \frac{\left(\Delta_E^t - \overline{E}_{df}\right)^2}{15}} \qquad (10)$$

[0036] The resulting deviation σ_E gives an indication on the energy stability of the decoded synthesis. Typically, music has a higher energy stability (lower statistical deviation of the energy variation history) than speech.

[0037] Additionally, the first step classification is used to evaluate the interval between two frames classified as unvoiced N_{UV} when the frame energy *Ef,* as formulated in equation (5) is higher than -12dB. When a frame is classified as unvoiced and the frame energy E_f is greater than -9dB, meaning that the signal is unvoiced but not silence, if the long term active speech energy E_{lt}, as formulated in Equation (6), is below 40dB the unvoiced interval counter is set to 16, otherwise the unvoiced interval counter N_{UV} is decreased by 8. The counter N_{UV} is also limited between 0 and 300 for active speech signal and between 0 and 125 for inactive speech signal. It is reminded that, in the illustrative example, the difference between active and inactive speech signal may be deduced from the voice activity detection VAD information included in the bitstream 101.

[0038] A long term average is derived from this unvoiced frame counter as follow for active speech signal:

$$N_{uv_{lt}} = 0.9 \cdot N_{uv_{lt}} + 0.1 \cdot N_{uv} \qquad (11)$$

[0039] And as follows for inactive speech signal:

$$N_{uv_{lt}} = 0.95 \cdot N_{uv_{lt}} \qquad (12)$$

[0040] Furthermore, when the long term average is very high and the deviation σ_E is high, for example when $N_{UVlt} > 140$ *and* $\sigma_E > 5$ in the current example of implementation, the long term average is modified as follow:

$$N_{uv_{lt}} = 0.2 \cdot N_{uv_{lt}} + 80 \qquad (13)$$

[0041] This parameter on long term average of the number of frames between frames classified as unvoiced is used by the classifier 104-105-106-301 to determine if the frame should be considered as generic audio or not. The more the unvoiced frames are close in time, the more likely the frame has speech characteristics (less probably generic audio). In the illustrative example, the threshold to decide if a frame is considered as generic audio G_A is defined as follows:

A frame is G_A if :

$$N_{uv_{lt}} > 140 \text{ and } \Delta_E^t < 12 \qquad (14)$$

[0042] The parameter

Δ_E^t

, defined in equation (9), is added to not classify large energy variation as generic audio, but to keep it as active speech.

[0043] The modification performed on the excitation depends on the classification of the frame and for some type of frames there is no modification at all. The next table 3 summarizes the case where a modification can be performed or not.

Table 3: Signal category for excitation modification

Frame Classification	Voice activity detected? Y/N	Category	Modification Y/N
ONSET	Y (VAD=1)	Active voice	N
VOICED			
UNVOICED TRANSITION			
ARTIFICIAL ONSET			
GENERIC AUDIO SOUND	Y	Generic audio	Y*
VOICED TRANSITION	Y	Active unvoiced	Y
UNVOICED			
ONSET	N	Inactive audio	Y
VOICED			
UNVOICED TRANSITION			
ARTIFICIAL ONSET			
GENERIC AUDIO SOUND			
VOICED TRANSITION			
UNVOICED			
* The generic audio category may be modified or not depending on the implementation. For example, generic audio may be modified only when inactive, or generic audio may be modified only when active, all the time or not at all.			

2) Frequency transfom

[0044] During the frequency-domain modification phase, the excitation needs to be represented into the transform-domain. For example, the time-to-frequency conversion is achieved by a time-to-frequency domain converter 201 of the converter/modifier 107 using a type II DCT (Discrete Cosine Transform) giving a frequency resolution of 25 Hz but any other suitable transform can be used. In case another transform is used the frequency resolution (defined above), the number of frequency bands and the number of frequency bins per bands (defined further below) may need to be revised accordingly. The frequency representation of the time-domain CELP excitation f_e calculated in the time-to-frequency domain converter 201 is given below:

$$f_e(k) = \begin{cases} \sqrt{\frac{1}{L}} \cdot \sum_{n=0}^{L-1} e_{td}(n), & k = 0 \\ \sqrt{\frac{2}{L}} \cdot \sum_{n=0}^{L-1} e_{td}(n) \cdot \cos\left(\frac{\pi}{L}\left(n + \frac{1}{2}\right)k\right), & 1 \leq k \leq L-1 \end{cases} \qquad (15)$$

$$\lfloor \sqrt{L} \sum_{n=0} \quad \left(L\left(\quad 2 \right) \right)$$

where $e_{td}(n)$ is the time-domain CELP excitation, and L is the frame length. In the example of AMR-WB, the frame length is 256 samples for a corresponding inner sampling frequency of 12.8 kHz.

[0045] In a time-domain CELP decoder such as 102, the time-domain excitation signal is given by

$$e_{td}(n) = bv(n) + gc(n) \qquad (15)$$

where *v(n)* is the adaptive codebook contribution, *b* is the adaptive codebook gain, *c(n)* is the fixed codebook contribution, *g* is the fixed codebook gain.

3) Energy per band analysis

[0046] Before any modification to the time-domain excitation, the converter/modifier 107 comprises a gain calculator 208-209-210 itself including a sub-calculator 209 to compute the energy per band E_b of the frequency-domain excitation and keeps the computed energy per band E_b in memory for energy adjustment after the excitation spectrum reshaping. For a 12.8 kHz sampling frequency, the energy can be computed by the sub-calculator 209 as follow :

$$E_b(i) = \sqrt{\sum_{j=C_{Bb}(i)}^{j=C_{Bb}(i)+B_b(i)} f_e(j)^2} \qquad (16)$$

where C_{Bb} represents the cumulative frequency bins per band and B_b the number of bins per frequency band defined as:

$$B_b = \{4, 4, 4, 4, 4, 5, 6, 6, 6, 8, 8, 10, 11, 13, 15, 18, 22, 16, 16, 20, 20, 20, 16\}$$

$$C_{Bb} = \left\{\begin{matrix} 0, 8, 12, 16, 20, 25, 31, 37, 43, 51, 59, 69, 80, 93, \\ 108, 126, 148, 164, 180, 200, 220, 240 \end{matrix}\right\}$$

[0047] The low frequency bands may correspond to the critical audio bands as described in [Milan Jelinek and Philippe Gournay. PCT Patent application WO03102921A1, "A method and device for efficient frame erasure concealment in linear predictive based speech codecs"], but the frequency bands above 3700 Hz may be a little shorter to better match the possible spectral energy variation in those bands. Any other configuration of spectral bands is also possible.

