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Vaillancourt et al.

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(54) **NON-SPEECH CONTENT FOR LOW RATE CELP DECODER**

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G10L 19/20	(2013.01)
G10L 19/26	(2013.01)
G10L 19/08	(2013.01)

(Continued)

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CPC ***H03G 3/20*** (2013.01); ***G10L 19/20*** (2013.01); ***G10L 19/26*** (2013.01); *G10L 19/08* (2013.01); *G10L 19/22* (2013.01); *G10L 25/78* (2013.01); *G10L 25/81* (2013.01); *G10L 25/93* (2013.01)

(58) **Field of Classification Search**

CPC H03G 5/00; G10L 19/12; G10L 19/02

USPC 381/22, 98; 704/206, 500, E19.001, 219

See application file for complete search history.

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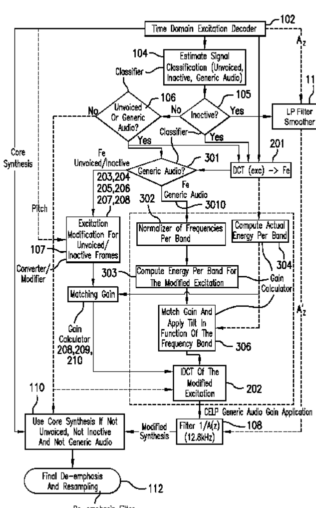
(57) **ABSTRACT**

A method and device for modifying a synthesis of a time-domain excitation decoded by a time-domain decoder, wherein the synthesis of the decoded time-domain excitation is classified into one of a number of categories. The decoded time-domain excitation is converted into a frequency-domain excitation, and the frequency-domain excitation is modified as a function of the category in which the synthesis of the decoded time-domain excitation is classified. The modified frequency-domain excitation is converted into a modified time-domain excitation, and a synthesis filter is supplied with the modified time-domain excitation to produce a modified synthesis of the decoded time-domain excitation.

32 Claims, 4 Drawing Sheets



(51) **Int. Cl.**

G10L 25/81	(2013.01)
G10L 25/78	(2013.01)
G10L 25/93	(2013.01)
G10L 19/22	(2013.01)

