AUDIO SIGNAL ENCODING AND DECODING METHOD, AND AUDIO SIGNAL ENCODING AND DECODING APPARATUS

TECHNICAL FIELD

[0002] The present invention relates to the field of communications technologies, and in particular, to an audio signal encoding method, an audio signal decoding method, an audio signal encoding apparatus, an audio signal decoding apparatus, a transmitter, a receiver, and a communications system.

BACKGROUND

[0003] With continuous progress of communications technologies, users are imposing an increasingly high requirement on voice quality. Generally, voice quality is improved by increasing bandwidth of the voice quality. If information whose bandwidth is increased is encoded in a traditional encoding manner, a bit rate is greatly improved and as a result, it is difficult to implement encoding because of a limitation condition of current network bandwidth. Therefore, encoding needs to be performed on a signal whose bandwidth is wider in a case in which a bit rate is unchanged or slightly changed, and a solution proposed for this issue is to use a bandwidth extension technology. The bandwidth extension technology may be completed in a time domain or a frequency domain, and bandwidth extension is completed in the time domain in the present invention.

[0004] A basic principle of performing bandwidth extension in a time domain is

that two different processing methods are used for a low band signal and a high band signal. For a low band signal in an original signal, encoding is performed at an encoder side according to a requirement by using various encoders; at a decoder side, a decoder corresponding to the encoder of the encoder side is used to decode and restore the low band signal. For a high band signal, at the encoder side, an encoder used for the low band signal is used to obtain a low frequency encoding parameter so as to predict a high band excitation signal; a linear predictive coding (LPC, linear Predictive Coding) analysis, for example, is performed on a high band signal of the original signal to obtain a high frequency LPC coefficient. The high band excitation signal is filtered by using a synthesis filter determined according to the LPC coefficient so as to obtain a predicted high band signal; the predicted high band signal is compared with the high band signal in the original signal so as to obtain a high frequency gain parameter; the high frequency gain parameter and the LPC coefficient are transferred to the decoder side to restore the high band signal. At the decoder side, the low frequency encoding parameter extracted during decoding of the low band signal is used to restore the high band excitation signal; the LPC coefficient is used to generate the synthesis filter; the high band excitation signal is filtered by using the synthesis filter so as to restore the predicted high band signal; the predicted high band signal is adjusted by using the high frequency gain parameter so as to obtain a final high band signal; the high band signal and the low band signal are combined to obtain a final output signal.

[0005] In the foregoing technology of performing bandwidth extension in a time domain, a high band signal is restored in a condition of a specific rate; however, a performance indicator is deficient. It can be learned by comparing a frequency spectrum of a restored output signal with a frequency spectrum of an original signal that, for a voiced sound of a general period, there is always an extremely strong harmonic component in a restored high band signal. However, a high band signal in an authentic voice signal does not have an extremely strong harmonic characteristic. Therefore, this difference causes that there is an obvious mechanical sound when the restored signal sounds.

[0006] An objective of embodiments of the present invention is to improve the foregoing technology of performing bandwidth extension in the time domain, so as to reduce or even remove the mechanical sound in the restored signal.

digital circuit or a chip, for example, a DSP (digital signal processor), or be implemented by using software code to drive a processor to execute a process in the software code.

[0049] In addition, the audio codec and an audio encoding and decoding method may also be applied to various communications systems, such as GSM, a Code Division Multiple Access (CDMA, Code Division Multiple Access) system, Wideband Code Division Multiple Access (WCDMA, Wideband Code Division Multiple Access), a general packet radio service (GPRS, General Packet Radio Service), and Long Term Evolution (LTE, Long Term Evolution).

[0050] FIG. 1 is a schematic flowchart of an audio signal encoding method 100 according to an embodiment of the present invention. The audio signal encoding method includes: dividing a to-be-encoded time domain signal into a low band signal and a high band signal (step 110); encoding the low band signal to obtain a low frequency encoding parameter (step 120); calculating a voiced degree factor according to the low frequency encoding parameter, and predicting a high band excitation signal according to the low frequency encoding parameter, where the voiced degree factor is used to indicate a degree of a voiced characteristic presented by the high band signal (step 130); weighting the high band excitation signal and random noise by using the voiced degree factor, so as to obtain a synthesized excitation signal (step 140); and obtaining a high frequency encoding parameter based on the synthesized excitation signal and the high band signal (step 150).