4) Excitation modification for inactive and active unvoiced frames

a) Cut off frequency of the time-domain contribution versus noise fill

[0048] To achieve a transparent switching between the non-modified excitation and the modified excitation for inactive frames and active unvoiced frames, at least the lower frequencies of the time-domain excitation contribution are kept. The converter/modifier 107 comprises a cut-off frequency calculator 203 to determine a frequency where the time-domain contribution stop to be used, the cut-off frequency f_c, having a minimum value of 1.2 kHz. This means that the first 1.2 kHz of the decoded excitation is always kept and depending on the decoded pitch value from the time-domain excitation decoder 102, this cut-off frequency can be higher. The 8th harmonic is computed from the lowest pitch of all sub-frames and the time-domain contribution is kept up to this 8th harmonic. An estimate of the 8th harmonic is calculated as follows:

$$h_{8th} = \frac{(8\cdot F_S)}{\min\limits_{0\leq i<N_{sub}}(T(i))} \qquad (17)$$

where F_S = 12800 *Hz*, N_{sub} is the number of sub-frames and T is the decoded sub-frame pitch. For all i < N_b where N_b is the maximum frequency band included in frequency range L_f, a verification is made to find the band in which the 8th harmonic is located by searching for the highest band for which the following inequality is still verified:

$$\left(h_{8^{th}} \geq L_f(i)\right) \qquad (18)$$

where L_f is defined as:

$$L_f = \begin{Bmatrix} 175, 275, 375, 475, 600, 750, 900, 1050, 1250, 1450, 1700, 1975, \\ 2300, 2675, 3125, 3675, 4075, 4475, 4975, 5475, 5975, 6375 \end{Bmatrix}$$

[0049] The index of that frequency band in L_f will be called $i_{8^{th}}$ and it indicates the frequency band where the 8th harmonic is likely to be located. The calculator cut-off frequency calculator 203 computes the final cut-off frequency f_{tc} as the higher frequency between 1.2 kHz and the last frequency of the frequency band in which the 8th harmonic is likely to be located (L_f ($i_{8^{th}}$)), using the following relation:

$$f_{tc} = \max\left(L_f\left(i_{8^{th}}\right), 1.2\text{ kHz}\right) \qquad (19)$$

b) Normalization and noise fill

[0050] The converter/modifier 107 further comprises a zeroer 204 that zeroes the frequency bins of the frequency bands above the cut-off frequency f_c.

[0051] For inactive frames and active unvoiced frames, a normalizer 205 of the converter/modifier 107 normalizes the frequency bins below f_c of the frequency bands of the frequency representation of the time-domain CELP excitation f_e between [0, 4] using the following relation:

$$f_{eN}(j) = \left\{ \begin{array}{ll} \dfrac{4\cdot f_e(j)}{\max\limits_{0\leq i<f_c}\left(\left|f_e(i)\right|\right)}, & for \quad 0\leq j<f_c \end{array} \right. \qquad (20)$$

$$\lfloor 0 \qquad\qquad , \quad for\ f_c \le j < 256$$

[0052] Then, the converter/modifier 107 comprises a random noise generator 206 to generate random noise and a simple noise fill is performed through an adder 207 to add noise over all the frequency bins at a constant level. The function describing the noise addition is defined below as:

for

$$j = 0 : L\text{-}1$$

$$f'_{eN}(j) = f_{eN}(j) + 0.75 \cdot rand(\ ) \qquad (21)$$

where r_{and} is a random number generator which is limited between -1 to 1.

c) Energy per band analysis of the modified excitation spectrum

[0053] Sub-calculator 208 of the gain calculator 208-209-210 determines the energy per band after the spectrum reshaping E_b' using the same method as described in above section 3.

d) Energy matching

[0054] For inactive frames and active unvoiced frames, the energy matching consists only in adjusting the energy per band after the excitation spectrum modification to its initial value. For each band *i*, sub-calculator 210 of the gain calculator 208-209-210 determines a matching gain G_b to apply to all bins in the frequency band for matching the energy as follows:

$$G_b(i) = \frac{E_b(i)}{E_b{}'(i)} \qquad (22)$$

where $E_b(i)$ is the energy per band before excitation spectrum modification as determined in sub-calculator 209 using the method of above section 3 and $E'_b(i)$ is the energy per band after excitation spectrum modification as calculated in sub-calculator 208. For a specific band *i*, the modified (de-normalized) frequency-domain excitation

$$f'_{edN}$$

as determined in sub-calculator 210 can be written as:

for

$$C_{Bb}(i) \le j < C_{Bb}(i) + B_b(i)$$

$$f'_{edN}(j) = G_b(i) \cdot f'_{eN}(j) \qquad (23)$$

where C_{Bb} and B_b are defined in above section 3.

5) Excitation modification for generic audio frames

a) Normalization and noise fill

[0055] Reference will now be made to Figure 3. For generic audio frames as determined by the classifier portion 301, the normalization is slightly different and performed by a normalizer 302. First the normalization factor N_f changes from band to band, using a higher value for low frequency bands and a lower value for high frequency bands. The idea is to allow for higher amplitude in the low frequency bands where the location of the pulses is more accurate and lower amplitude in the higher frequency bands where the location of the pulses is less accurate. In this illustrative example, the varying normalization factor N_f by frequency band is defined as :

$N_f = \{16, 16, 16, 16, 16, 16, 16, 12, 12, 12, 12,8, 8, 8, 8, 8, 4, 4, 2, 2, 1, 1, 1\}$

[0056] For a specific frequency band *i*, the normalization of the frequency representation of the time-domain excitation (frequency-domain excitation) f_e of generic audio frames can be described as follow:

$$f_{eN}(j) = \frac{N_f(i) \cdot f_e(j)}{\max\limits_{k=C_{Bb}(i)}^{C_{Bb}(i)+B_b(i)} \left(\left|f_e(k)\right|\right)} \quad , \quad for \quad C_{Bb}(i) \le j < C_{Bb}(i) + B_b(i) \qquad (24)$$

where B_b is the number of bins per frequency band, the cumulative frequency bins per bands is C_{Bb} and $f_{eN}(j)$ is the normalized frequency-domain excitation. B_b and C_{Bb} are described in the above section 3.