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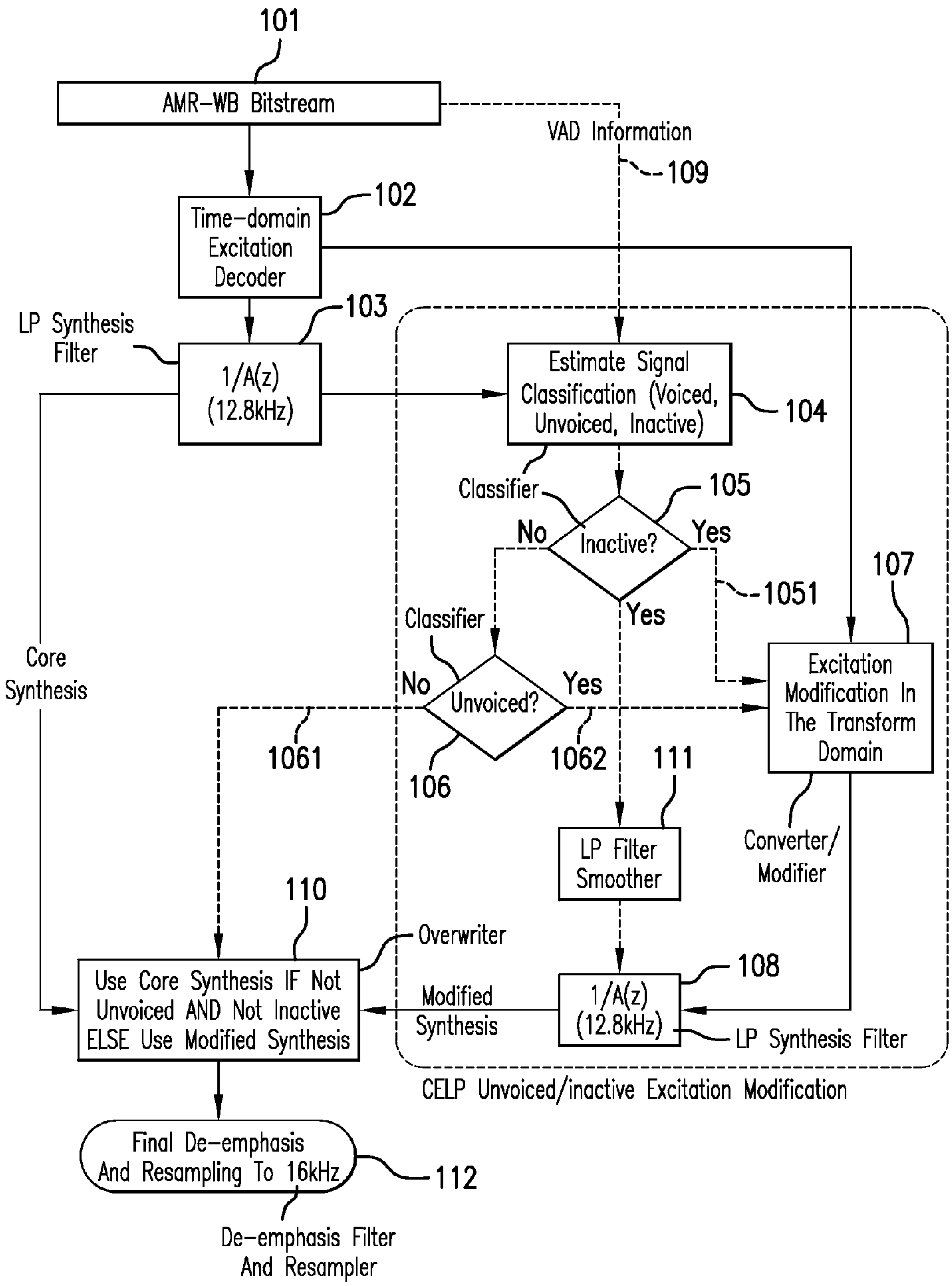
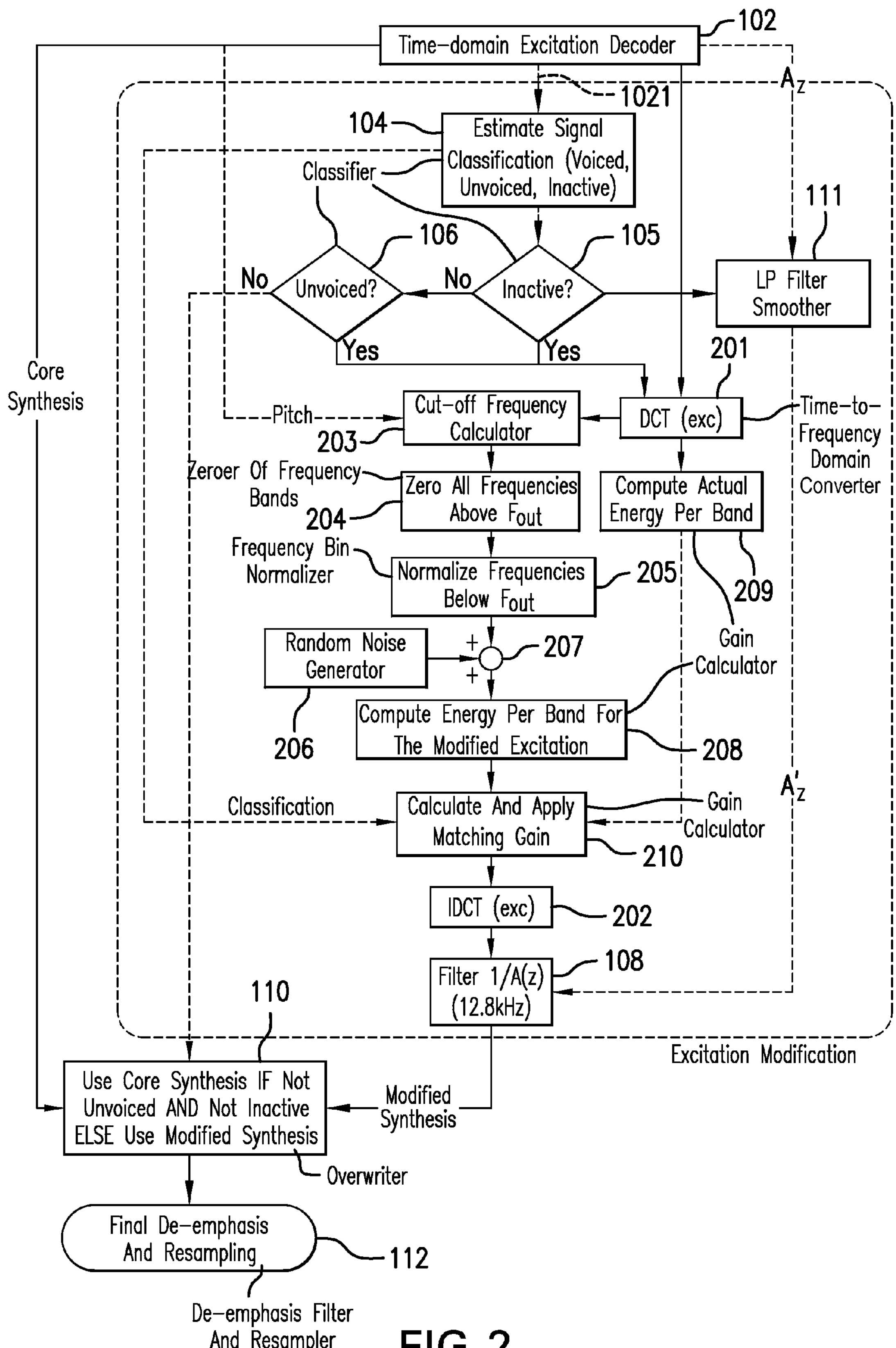