[0051] In step 110, the to-be-encoded time domain signal is divided into the low band signal and the high band signal. The division is to divide the time domain signal into two signals for processing, so that the low band signal and the high band signal can be separately processed. The division may be implemented by using any conventional or future division technology. The meaning of the low frequency herein is relative to the meaning of the high frequency. For example, a frequency threshold may be set, where a frequency lower than the frequency threshold is a low frequency, and a frequency higher than the frequency threshold is a high frequency. In practice, the frequency threshold may be set according to a requirement, and a low band signal component and a high band signal component in a signal may also be distinguished by using another manner, so as to implement division.

[0052] In step 120, the low band signal is encoded to obtain the low frequency encoding parameter. By the encoding, the low band signal is processed so as to obtain

the low frequency encoding parameter, so that a decoder side restores the low band signal according to the low frequency encoding parameter. The low frequency encoding parameter is a parameter required by the decoder side to restore the low band signal. As an example, encoding may be performed by using an encoder (ACELP encoder) using an algebraic code excited linear prediction (ACELP, Algebraic Code Excited Linear Prediction) algorithm, and a low frequency encoding parameter obtained in this case may include, for example, an algebraic codebook, an algebraic codebook gain, an adaptive codebook, an adaptive codebook gain, and a pitch period, and may also include another parameter. The low frequency encoding parameter may be transferred to the decoder side to restore the low band signal. In addition, when the algebraic codebook and the adaptive codebook are transferred from an encoder side to the decoder side, only an algebraic codebook index and an adaptive codebook index may be transferred, and the decoder side obtains a corresponding algebraic codebook and adaptive codebook according to the algebraic codebook index and the adaptive codebook index, so as to implement restoration.

[0053] In practice, the low band signal may be encoded by using a proper encoding technology according to a requirement. When an encoding technology changes, composition of the low frequency encoding parameter may also change. In this embodiment of the present invention, an encoding technology using the ACELP algorithm is used as an example for description.

[0054] In step 130, the voiced degree factor is calculated according to the low frequency encoding parameter, and the high band excitation signal is predicted according to the low frequency encoding parameter, where the voiced degree factor is used to indicate the degree of the voiced characteristic presented by the high band signal. Therefore, step 130 is used to obtain the voiced degree factor and the high band excitation signal from the low frequency encoding parameter, where the voiced degree factor and the high band excitation signal are used to indicate different characteristics of the high band signal, that is, a high frequency characteristic of an input signal is obtained in step 130, so that the high frequency characteristic is used for encoding of the high band signal. The encoding technology using the ACELP algorithm is used as an example below, so as to describe calculation of both the voiced degree factor and the high band excitation signal.

[0055] The voiced degree factor voice_fac may be calculated according to the following formula (1):

codebook gain. It may be learned that, in formula (3) or (4), the algebraic codebook FixCB and the random noise seed are weighted by using the voiced degree factor, so as to obtain a weighting result; and a product of the weighting result and the algebraic codebook gain gc, and a product of the adaptive codebook AdpCB and the adaptive codebook gain ga are added, so as to obtain the high band excitation signal Ex. Alternatively, in formula (3) or (4), the voiced degree factor voice_fac may be replaced with the modified voiced degree factor voice_fac_A in formula (2), so as to more accurately indicate the degree of the voiced characteristic presented by the high band signal, that is, a high band signal in a voice signal is more realistically indicated, thereby improving an encoding effect.

[0058] It should be noted that, the foregoing manners of calculating the voiced degree factor and the high band excitation signal are merely exemplary, and are not intended to limit this embodiment of the present invention. In another encoding technology without using the ACELP algorithm, the voiced degree factor and the high band excitation signal may also be calculated by using another manner.

[0059] In step 140, the high band excitation signal and the random noise are weighted by using the voiced degree factor, so as to obtain the synthesized excitation signal. As described above, in the prior art, for the voiced signal of a general period, because periodicity of the high band excitation signal predicted according to the low frequency encoding parameter is extremely strong, there is a strong mechanical sound when a restored audio signal sounds. By step 140, the high band excitation signal predicted according to the low band signal and the noise are weighted by using the voiced degree factor, which can weaken periodicity of the high band excitation signal predicted according to the low frequency encoding parameter, thereby weakening a mechanical sound in the restored audio signal.