[0057] Furthermore, the normalizer 302 comprises a zeroer (not shown) to zero all the frequency bins below a fraction Z_f of the maximum value of $f_{eN}(j)$ in each frequency band to obtain $f'_{eN}(j)$:

$$f'_{eN}(j) = \left. \begin{cases} 0 & if\left(f_{eN}(j) < Z_f(i)\right) \\ f_{eN}(j) & otherwise \end{cases} \right|_{for\ C_{Bb}(i) \le j < C_{Bb}(i) + \mathrm{B}_b(i)} \qquad (25)$$

where Z_f can be represented as:

$Z_f = \{1, 1, 1, 1, 1, 1, 1, 1,1, 1, 1,1, 1, 1, 1, 1, 1, 1, 1, 1, 0.5, 0.5, 0.5\}$

[0058] A more aggressive zeroing can be performed by increasing the value of the vector Z_f, if it is desired to increase the peakyness of the spectrum.

b) Energy per band analysis of the modified excitation spectrum

[0059] Calculator portion 303 of a gain calculator 303-304-306 determines the energy per band after spectrum reshaping E_b' using the same method as described in above section 3.

c) Energy matching

[0060] Figure 3 shows the gain calculator 303-304-306 and Figure 4 describes in more detail calculator portion 306 of this gain calculator.

[0061] For generic audio frames, the energy matching is trickier since it aims at increasing the spectral dynamic as well. For each frequency band *i*, a sub-calculator 413 of calculator portion 306 of the gain calculator 303-304-306 computes an estimated gain G_e defined similarly as in equation (22):

$$G_e(i) = \frac{E_b(i)}{E_b{}'(i)} \qquad (26)$$

where $E_b(i)$ is the energy per band before excitation spectrum modification as determined in calculator portion 304 using the method as described in above section 3, and $E'_b(i)$ is the energy per band after excitation spectrum modification as calculated in calculator portion 303.

[0062] A sub-calculator 414 of the calculator portion 306 applies the gain G_e to the first 400 Hz (or first 4 bands) of the normalized frequency-domain excitation f'_{eN} from the normalizer 302 and spectrum splitter 401-420 to provide a modified (de-normalized) frequency-domain excitation f'_{edN} using the following relation:

$$f'_{edN}(j) = G_e(i) \cdot f'_{eN}(j), \quad \text{for } C_{Bb}(i) \le j < C_{Bb}(i) + B_b(i)|_{0\le i<4} \quad (27)$$

[0063] A finder 404 determines the maximum value max

$$\max_{a\le j<b}(|f_{eN}(j)|)$$

per band *i* above 400 Hz, where $a = C_{Bb}(i)$ and $b = C_{Bb}(i) + B_b(i)$ are defined in above section 3.

[0064] For the frequency bands comprised between 400 Hz and 2 kHz (bands 4 to 12) of the normalized frequency-domain excitation (see module 420 and 450), if the normalized frequency-domain excitation in a frequency bin

$$f'_{eN}(j) \ge 0.86 \max_{a\le j<b}(|f_{eN}(j)|)$$

(see module 451), an amplifier 402 amplifies the gain G_o from the sub-calculator 413 by a factor 1.1 as shown in the upper line of Equation (28). A sub-calculator 403 applies the amplified gain from amplifier 402 to the normalized spectral excitation f'_{eN} in the frequency bin according to the first line of Equation (28) to obtain the modified (de-normalized) frequency-domain excitation f'_{edN}.

[0065] Again for the frequency bands comprised between 400 Hz and 2 kHz (bands 4 to 12) of the normalized frequency-domain excitation (see module 420 and 450), if the normalized frequency-domain excitation in a frequency bin

$$f'_{eN}(j) < 0.86 \max_{a\le j<b}(|f_{eN}(j)|)$$

(see module 451), an attenuator 405 attenuates the gain G_e from the sub-calculator 413 by a factor 0.86 as shown in the lower line of Equation (28). A sub-calculator 406 applies the attenuated gain from attenuator 405 to the normalized spectral excitation f'_{eN} in the frequency bin according to the lower line of Equation (28) to obtain the modified (de-normalized) frequency-domain excitation f'_{edN}.

[0066] To summarize, the modified (de-normalized) spectral excitation f'_{edN} is given as follows:

$$f'_{edN}(j) = \begin{cases} 1.1 \cdot G_e(i) \cdot f'_{eN}(j), & \text{if} \quad f'_{eN}(j) \geq 0.86 \cdot \max\limits_{a \leq j < b}\left(\left|f_{eN}(j)\right|\right) \\ 0.86 \cdot G_e(i) \cdot f'_{eN}(j), & \text{if} \quad f'_{eN}(j) < 0.86 \cdot \max\limits_{a \leq j < b}\left(\left|f_{eN}(j)\right|\right) \end{cases} \tag{28}$$

[0067] Finally for higher parts of the spectrum, in this example the frequency bands above 2 kHz (bands > 12) of the normalized frequency-domain excitation (see module 420 and 450), if the normalized frequency-domain excitation in a frequency bin

$f'_{eN}(j) \geq 0.86 \max\limits_{a \leq j < b}(|f_{eN}(j)|)$

(see module 452), a tilt which is a function of the frequency band i and which can also be a function of the bit rate is added to the gain G_e to compensate for the too low energy estimation of the LPC filter. The value of the tilt per frequency band $\delta(i)$ is formulated as:

$$\delta(i) = 1.5 \cdot G_e(i) \cdot \frac{(j-12)}{32} \tag{29}$$

[0068] The tilt is calculated by tilt calculator 407-408 and is applied to the normalized frequency-domain excitation f'_{eN} by frequency bin according to the upper line of Equation (30) by a sub-calculator 409 to obtain the modified (denormalized) frequency-domain excitation f'_{odN}.

[0069] Again for higher parts of the spectrum, in this illustrative example the frequency bands above 2 kHz (bands > 12) of the normalized frequency-domain excitation (see module 420 and 450), if the normalized frequency-domain excitation in a frequency bin

$f'_{eN}(j) < 0.86 \max\limits_{a \leq j < b}(|f_{eN}(j)|)$

(see module 452), an attenuator 410 calculates an attenuation gain

$[f'_{eN}(j)/\max\limits_{a \leq j < b}(|f_{eN}(j)|)]^2$

applied to the normalized spectral excitation f'_{eN} by frequency bin according to the lower line of Equation (30) by a sub-calculator 411 to obtain the modified (de-normalized) frequency-domain excitation f'_{edN}.