101
AMR-WB Bitstream
VAD Information
109
Time-domain Excitation Decoder
102
LP Synthesis Filter
103
1/A(z) (12.8kHz)
Estimate Signal Classification (Voiced, Unvoiced, Inactive)
104
Classifier
105
No
Inactive?
Yes
1051
Yes
107
Classifier
Core Synthesis
No
Unvoiced?
Yes
Excitation Modification In The Transform Domain
1061
1062
106
111
LP Filter Smoother
Converter/ Modifier
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
201
Core Synthesis
Pitch
Cut-off Frequency Calculator
DCT (exc)
Time-to-Frequency Domain Converter
203
Zeroer Of Frequency Bands
Zero All Frequencies Above Fout
Compute Actual Energy Per Band
204
Frequency Bin Normalizer
Normalize Frequencies Below Fout
205
209
Random Noise Generator
207
Gain Calculator
206
Compute Energy Per Band For The Modified Excitation
208
A'_z
Classification
Calculate And Apply Matching Gain
Gain Calculator
210
IDCT (exc)
202
108
Filter 1/A(z) (12.8kHz)
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

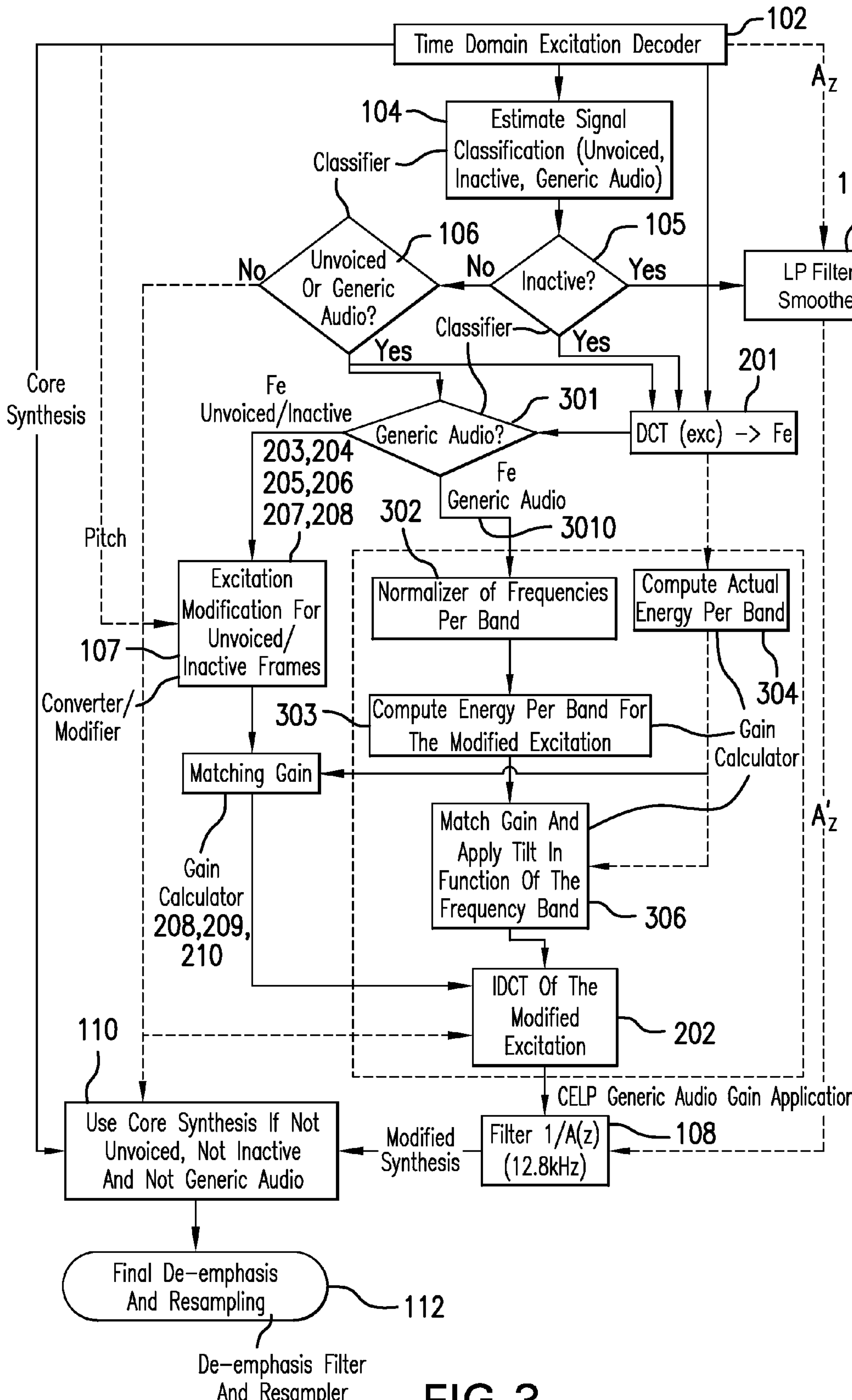

Time Domain Excitation Decoder
102
A_Z
104
Estimate Signal Classification (Unvoiced, Inactive, Generic Audio)
Classifier
111
106
105
Unvoiced Or Generic Audio?
No
No
Inactive?
Yes
LP Filter Smoother
Classifier
Yes
Yes
201
Core Synthesis
Fe Unvoiced/Inactive
301
Generic Audio?
DCT (exc) -> Fe
203,204
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
110
202
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