[0060] The weighting may be implemented by using a proper weight according to a requirement. As an example, the synthesized excitation signal SEx may be obtained according to the following formula (5):

$$SEx = Ex * \sqrt{\sqrt{voice_fac}} + seed\sqrt{pow1 * \left(1 - \sqrt{voice_fac}\right) / pow2}$$

formula (5)

where Ex is the high band excitation signal, seed is the random noise, voice_fac is the voiced degree factor, pow1 is energy of the high band excitation signal, and pow2 is energy of the random noise. Alternatively, in formula (5), the

voiced degree factor voice_fac may be replaced with the modified voiced degree factor voice_fac_A in formula (2), so as to more accurately indicate the high band signal in the voice signal, thereby improving an encoding effect. In a case that in formula (2), a1=0.0126, b1=1.23, a2=0.0087, b2=0, threshold_min=57.75, and threshold_max=115.5, if the synthesized excitation signal SEx is obtained according to formula (5), a high band excitation signal of which a pitch period T0 is greater than threshold_max and less than threshold_min has a greater weight, and another high band excitation signal has a less weight. It should be noted that, according to a requirement, the synthesized excitation signal may also be calculated by using another manner in addition to formula (5).

[0061] In addition, when the high band excitation signal and the random noise are weighted by using the voiced degree factor, pre-emphasis may also be performed on the random noise in advance, and de-emphasis may be performed on the random noise after weighting. Specifically, step 140 may include: performing, on the random noise by using a pre-emphasis factor, a pre-emphasis operation for enhancing a high frequency part of the random noise, so as to obtain pre-emphasis noise; weighting the high band excitation signal and the pre-emphasis noise by using the voiced degree factor, so as to generate a pre-emphasis excitation signal; and performing, on the pre-emphasis excitation signal by using a de-emphasis factor, a de-emphasis operation for lowering a high frequency part of the pre-emphasis excitation signal, so as to obtain the synthesized excitation signal. For a general voiced sound, a noise component usually becomes stronger from a low frequency to a high frequency. Based on this, the pre-emphasis operation is performed on the random noise, so as to accurately indicate a noise signal characteristic of a voiced sound, that is, a high frequency part of noise is improved and a low frequency part of the noise is lowered. As an example of the pre-emphasis operation, a pre-emphasis operation may be performed on the random noise seed(n) by using the following formula (6):

$$seed(n)=seed(n)-\alpha seed(n-1) \qquad \text{formula (6)}$$

where n=1, 2, ... N, and α is the pre-emphasis factor and $0< \alpha <1$. The pre-emphasis factor may be properly set based on a characteristic of the random noise, so as to accurately indicate the noise signal characteristic of the voiced sound. In a case that the pre-emphasis operation is performed by using formula (6), a de-emphasis operation may be performed on the pre-emphasis excitation signal S(i) by using the following formula (7):

$$S(n)= S(n)+\beta S(n-1) \quad \text{formula (7)}$$

where n=1, 2, ... N, and β is a preset de-emphasis factor. It should be noted that, the pre-emphasis operation shown in the foregoing formula (6) is merely exemplary, and in practice, pre-emphasis may be performed by using another manner. In addition, when a used pre-emphasis operation changes, the de-emphasis operation also needs to correspondingly change. The de-emphasis factor β may be determined based on the pre-emphasis factor α and a proportion of the pre-emphasis noise in the pre-emphasis excitation signal. As an example, when the high band excitation signal and the pre-emphasis noise are weighted according to formula (5) by using the voiced degree factor (the pre-emphasis excitation signal is obtained in this case, and the synthesized excitation signal is obtained only after de-emphasis is performed on the pre-emphasis excitation signal), the de-emphasis factor β may be determined according to the following formula (8) or formula (9):

$$\beta = \alpha * weight1/(weight1 + weight2)$$
$$where, weight1 = 1 - \sqrt{1 - voice_fac}, weight2 = \sqrt{voice_fac} \quad \text{formula (8)}$$

$$\beta = \alpha * weight1/(weight1 + weight2)$$
$$where, weight1 = \sqrt{\left(1 - \sqrt{1 - voice_fac}\right)}, weight2 = \sqrt{\sqrt{voice_fac}}$$

formula (9)