[0070] To summarize, the denormalized spectral excitation f'_{edN} is determined as follows:

$$f'_{edN}(j) = \begin{cases} \delta(i) \cdot f'_{eN}(j), & \text{if} \quad f'_{eN}(j) \geq 0.86 \cdot \max\limits_{a \leq j < b}\left(\left|f_{eN}(j)\right|\right) \\ \left(\dfrac{f'_{eN}(j)}{\max\limits_{a \leq j < b}\left(\left|f_{eN}(j)\right|\right)}\right)^2 \cdot f'_{eN}(j), & \textit{otherwise} \end{cases} \tag{30}$$

where *a* and *b* are described herein above. It is also possible to further increase the gain applied to the latest bands, where the energy matching of the LPC is the worst.

6) Inverse frequency transform

[0071] A combiner 453 combines the contributions to the modified (denormalized) frequency-domain excitation f'_{edN} from the sub-calculators 414, 403, 406, 409 and 411 to form the complete modified (de-normalized) frequency-domain excitation f'_{edN} .

[0072] After the frequency domain processing is completed, an inverse frequency-time transform 202 is applied to the modified (de-normalized) frequency-domain excitation f'_{edN} from combiner 453 to find the time-domain modified excitation. In this illustrative embodiment, the frequency-to-time conversion is achieved with the inverse of the same type II DCT as used for the time-to-frequency conversion giving a resolution of 25 Hz. Again, any other transforms can be used. The modified time-domain excitation
e'_{td}
is obtained as below:

$$e'_{td}(k) = \begin{cases} \sqrt{\frac{1}{L}} \cdot \sum_{n=0}^{L-1} f'_{edN}(n), & k = 0 \\ \sqrt{\frac{2}{L}} \cdot \sum_{n=0}^{L-1} f'_{edN}(n) \cdot \cos\left(\frac{\pi}{L}\left(n + \frac{1}{2}\right)k\right), & 1 \le k \le L-1 \end{cases} \qquad (31)$$

where
$f'_{edN}(n)$
is the frequency representation of the modified excitation, and *L* is the frame length. In this illustrative example, the frame length is 256 samples for a corresponding inner sampling frequency of 12.8 kHz (AMR-WB).

7) Synthesis filtering and overwriting the current CELP synthesis

[0073] Once the excitation modification is completed, the modified excitation is processed through the synthesis filter 108 to obtain a modified synthesis for the current frame. The overwriter 110 uses this modified synthesis to overwrite the decoded synthesis thus to increase the perceptual quality.

[0074] Final de-emphasis and resampling to 16 kHz can then be performed in de-emphasis filter and resampler 112.

REFERENCES CITED IN THE DESCRIPTION

Cited references

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.

Patent documents cited in the description

- US6240386B [0004]
- WO03102921A1 [0019]
- WO2007073604A1 [0019]
- WO03102921A [0047]

Non-patent literature cited in the description

- **J. D. JOHNSTON**Transform coding of audio signal using perceptual noise criteriaIEEE J. Select. Areas Commun., 1988, vol. 6, 314-323 [0014]

Patentkrav

1. Anordning til modificering, under afkodning af et lydsignal, af en syntese af en tidsdomæneexcitation afkodet af en tidsdomæneafkoder(102), og som omfatter:

en klassifikator (104, 105, 106) konfigureret til at klassificere syntesen af den afkodede tidsdomæneexcitation i én af et antal kategorier;

en første konverter (107, 201) konfigureret til at konvertere den afkodede tidsdomæneexcitation til en frekvensdomæneexcitation;

en modifikator (107, 203, 204, 205, 206, 207, 208, 209,210) konfigureret til at modificere frekvensdomæneexcitationen afhængigt af den kategori, hvori syntesen af den afkodede tidsdomæneexcitation klassificeres, ved hjælp af klassifikatoren (104, 105, 106);

en anden konverter (107, 202) konfigureret til at konvertere den modificerede frekvensdomæneexcitation til en modificeret tidsdomæneexcitation;

et syntesefilter (108) konfigureret til at få tilført den modificerede tidsdomæneexcitation for at frembringe en modificeret syntese af den afkodede tidsdomæneexcitation;

hvor modifikatoren (107, 203, 204, 205, 206, 207, 208, 209, 210) omfatter:

en kalkulator (203) konfigureret til at beregne en afskæringsfrekvens, hvor et tidsdomæneexcitationsbidrag ophører med at blive anvendt,

hvor afskæringsfrekvensen har en minimumværdi på 1,2 kHz;

en nulstiller (204) konfigureret til at nulstille frekvensdomæneexcitationen over afskæringsfrekvensen;

en normalisator (205) af frekvensdomæneexcitationen under afskæringsfrekvensen;

en generator (206) af tilfældig støj konfigureret til at generere en tilfældig støj og

en tilføjer (207) konfigureret til at tilføje den tilfældige støj til frekvensdomæneexcitationen nulstillet over afskæringsfrekvensen og normaliseret under afskæringsfrekvensen.

2. Anordning til modificering af en syntese af en tidsdomæneexcitation ifølge krav 1, hvor afskæringsfrekvensen beregnes fra en funktion af en afkodet pitch for tidsdomæneexcitationen.

3. Anordning til modificering af en syntese af en tidsdomæneexcitation ifølge et hvilket som helst af kravene 1 til 2, hvor beregningen af afskæringsfrekvensen omfatter beregningen af et estimat over excitationens ottende harmoniske.

4. Anordning til modificering af en syntese af en tidsdomæneexcitation ifølge et hvilket som helst af kravene 1 til 3, hvor frekvensdomæneexcitationen divideres i frekvensbånd, og hvor modifikatoren (107, 203, 204, 205, 206, 207, 208, 209, 210) endvidere omfatter:

en kalkulator (208, 209, 210) af en matchende forstærkning til justering af energien pr. bånd efter excitationsspektrummodifikationen til dens oprindelige værdi,

hvor kalkulatoren af den matchende forstærkning er konfigureret til at beregne den matchende forstærkning ved hjælp af en energi pr. bånd af frekvensdomæneexcitationen før modifikation og en energi pr. bånd af frekvensdomæneexcitationen efter modifikation.