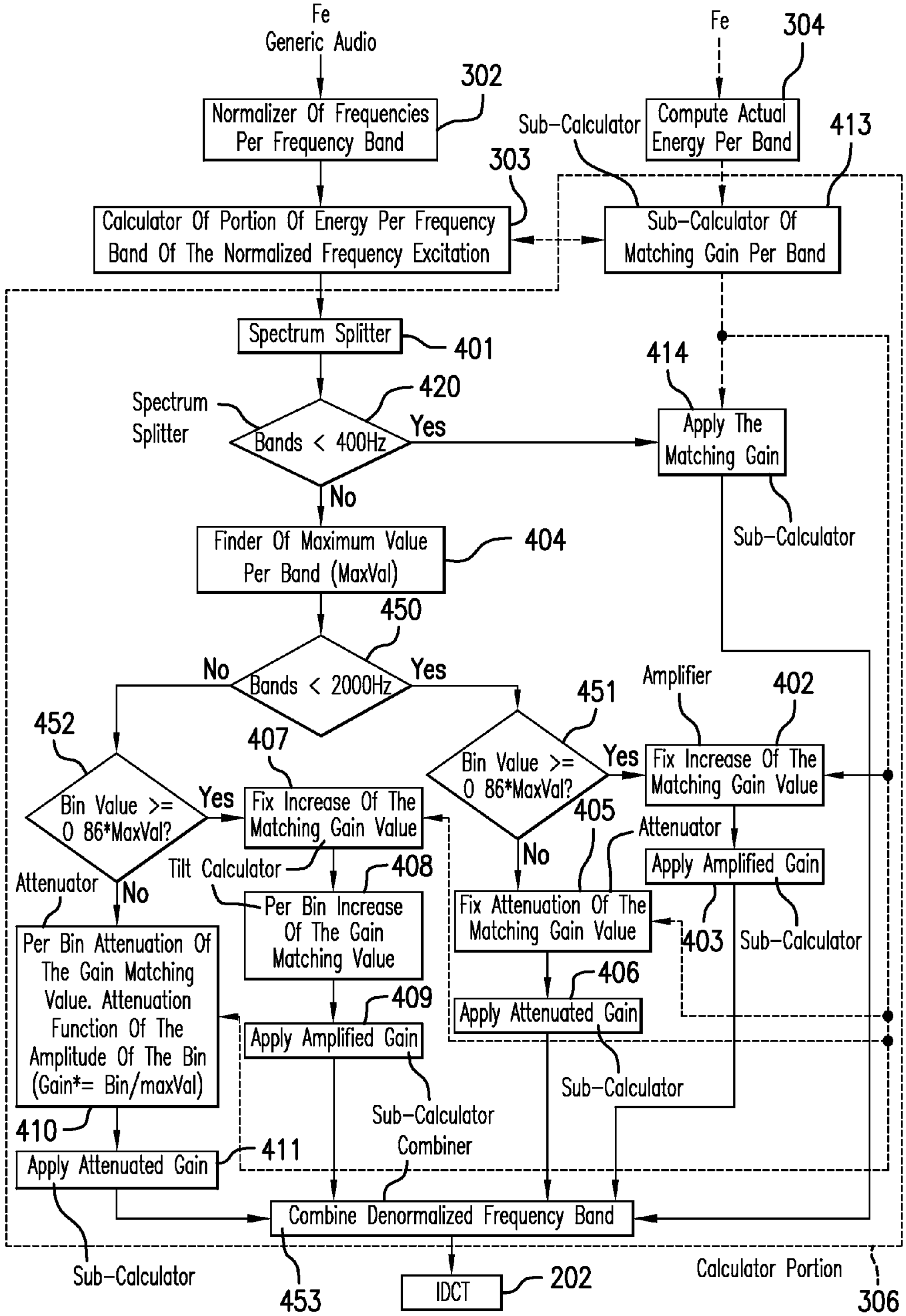

Fe
Generic Audio
Fe
304
302
Normalizer Of Frequencies Per Frequency Band
Sub-Calculator
Compute Actual Energy Per Band
413
303
Calculator Of Portion Of Energy Per Frequency Band Of The Normalized Frequency Excitation
Sub-Calculator Of Matching Gain Per Band
Spectrum Splitter
401
414
420
Spectrum Splitter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

FIG.4

NON-SPEECH CONTENT FOR LOW RATE CELP DECODER

CROSS-REFERENCE TO RELATED APPLICATIONS

This application claims priority to and the benefit of U.S. provisional patent application No. 61/555,246, filed on Nov. 3, 2011, the entire contents of which are hereby incorporated by reference herein.

FIELD

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

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.

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.