[0062] In step 150, the high frequency encoding parameter is obtained based on the synthesized excitation signal and the high band signal. As an example, the high frequency encoding parameter includes a high frequency gain parameter and a high frequency LPC coefficient. The high frequency LPC coefficient may be obtained by performing an LPC analysis on a high band signal in an original signal; a predicted high band signal is obtained after the high band excitation signal is filtered by using a synthesis filter determined according to the LPC coefficient; the high frequency gain parameter is obtained by comparing the predicted high band signal with the high band signal in the original signal, where the high frequency gain parameter and the LPC coefficient are transferred to the decoder side to restore the high band signal. In addition, the high frequency encoding parameter may also be obtained by using various conventional or future technologies, and a specific manner of obtaining the high frequency encoding parameter based on the synthesized excitation signal and the high band signal does not constitute a limitation to the present invention. After the low frequency encoding parameter and the high frequency encoding parameter are

obtained, encoding of a signal is implemented, so that the signal can be transferred to the decoder side for restoration.

[0063] After the low frequency encoding parameter and the high frequency encoding parameter are obtained, the audio signal encoding method 100 may further include: generating a coded bitstream according to the low frequency encoding parameter and the high frequency encoding parameter, so as to send the coded bitstream to the decoder side.

[0064] In the foregoing audio signal encoding method in this embodiment of the present invention, a high band excitation signal and random noise are weighted by using a voiced degree factor, so as to obtain a synthesized excitation signal, and a characteristic of a high band signal may be more accurately presented based on a voiced signal, thereby improving an encoding effect.

[0065] FIG. 2 is a schematic flowchart of an audio signal decoding method 200 according to an embodiment of the present invention. The audio signal decoding method includes: distinguishing a low frequency encoding parameter and a high frequency encoding parameter in encoded information (step 210); decoding the low frequency encoding parameter to obtain a low band signal (step 220); calculating a voiced degree factor according to the low frequency encoding parameter, and predicting a high band excitation signal according to the low frequency encoding parameter, where the voiced degree factor is used to indicate a degree of a voiced characteristic presented by a high band signal (step 230); weighting the high band excitation signal and random noise by using the voiced degree factor, so as to obtain a synthesized excitation signal (step 240); obtaining the high band signal based on the synthesized excitation signal and the high frequency encoding parameter (step 250); and combining the low band signal and the high band signal to obtain a final decoded signal (step 260).

[0066] In step 210, the low frequency encoding parameter and the high frequency encoding parameter are distinguished in the encoded information. The low frequency encoding parameter and the high frequency encoding parameter are parameters that are transferred from an encoder side and used to restore the low band signal and the high band signal. The low frequency encoding parameter may include, for example, an algebraic codebook, an algebraic codebook gain, an adaptive codebook, an adaptive codebook gain, a pitch period, and another parameter, and the high frequency encoding parameter may include, for example, an LPC coefficient, a high frequency gain parameter, and another parameter. In addition, according to a different encoding

technology, the low frequency encoding parameter and the high frequency encoding parameter may alternatively include another parameter.

[0067] In step 220, the low frequency encoding parameter is decoded to obtain the low band signal. A specific decoding mode is corresponding to an encoding manner of the encoder side. As an example, when encoding is performed on the encoder side by using an ACELP encoder using an ACELP algorithm, an ACELP decoder is used in step 220 to obtain the low band signal.

[0068] In step 230, the voiced degree factor is calculated according to the low frequency encoding parameter, and the high band excitation signal is predicted according to the low frequency encoding parameter, where the voiced degree factor is used to indicate the degree of the voiced characteristic presented by the high band signal. step 230 is used to obtain a high frequency characteristic of an encoded signal according to the low frequency encoding parameter, so that the high frequency characteristic is used for decoding (or restoration) of the high band signal. A decoding technology that is corresponding to an encoding technology using the ACELP algorithm is used as an example for description in the following.

[0069] The voiced degree factor voice_fac may be calculated according to the foregoing formula (1), and to better present a characteristic of the high band signal, the voiced degree factor voice_fac may be modified as shown in the foregoing formula (2) by using the pitch period in the low frequency encoding parameter, and a modified voiced degree factor voice_fac_A may be obtained. Compared with an unmodified voiced degree factor voice_fac, the modified voiced degree factor voice_fac_A can more accurately indicate the degree of the voiced characteristic presented by the high band signal, thereby helping to weaken a mechanical sound introduced after a voiced signal of a general period is extended.