5. Anordning til modificering af en syntese af en tidsdomæneexcitation ifølge et hvilket som helst af kravene 1 til 4, hvor klassifikatoren (104, 105, 106) klassificerer syntesen af den afkodede tidsdomæneexcitation som inaktiv eller aktiv stemmeløs.

6. Anordning til modificering af en syntese af en tidsdomæneexcitation ifølge et hvilket som helst af kravene 1 til 5, og som omfatter en udjævner (111) af syntesefilteret, når syntesen af den afkodede tidsdomæneexcitation klassificeres som en given én af kategorierne ved hjælp af klassifikatoren.

7. Anordning til afkodning af et lydsignal kodet af kodningsparametre, og som omfatter:

en afkoder (102) af en tidsdomæneexcitation som reaktion på lydsignalkodningsparametrene;

et syntesefilter (103), der reagerer på den afkodede tidsdomæneexcitation for frembringelse af en syntese af tidsdomæneexcitationen; og en anordning ifølge et hvilket som helst af kravene 1 til 6 til modificering af syntesen af tidsdomæneexcitationen.

8. Fremgangsmåde til modificering, under afkodning af et lydsignal, af en syntese af en tidsdomæneexcitation afkodet af en tidsdomæneafkoder(102), og som omfatter:

klassificering (104, 105, 106) af syntesen af den afkodede tidsdomæneexcitation i én af et antal kategorier;

konvertering (107, 201) af den afkodede tidsdomæneexcitation til en frekvensdomæneexcitation;

modificering (107, 203, 204, 205, 206, 207, 208, 209, 210) af frekvensdomæneexcitationen afhængigt af den kategori, hvori syntesen af den afkodede tidsdomæneexcitation klassificeres;

konvertering (107, 202) af den modificerede frekvensdomæneexcitation til en modificeret tidsdomæneexcitation;

syntetisering (108) af den modificerede tidsdomæneexcitation for at frembringe en modificeret syntese af den afkodede tidsdomæneexcitation;

hvor trinnet med modificering (107, 203, 204, 205, 206, 207, 208, 209, 210) af frekvensdomæneexcitationen omfatter:

beregning (203) af en afskæringsfrekvens, hvor et tidsdomæneexcitationsbidrag ophører med at blive anvendt,

hvor afskæringsfrekvensen har en minimumværdi på 1,2 kHz;

nulstilling (204) af frekvensdomæneexcitationen over afskæringsfrekvensen;

normalisering (205) af frekvensdomæneexcitationen under afskæringsfrekvensen;

generering (206) af en tilfældig støj og

tilføjelse (207) af den tilfældige støj til frekvensdomæneexcitationen nulstillet over afskæringsfrekvensen og normaliseret under afskæringsfrekvensen.

9. Fremgangsmåde til modificering af en syntese af en tidsdomæneexcitation ifølge krav 8, hvor afskæringsfrekvensen beregnes fra en funktion af en afkodet pitch for the tidsdomæneexcitation.

10. Fremgangsmåde til modificering af en syntese af en tidsdomæneexcitation ifølge et hvilket som helst af kravene 8 til 9, hvor beregningen af afskæringsfrekvensen omfatter: beregnet af et estimat over excitationens ottende harmoniske.

11. Fremgangsmåde til modificering af en syntese af en tidsdomæneexcitation ifølge et hvilket som helst af kravene 8 til 10, hvor frekvensdomæneexcitationen divideres i frekvensbånd, og hvor trinnet med modificering af frekvensdomæneexcitationen endvidere omfatter:
beregning af en matchende forstærkning til justering af energien pr. bånd efter excitationsspektrummodifikationen til dens oprindelige værdi ved hjælp af en energi pr. bånd af frekvensdomæneexcitationen før modifikation og en energi pr. bånd af frekvensdomæneexcitationen efter modifikation.

12. Fremgangsmåde til modificering af en syntese af en tidsdomæneexcitation ifølge et hvilket som helst af kravene 8 til 11, hvor trinnet med klassificering af syntesen af den afkodede tidsdomæneexcitation i én af et antal kategorier omfatter
klassificering af syntesen af den afkodede tidsdomæneexcitation som inaktiv eller aktiv stemmeløs.

13. Fremgangsmåde til modificering af en syntese af en tidsdomæneexcitation ifølge et hvilket som helst af kravene 8 til 12, og som endvidere omfatter
udjævning (111) af et syntesefilter, der udfører syntesen af den modificerede tidsdomæneexcitation, når syntesen af den afkodede tidsdomæneexcitation klassificeres som en given én af kategorierne.

14. Fremgangsmåde til afkodning af et lydsignal kodet af kodningsparametre, og som omfatter:

afkodning (102) af en tidsdomæneexcitation som reaktion på lydsignalkodningsparametrene;

syntetisering (103) af den afkodede tidsdomæneexcitation for at frembringe en syntese af tidsdomæneexcitationen og

en fremgangsmåde ifølge et hvilket som helst af kravene 8 til 13 til modificering af syntesen af tidsdomæneexcitationen.