BRIEF DESCRIPTION OF THE DRAWINGS

In the appended drawings:

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

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

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

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

DESCRIPTION

According to a first aspect, the present disclosure is concerned with a device for modifying a synthesis of a time-domain excitation decoded by a time-domain decoder, comprising: a classifier of the synthesis of the decoded time-domain excitation into one of a number of categories; a converter of the decoded time-domain excitation into a frequency-domain excitation; a modifier of the frequency-domain excitation as a function of the category in which the synthesis of the decoded time-domain excitation is classified by the classifier; a converter of the modified frequency-domain excitation into a modified time-domain excitation; and a synthesis filter supplied with the modified time-domain excitation to produce a modified synthesis of the decoded time-domain excitation.

According to another aspect, the present disclosure provides a 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.

According to a third aspect, the present disclosure is concerned with a method for modifying a synthesis of a time-domain excitation decoded by a time-domain decoder, comprising: classifying the synthesis of the decoded time-domain excitation into one of a number of categories; converting the decoded time-domain excitation into a frequency-domain excitation; modifying the frequency-domain excitation as a function of the category in which the synthesis of the decoded time-domain excitation is classified; converting the modified frequency-domain excitation into a modified time-domain excitation; and synthesizing the modified time-domain excitation to produce a modified synthesis of the decoded time-domain excitation.

According to a further aspect, the present disclosure provides 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.

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.

The present disclosure proposes 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] of which the full content is incorporated herein by reference, 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.

Referring to FIG. **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.

The time-domain synthesis of the current frame from the LP synthesis filter **103** is processed through a classifier **104-105-106-301** (FIGS. **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.

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, February 1988], of which the full content is incorporated herein by reference, 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.

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.

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.

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 de-emphasizing filter and resampler **112**.

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 FIG. **2**.

1) Signal Classification

Referring to FIG. **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 FIG. **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"], of which the full contents are incorporated herein by reference, 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.

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:

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

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 3N/2 (N being the sub-frame size), T is set to the average pitch lag of the last two sub-frames.

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.

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:

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

where $x(n)=s_{out}(n)$ 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).

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

$$pc=|p_3+p_2-p_1-p_0| \tag{3}$$

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).

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}

$$E_s=E_f-E_{lt} \tag{4}$$

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

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

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).

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

$$E_{lt}=0.99E_{lt}+0.01E_f \tag{6}$$

The last parameter is the zero-crossing counter zc computed on one frame of the time-domain synthesis $s_{out}(n)$. 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.

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:

$$p^s=k_p \cdot p_x+c_p \tag{7}$$

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

The function of merit is defined as:

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

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

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 VOICED VOICED TRANSITION ARTIFICIAL ONSET GENERIC AUDIO SOUND	$f_m \geq 0.63$	VOICED
	$0.39 \leq f_m < 0.63$	VOICED TRANSITION
	$f_m < 0.39$	UNVOICED
UNVOICED TRANSITION UNVOICED	$f_m > 0.56$	ONSET
	$0.56 \geq f_m > 0.45$	UNVOICED TRANSITION
	$f_m \leq 0.45$	UNVOICED
Current frame VAD information		
	VAD = 0	UNVOICED

In addition to this classification, the information **109** on the voice activity detection (VAD) by the encoder can be transmitted into the bitstream **101** (FIG. **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.

The classification scheme also includes a generic audio detection (see classifier portion **301** of FIG. **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).

First, a mean of the past forty (40) total frame energy variations $\overline{E}_{df}$ is calculated using the following relation:

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

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

$$\sigma_E = 0.77459647 \cdot \sqrt{\sum_{t=-15}^{t=-1} \frac{(\Delta_E^t - \overline{E}_{df})^2}{15}} \tag{10}$$

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.

Additionally, the first step classification is used to evaluate the interval between two frames classified as unvoiced N_{UV} when the frame energy E_f, as formulated in equation (5) is higher than −12 dB. When a frame is classified as unvoiced and the frame energy E_f is greater than −9 dB, 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 40 dB 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**.

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

$$N_{uv_{lt}} = 0.9 \cdot N_{uv_{lt}} + 0.1 \cdot N_{uv} \tag{11}$$

And as follows for inactive speech signal:

$$N_{uv_{lt}} = 0.95 \cdot N_{uv_{lt}} \tag{12}$$

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

$$N_{uv_{lt}} = 0.2 \cdot N_{uv_{lt}} + 80 \tag{13}$$

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:

$$A \text{ frame is } G_A \text{ if: } N_{uv_{lt}} > 140 \text{ and } \Delta_E^t < 12 \tag{14}$$

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.

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 VOICED UNVOICED TRANSITION ARTIFICIAL ONSET	Y (VAD = 1)	Active voice	N
GENERIC AUDIO SOUND	Y	Generic audio	Y*
VOICED TRANSITION UNVOICED	Y	Active unvoiced	Y
ONSET VOICED UNVOICED TRANSITION ARTIFICIAL ONSET GENERIC AUDIO SOUND VOICED TRANSITION UNVOICED	N	Inactive audio	Y

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 Transform

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:

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

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.