[0070] The high band excitation signal Ex may be calculated according to the foregoing formula (3) or formula (4), that is, the algebraic codebook and the random noise are weighted by using the voiced degree factor, so as to obtain a weighting result; and a product of the weighting result and the algebraic codebook gain, and a product of the adaptive codebook and the adaptive codebook gain are added, so as to obtain the high band excitation signal Ex. Similarly, the voiced degree factor voice_fac may be replaced with the modified voiced degree factor voice_fac_A in formula (2), so as to further improve a decoding effect.

[0071] The foregoing manners of calculating the voiced degree factor and the

high band excitation signal are merely exemplary, and are not used to limit this embodiment of the present invention. In another encoding technology without using the ACELP algorithm, the voiced degree factor and the high band excitation signal may also be calculated by using another manner.

[0072] For description of step 230, refer to the foregoing description of step 130 with reference to FIG. 1.

[0073] In step 240, the high band excitation signal and the random noise are weighted by using the voiced degree factor, so as to obtain the synthesized excitation signal. By step 240, the high band excitation signal predicted according to the low frequency encoding parameter and the noise are weighted by using the voiced degree factor, which can weaken periodicity of the high band excitation signal predicted according to the low frequency encoding parameter, thereby weakening a mechanical sound in the restored audio signal.

[0074] As an example, in step 240, the synthesized excitation signal SEex may be obtained according to the foregoing formula (5), and the voiced degree factor voice_fac in formula (5) may be replaced with the modified voiced degree factor voice_fac_A in formula (2), so as to more accurately indicate a high band signal in a voice signal, thereby improving an encoding effect. According to a requirement, the synthesized excitation signal may also be calculated by using another manner.

[0075] In addition, when the high band excitation signal and the random noise are weighted by using the voiced degree factor voice_fac (or the modified voiced degree factor voice_fac_A), pre-emphasis may also be performed on the random noise in advance, and de-emphasis may be performed on the random noise after weighting. Specifically, step 240 may include: performing, on the random noise by using a pre-emphasis factor α, a pre-emphasis operation (for example, the pre-emphasis operation is implemented by using formula (6)) for enhancing a high frequency part of the random noise, so as to obtain pre-emphasis noise; weighting the high band excitation signal and the pre-emphasis noise by using the voiced degree factor, so as to generate a pre-emphasis excitation signal; and performing, on the pre-emphasis excitation signal by using a de-emphasis factor β, a de-emphasis operation (for example, the de-emphasis operation is implemented by using formula (7)) for lowering a high frequency part of the pre-emphasis excitation signal, so as to obtain the synthesized excitation signal. The pre-emphasis factor α may be preset according to a requirement, so as to accurately indicate a noise signal characteristic of a voiced

sound, that is, a high frequency part of noise has a strong signal and a low frequency part of the noise has a weak signal. In addition, noise of another type may also be used, and in this case, the pre-emphasis factor α needs to correspondingly change, so as to indicate a noise characteristic of a general voiced sound. The de-emphasis factor β may be determined based on the pre-emphasis factor α and a proportion of the pre-emphasis noise in the pre-emphasis excitation signal. As an example, the de-emphasis factor β may be determined according to the foregoing formula (8) or formula (9).

[0076] For description of step 240, refer to the foregoing description of step 140 with reference to FIG. 1.

[0077] In step 250, the high band signal is obtained based on the synthesized excitation signal and the high frequency encoding parameter. Step 250 is implemented in an inverse process of obtaining the high frequency encoding parameter based on the synthesized excitation signal and the high band signal on the encoder side. As an example, the high frequency encoding parameter includes a high frequency gain parameter and a high frequency LPC coefficient; a synthesis filter may be generated by using the LPC coefficient in the high frequency encoding parameter; the predicted high band signal is restored after the synthesized excitation signal obtained in step 240 is filtered by the synthesis filter; and a final high band signal is obtained after the predicted high band signal is adjusted by using the high frequency gain parameter in the high frequency encoding parameter. In addition, step 240 may also be implemented by using various conventional or future technologies, and a specific manner of obtaining the high band signal based on the synthesized excitation signal and the high frequency encoding parameter does not constitute a limitation to the present invention.