DRAWINGS

Drawing

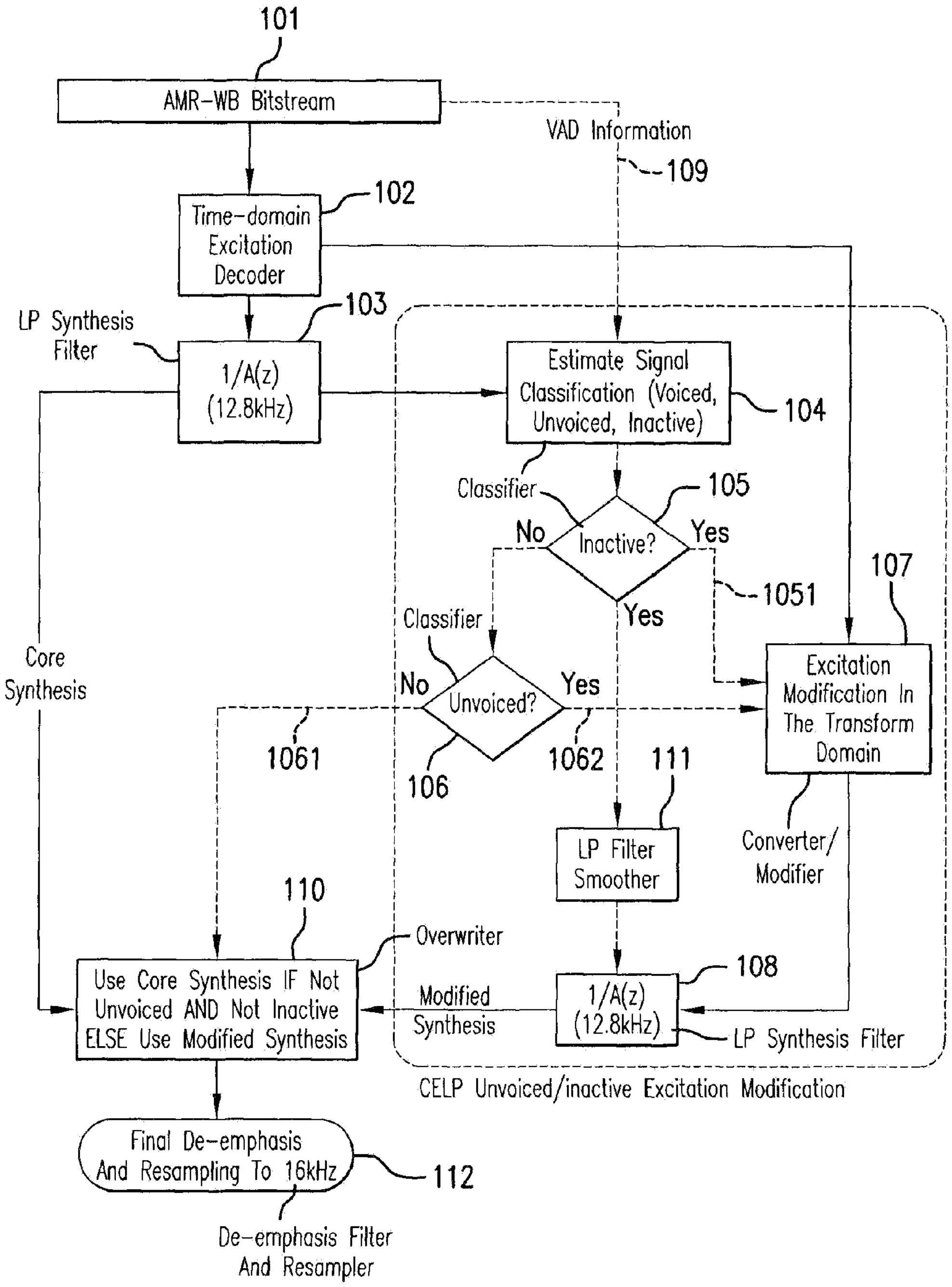

101
AMR-WB Bitstream
VAD Information
109
102
Time-domain Excitation Decoder
103
LP Synthesis Filter
1/A(z) (12.8kHz)
Estimate Signal Classification (Voiced, Unvoiced, Inactive)
104
Classifier
105
Inactive?
No
Yes
Yes
1051
107
Classifier
Core Synthesis
No
Unvoiced?
Yes
1061
106
1062
111
Excitation Modification In The Transform Domain
Converter/ Modifier
LP Filter Smoother
110
Overwriter
108
Use Core Synthesis IF Not Unvoiced AND Not Inactive ELSE Use Modified Synthesis
Modified Synthesis
1/A(z) (12.8kHz)
LP Synthesis Filter
CELP Unvoiced/inactive Excitation Modification
Final De-emphasis And Resampling To 16kHz
112
De-emphasis Filter And Resampler