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

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

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

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 follows:

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

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

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

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

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"], of which the full content is incorporated herein by reference, 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

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 8^{th} harmonic is computed from the lowest pitch of all sub-frames and the time-domain contribution is kept up to this 8^{th} harmonic. An estimate of the 8^{th} harmonic is calculated as follows:

$$h_{8th}=\frac{(8\cdot F_s)}{\min_{0\le i<N_{sub}}(T(i))} \quad (17)$$

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 8^{th} harmonic is located by searching for the highest band for which the following inequality is still verified:

$$(h_{8^{th}}\ge L_f(i)) \quad (18)$$

where L_f is defined as:

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

The index of that frequency band in L_f will be called $i_{8^{th}}$ and it indicates the frequency band where the 8^{th} 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 8^{th} harmonic is likely to be located ($L_f(i_{8^{th}})$), using the following relation:

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

b) Normalization and Noise Fill

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.

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

$$f_{eN}(j)=\begin{cases}\dfrac{4\cdot f_e(j)}{\max\limits_{0\le i<f_c}(|f_e(i)|)}, & \text{for } 0\le j<f_c\\ 0, & \text{for } f_c\le j<256\end{cases} \quad (20)$$

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:

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

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

where rand is a random number generator which is limited between −1 to 1.

c) Energy Per Band Analysis of the Modified Excitation Spectrum

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

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:

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

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:

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

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

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

5) Excitation Modification for Generic Audio Frames

a) Normalization and Noise Fill

Reference will now be made to FIG. **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:

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

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:

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

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.

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 $f'_{eN}(j)$:

$$f'_{eN}(j)=\begin{cases}0 & \text{if } (f_{eN}(j)<Z_f(i))\\ f_{eN}(j) & \text{otherwise}\end{cases}\Bigg|_{\text{for } C_{Bb}(i)\le j<C_{Bb}(i)+B_b(i)} \tag{25}$$

where Z_f can be represented as:

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

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

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

FIG. **3** shows the gain calculator **303-304-306** and FIG. **4** describes in more detail calculator portion **306** of this gain calculator.

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):

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

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**.

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:

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

A finder **404** determines the maximum value $\max_{a\le j<b}(|f_{eN}(j)|)$ 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.

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_e 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}.

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 rom 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}

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

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

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

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

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 (de-normalized) frequency-domain excitation f'_{edN}

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 $f'_{eN}(j)<0.86\max_{a\le j<b}(|f_{eN}(j)|)$ (see module **452**), an attenuator **410** calculates an attenuation gain $[f'_{eN}$

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

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

$$f'_{edN}(j)=\begin{cases}\delta(i)\cdot f'_{eN}(j), & \text{if } f'_{eN}(j)\ge 0.86\cdot\max\limits_{a\le j<b}(|f_{eN}(j)|)\\ \dfrac{f'_{eN}(j)}{\max\limits_{a\le j<b}(|f_{eN}(j)|)}\cdot f'_{eN}(j), & \text{otherwise}\end{cases}\qquad(30)$$

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

A combiner **453** combines the contributions to the modified (de-normalized) 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}

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:

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

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

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.

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

What is claimed is:

1. A device for modifying a synthesis of a time-domain code-excited linear prediction (CELP) excitation decoded by a time-domain CELP decoder, comprising:

at least one processor; and

a memory coupled to the processor and comprising non-transitory instructions that when executed cause the processor to implement:

a classifier of the synthesis of the decoded time-domain CELP excitation into one of a number of categories;

a first converter of the decoded time-domain CELP excitation into a frequency-domain excitation;

a modifier of the frequency-domain excitation as a function of the category in which the synthesis of the decoded time-domain CELP excitation is classified by the classifier;

a second converter of the modified frequency-domain excitation into a modified time-domain CELP excitation; and

a first linear prediction synthesis filter supplied with the modified time-domain CELP excitation to produce a modified synthesis of the decoded time-domain CELP excitation.

2. A device for modifying a synthesis of a time-domain CELP excitation according to claim **1**, wherein the modifier comprises:

a first calculator of a cut-off frequency where a time-domain excitation contribution stops to be used.

3. A device for modifying a synthesis of a time-domain CELP excitation according to claim **2**, wherein the modifier comprises:

a zeroer of the frequency-domain excitation above the cut-off frequency; and

a normalizer of the frequency-domain excitation below the cut-off frequency to produce a normalized frequency-domain excitation.

4. A device for modifying a synthesis of a time-domain CELP excitation according to claim **3**, wherein the modifier comprises:

a random noise generator; and

an adder of the random noise to the normalized frequency-domain excitation.

5. A device for modifying a synthesis of a time-domain CELP excitation according to claim **3**, wherein the modifier comprises:

a second calculator of a matching gain using an energy of the frequency-domain excitation before and after modification, the second calculator applying the matching gain to the normalized frequency-domain excitation to produce the modified frequency-domain excitation.

6. A device for modifying a synthesis of a time-domain CELP excitation according to claim **2**, wherein the classifier classifies the synthesis of the decoded time-domain CELP excitation as inactive or active unvoiced.