FIG.1

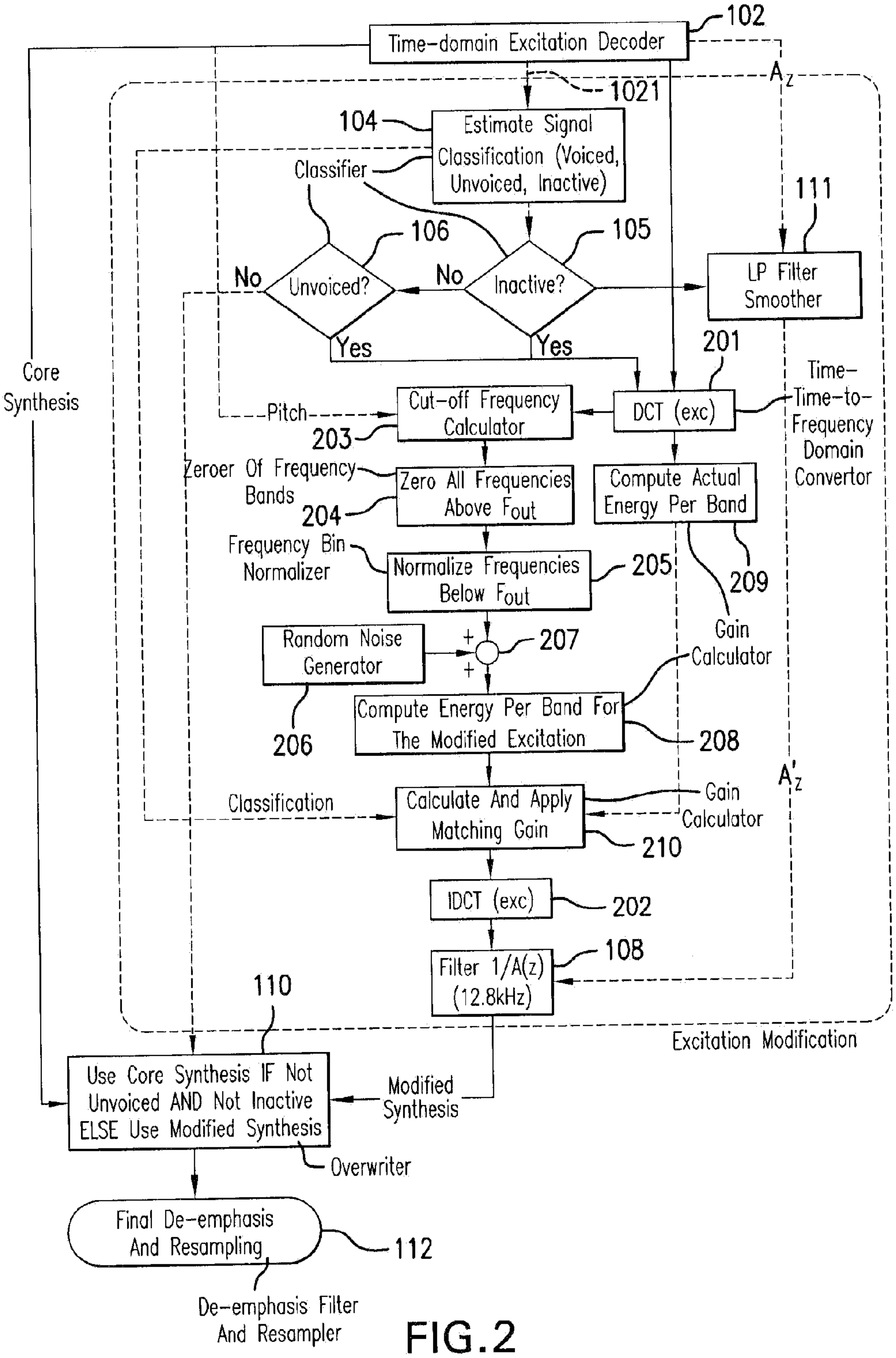
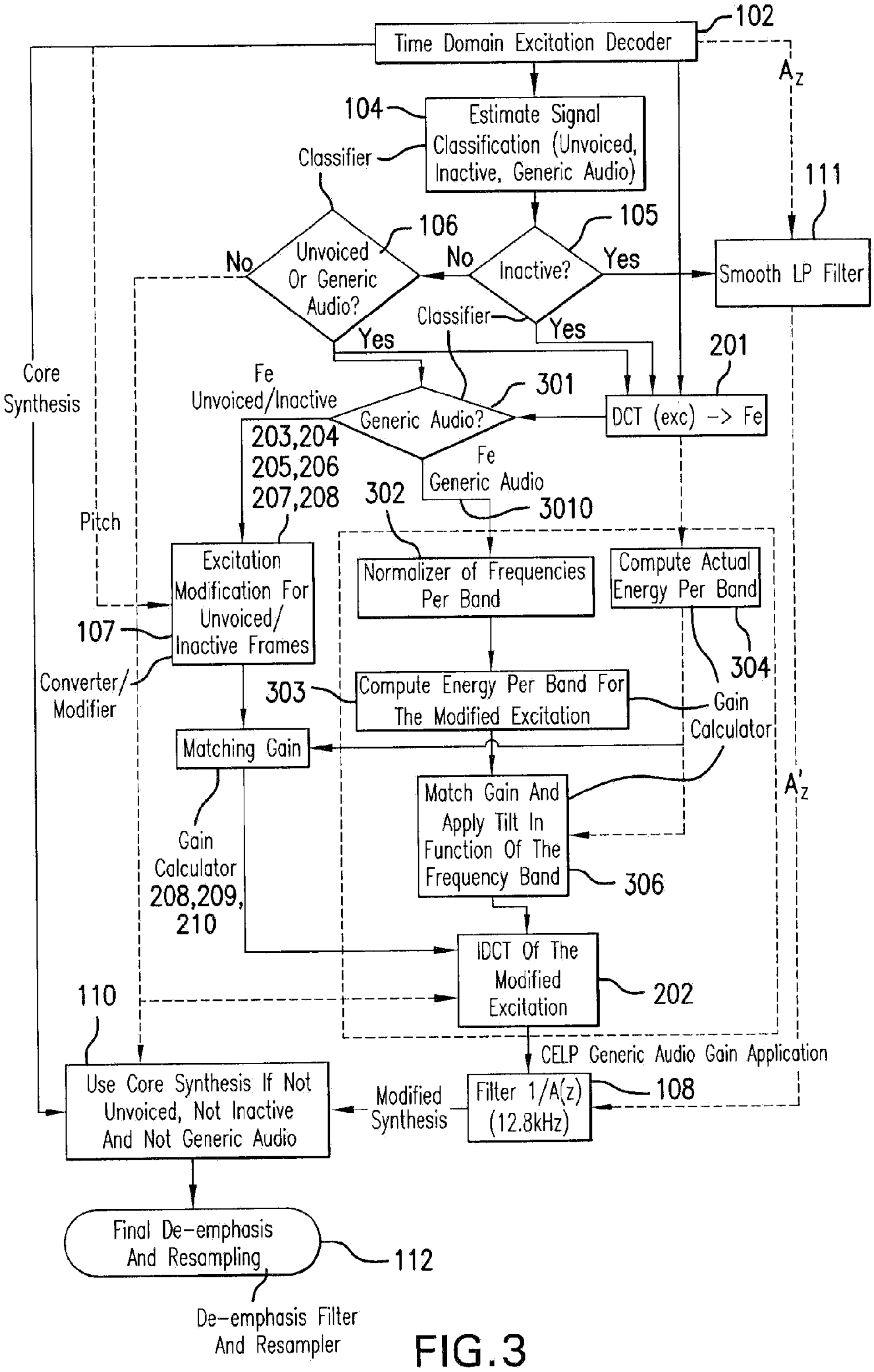

102
Time-domain Excitation Decoder
A_z
1021
104
Estimate Signal Classification (Voiced, Unvoiced, Inactive)
Classifier
111
106
105
No
Unvoiced?
No
Inactive?
LP Filter Smoother
Yes
Yes
Core Synthesis
201
Time-Time-to-Frequency Domain Convertor
Pitch
Cut-off Frequency Calculator
203
DCT (exc)
Zeroer Of Frequency Bands
Zero All Frequencies Above Fout
204
Compute Actual Energy Per Band
Frequency Bin Normalizer
Normalize Frequencies Below Fout
205
209
Random Noise Generator
+
+
207
Gain Calculator
Compute Energy Per Band For The Modified Excitation
206
208
A'_z
Classification
Calculate And Apply Matching Gain
Gain Calculator
210
IDCT (exc)
202
Filter 1/A(z) (12.8kHz)
108
110
Excitation Modification
Use Core Synthesis IF Not Unvoiced AND Not Inactive ELSE Use Modified Synthesis
Modified Synthesis
Overwriter
Final De-emphasis And Resampling
112
De-emphasis Filter And Resampler

FIG.2



102
Time Domain Excitation Decoder
A_Z
104
Estimate Signal Classification (Unvoiced, Inactive, Generic Audio)
Classifier
111
106
105
No
Unvoiced Or Generic Audio?
No
Inactive?
Yes
Smooth LP Filter
Classifier
Yes
Yes
201
Core Synthesis
Fe Unvoiced/Inactive
301
Generic Audio?
DCT (exc) -> Fe
203,204
205,206
207,208
Fe Generic Audio
302
3010
Pitch
Excitation Modification For Unvoiced/ Inactive Frames
Normalizer of Frequencies Per Band
Compute Actual Energy Per Band
107
304
Converter/ Modifier
303
Compute Energy Per Band For The Modified Excitation
Gain Calculator
Matching Gain
A'_Z
Match Gain And Apply Tilt In Function Of The Frequency Band
Gain Calculator
208,209,
210
306
IDCT Of The Modified Excitation
202
110
CELP Generic Audio Gain Application
Use Core Synthesis If Not Unvoiced, Not Inactive And Not Generic Audio
Modified Synthesis
Filter 1/A(z) (12.8kHz)
108
Final De-emphasis And Resampling
112
De-emphasis Filter And Resampler

FIG.3

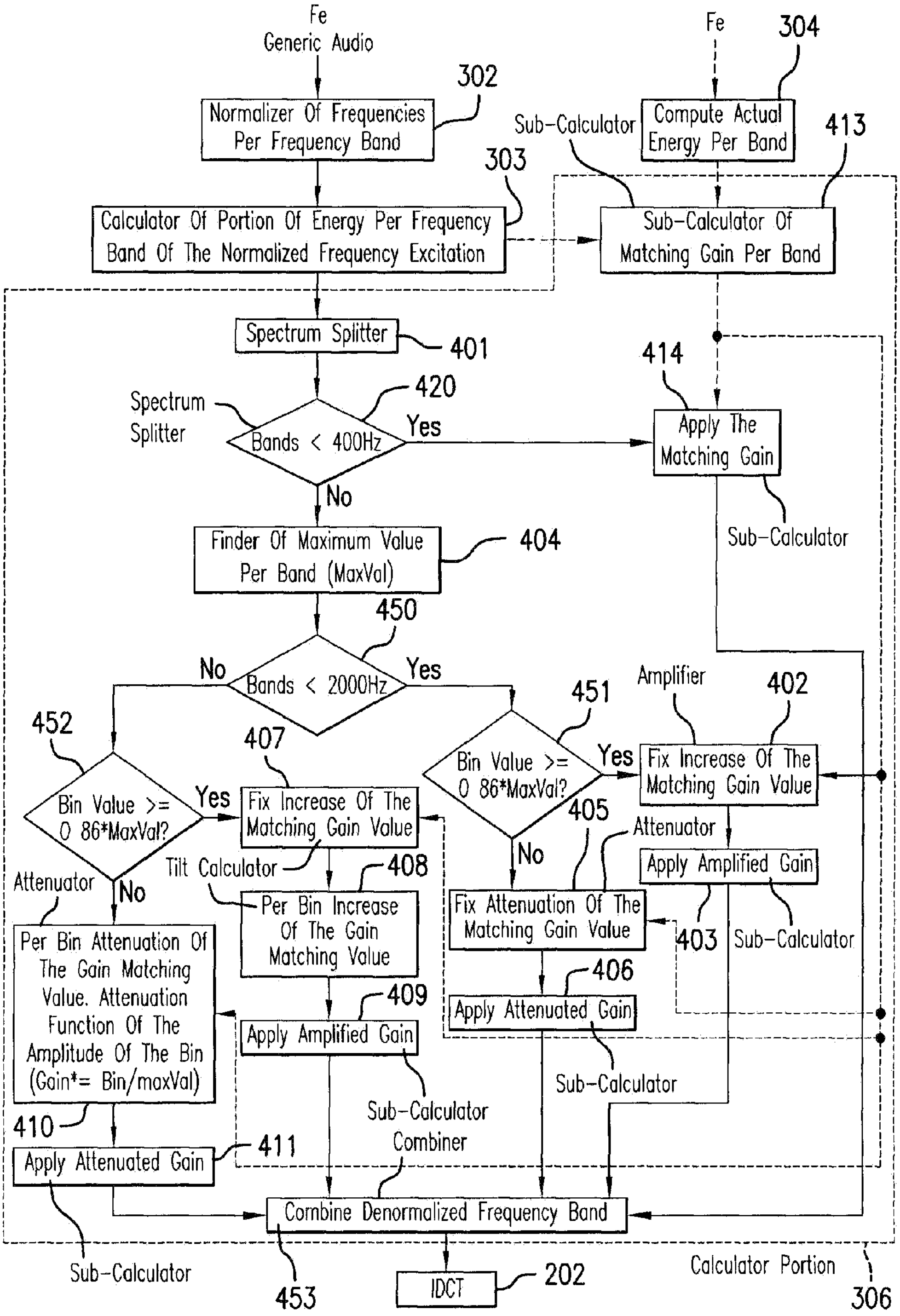

Fe
Generic Audio
Normalizer Of Frequencies Per Frequency Band
302
Calculator Of Portion Of Energy Per Frequency Band Of The Normalized Frequency Excitation
303
Fe
304
Compute Actual Energy Per Band
Sub-Calculator
413
Sub-Calculator Of Matching Gain Per Band
Spectrum Splitter
401
Spectrum Splitter
420
Bands < 400Hz
Yes
No
414
Apply The Matching Gain
Sub-Calculator
Finder Of Maximum Value Per Band (MaxVal)
404
450
Bands < 2000Hz
No
Yes
452
Bin Value >= 0 86*MaxVal?
Yes
No
407
Fix Increase Of The Matching Gain Value
Tilt Calculator
408
Per Bin Increase Of The Gain Matching Value
409
Apply Amplified Gain
Attenuator
Per Bin Attenuation Of The Gain Matching Value. Attenuation Function Of The Amplitude Of The Bin (Gain*= Bin/maxVal)
410
Apply Attenuated Gain
411
451
Bin Value >= 0 86*MaxVal?
Yes
No
405
Amplifier
402
Fix Increase Of The Matching Gain Value
Attenuator
Apply Amplified Gain
Fix Attenuation Of The Matching Gain Value
403
Sub-Calculator
406
Apply Attenuated Gain
Sub-Calculator
Sub-Calculator
Combiner
Combine Denormalized Frequency Band
Sub-Calculator
453
IDCT
202
Calculator Portion
306

FIG.4