7. A device for modifying a synthesis of a time-domain CELP excitation according to claim **1**, wherein the memory comprises non-transitory instructions that when executed cause the processor to implement:

a smoother of the first linear prediction synthesis filter when the synthesis of the decoded time-domain CELP excitation is classified as a given one of the categories by the classifier.

8. A device for modifying a synthesis of a time-domain CELP excitation according to claim **1**, wherein the frequency-domain excitation is divided into frequency bands each divided into frequency bins, and wherein the modifier comprises:

a normalizer of the frequency-domain excitation using a frequency band-varying normalization factor to produce a normalized frequency-domain excitation.

9. A device for modifying a synthesis of a time-domain CELP excitation according to claim **8**, wherein the normalizer comprises:

a zeroer of the frequency bins below a fraction of a maximum value of the normalized frequency-domain excitation in the frequency band comprising the frequency bins.

10. A device for modifying a synthesis of a time-domain CELP excitation according to claim **8**, wherein the modifier comprises:

a first calculator of a matching gain per frequency band using an energy of the frequency-domain excitation before and after modification.

11. A device for modifying a synthesis of a time-domain CELP excitation according to claim **10**, wherein the modifier comprises, for the frequency bands below a first frequency:

a second calculator for applying the matching gain to the normalized frequency-domain excitation to produce the modified frequency-domain excitation.

12. A device for modifying a synthesis of a time-domain CELP excitation according to claim **10**, wherein, for the frequency bands between a first lower frequency and a second higher frequency, the memory comprises non-transitory instructions that when executed cause the processor to implement:

a finder of a maximum value per frequency band of the normalized frequency-domain excitation;

an amplifier for amplifying the matching gain by an amplification factor per frequency bin when the normalized frequency-domain excitation in the frequency bin is equal to or higher than a value proportional to said maximum value of the frequency band; and

a second calculator for applying the amplified matching gain to the normalized frequency-domain excitation in the frequency bin to produce in said frequency bin the modified frequency-domain excitation.

13. A device for modifying a synthesis of a time-domain CELP excitation according to claim **10**, wherein, for the frequency bands between a first lower frequency and a second higher frequency, the memory comprises non-transitory instructions that when executed cause the processor to implement:

a finder of a maximum value per frequency band of the normalized frequency-domain excitation;

an attenuator for attenuating the matching gain by an attenuation factor per frequency bin of the frequency band when the normalized frequency-domain excitation in the frequency bin is lower than a value proportional to said maximum value of the frequency band; and

a second calculator for applying the attenuated matching gain to the normalized frequency-domain excitation in said frequency bin to produce in said frequency bin the modified frequency-domain excitation.

14. A device for modifying a synthesis of a time-domain CELP excitation according to claim **10**, wherein, for the frequency bands above a given frequency, the memory comprises non-transitory instructions that when executed cause the processor to implement:

a finder of a maximum value per frequency band of the normalized frequency-domain excitation;

a second calculator of a tilt for the matching gain when the normalized frequency-domain excitation in the frequency bin is higher than a value proportional to said maximum value of the frequency band, the second calculator applying the calculated tilt to the matching gain; and

a third calculator for applying the matching gain to which the calculated tilt has been applied to the normalized frequency-domain excitation in said frequency bin to produce in said frequency bin the modified frequency-domain excitation.

15. A device for modifying a synthesis of a time-domain CELP excitation according to claim **10**, wherein, for the frequency bands above a given frequency, the memory comprises non-transitory instructions that when executed cause the processor to implement:

a finder of a maximum value per frequency band of the normalized frequency-domain excitation;

an attenuator for attenuating the matching gain by an amplification factor per frequency bin of the frequency band when the normalized frequency-domain excitation in the frequency bin is lower than a value proportional to the maximum value of the frequency band; and

a second calculator for applying the attenuated matching gain to the normalized frequency-domain excitation in said frequency bin to produce in said frequency bin the modified frequency-domain excitation.

16. A device for decoding a sound signal encoded by encoding parameters, comprising:

a decoder of a time-domain CELP excitation in response to the sound signal encoding parameters;

a second synthesis filter responsive to the decoded time-domain CELP excitation to produce a synthesis of said time-domain CELP excitation; and

a device according to claim **1**, for modifying the synthesis of the time-domain CELP excitation.

17. A method for modifying a synthesis of a time-domain code-excited linear prediction (CELP) excitation decoded by a time-domain CELP decoder, comprising:

classifying the synthesis of the decoded time-domain CELP excitation into one of a number of categories;

converting the decoded time-domain CELP excitation into a frequency-domain excitation;

modifying the frequency-domain excitation as a function of the category in which the synthesis of the decoded time-domain CELP excitation is classified;

converting the modified frequency-domain excitation into a modified time-domain CELP excitation;

synthesizing, using a linear prediction synthesis filter, the modified time-domain CELP excitation to produce a modified synthesis of the decoded time-domain CELP excitation.

18. A method for modifying a synthesis of a time-domain CELP excitation according to claim **17**, wherein modifying the frequency-domain excitation comprises:

calculating a cut-off frequency where a time-domain excitation contribution stops to be used.

19. A method for modifying a synthesis of a time-domain CELP excitation according to claim **18**, wherein modifying the frequency-domain excitation comprises:

zeroing the frequency-domain excitation above the cut-off frequency; and

normalizing the frequency-domain excitation below the cut-off frequency to produce a normalized frequency-domain excitation.

20. A method for modifying a synthesis of a time-domain CELP excitation according to claim **19**, wherein modifying the frequency-domain excitation comprises generating a random noise and adding the random noise to the normalized frequency-domain excitation.

21. A method for modifying a synthesis of a time-domain CELP excitation according to claim **19**, wherein modifying the frequency-domain excitation comprises:

calculating a matching gain using an energy of the frequency-domain excitation before and after modification, and applying the matching gain to the normalized frequency-domain excitation to produce the modified frequency-domain excitation.

22. A method for modifying a synthesis of a time-domain CELP excitation according to claim **18**, wherein the synthesis of the decoded time-domain CELP excitation is classified as inactive or active unvoiced.

23. A method for modifying a synthesis of a time-domain CELP excitation according to claim **17**, comprising smoothing the linear prediction synthesis filter performing the synthesis of the modified time-domain CELP excitation when the synthesis of the decoded time-domain CELP excitation is classified as a given one of the categories by a classifier.

24. A method for modifying a synthesis of a time-domain excitation according to claim **17**, wherein the frequency-domain excitation is divided into frequency bands each divided into frequency bins, and wherein modifying the frequency-domain excitation comprises:

normalizing the frequency-domain excitation using a frequency band-varying normalization factor to produce a normalized frequency-domain excitation.

25. A method for modifying a synthesis of a time-domain CELP excitation according to claim **24**, wherein modifying the frequency-domain excitation comprises zeroing the frequency bins below a fraction of a maximum value of the normalized frequency-domain excitation in the frequency band comprising the frequency bins.

26. A method for modifying a synthesis of a time-domain CELP excitation according to claim **24**, wherein modifying the frequency-domain excitation comprises:

calculating a matching gain per frequency band using an energy of the frequency-domain excitation before and after modification.

27. A method for modifying a synthesis of a time-domain CELP excitation according to claim **26**, wherein modifying the frequency-domain excitation comprises, for the frequency bands below a first frequency, applying the matching gain to the normalized frequency-domain excitation to produce the modified frequency-domain excitation.

28. A method for modifying a synthesis of a time-domain CELP excitation according to claim **26**, comprising, for the frequency bands between a first lower frequency and a second higher frequency:

finding a maximum value per frequency band of the normalized frequency-domain excitation;

amplifying the matching gain by an amplification factor per frequency bin when the normalized frequency-domain excitation in the frequency bin is equal to or higher than a value proportional to said maximum value of the frequency band; and

applying the amplified matching gain to the normalized frequency-domain excitation in the frequency bin to produce in said frequency bin the modified frequency-domain excitation.

29. A method for modifying a synthesis of a time-domain CELP excitation according to claim **26**, comprising, for the frequency bands between a first lower frequency and a second higher frequency:

finding a maximum value per frequency band of the normalized frequency-domain excitation;

attenuating the matching gain by an attenuation factor per frequency bin of the frequency band when the normalized frequency-domain excitation in the frequency bin is lower than a value proportional to said maximum value of the frequency band; and

applying the attenuated matching gain to the normalized frequency-domain excitation in said frequency bin to produce in said frequency bin the modified frequency-domain excitation.

30. A method for modifying a synthesis of a time-domain CELP excitation according to claim **26**, comprising, for the frequency bands above a given frequency:

finding a maximum value per frequency band of the normalized frequency-domain excitation;

calculating a tilt for the matching gain when the normalized frequency-domain excitation in the frequency bin is higher than a value proportional to said maximum value of the frequency band, and applying the calculated tilt to the matching gain; and

applying the matching gain to which the calculated tilt has been applied to the normalized frequency-domain excitation in said frequency bin to produce in said frequency bin the modified frequency-domain excitation.

31. A method for modifying a synthesis of a time-domain excitation according to claim **26**, comprising, for the frequency bands above a given frequency:

finding a maximum value per frequency band of the normalized frequency-domain excitation;

attenuating the matching gain by an amplification factor per frequency bin of the frequency band when the normalized frequency-domain excitation in the frequency bin is lower than a value proportional to the maximum value of the frequency band; and

applying the attenuated matching gain to the normalized frequency-domain excitation in said frequency bin to produce in said frequency bin the modified frequency-domain excitation.

32. A method for decoding a sound signal encoded by encoding parameters, comprising:

decoding a time-domain CELP excitation in response to the sound signal encoding parameters;

synthesizing the decoded time-domain CELP excitation to produce a synthesis of said time-domain CELP excitation; and

a method according to claim **17**, for modifying the synthesis of the time-domain CELP excitation.

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