[0078] In step 260, the low band signal and the high band signal are combined to obtain the final decoded signal. This combining manner is corresponding to a division manner in step 110 in FIG. 1, so that decoding is implemented to obtain a final output signal.

[0079] In the foregoing audio signal decoding method in this embodiment of the present invention, a high band excitation signal and random noise are weighted by using a voiced degree factor, so as to obtain a synthesized excitation signal, and a characteristic of a high band signal may be more accurately presented based on a voiced signal, thereby improving a decoding effect.

[0080] FIG. 3 is a schematic block diagram of an audio signal encoding apparatus

decoding processor 805.

[0111] In addition, for example, the transmitter 610 in FIG. 6 may be implemented by the encoding processor 804, the transmitting circuit 802, the antenna 801, and the like. The receiver 710 in FIG. 7 may be implemented by the antenna 801, the receiving circuit 803, the decoding processor 805, and the like. However, the foregoing examples are merely exemplary, and are not intended to limit the embodiments of the present invention to this specific implementation form.

[0112] Specifically, the memory 807 stores an instruction that enables the processor 806 and/or the encoding processor 804 to implement the following operations: dividing a to-be-encoded time domain signal into a low band signal and a high band signal; encoding the low band signal to obtain a low frequency encoding parameter; calculating a voiced degree factor according to the low frequency encoding parameter, and predicting a high band excitation signal according to the low frequency encoding parameter, where the voiced degree factor is used to indicate a degree of a voiced characteristic presented by the high band signal; weighting the high band excitation signal and random noise by using the voiced degree factor, so as to obtain a synthesized excitation signal; and obtaining a high frequency encoding parameter based on the synthesized excitation signal and the high band signal. The memory 807 stores an instruction that enables the processor 806 or the decoding processor 805 to implement the following operations: distinguishing a low frequency encoding parameter and a high frequency encoding parameter in encoded information; decoding the low frequency encoding parameter to obtain a low band signal; calculating a voiced degree factor according to the low frequency encoding parameter, and predicting a high band excitation signal according to the low frequency encoding parameter, where the voiced degree factor is used to indicate a degree of a voiced characteristic presented by a high band signal; weighting the high band excitation signal and random noise by using the voiced degree factor, so as to obtain a synthesized excitation signal; obtaining the high band signal based on the synthesized excitation signal and the high frequency encoding parameter; and combining the low band signal and the high band signal to obtain a final decoded signal.

[0113] A communications system or communications apparatus according to an embodiment of the present invention may include a part of or all of the foregoing audio signal encoding apparatus 300, transmitter 600, audio signal decoding apparatus 500, receiver 700, and the like.

CLAIMS:

1. An audio signal encoding method, comprising:

dividing a to-be-encoded time domain signal into a low band signal and a high band signal;

encoding the low band signal to obtain a low frequency encoding parameter;

calculating a voiced degree factor according to the low frequency encoding parameter, and predicting a high band excitation signal according to the low frequency encoding parameter, wherein the voiced degree factor is used to indicate a degree of a voiced characteristic presented by the high band signal;

weighting the high band excitation signal and random noise by using the voiced degree factor, so as to obtain a synthesized excitation signal; and

obtaining a high frequency encoding parameter based on the synthesized excitation signal and the high band signal;

wherein the weighting the high band excitation signal and random noise by using the voiced degree factor, so as to obtain a synthesized excitation signal comprises:

performing, on the random noise by using a pre-emphasis factor, a pre-emphasis operation for enhancing a high frequency part of the random noise, so as to obtain pre-emphasis noise;

weighting the high band excitation signal and the pre-emphasis noise by using the voiced degree factor, so as to generate a pre-emphasis excitation signal; and

performing, on the pre-emphasis excitation signal by using a de-emphasis factor, a de-emphasis operation for lowering a high frequency part of the pre-emphasis

excitation signal, so as to obtain the synthesized excitation signal.

2. The method according to claim 1, wherein the de-emphasis factor is determined based on the pre-emphasis factor and a proportion of the pre-emphasis noise in the pre-emphasis excitation signal.

3.An audio signal encoding method, comprising:

dividing a to-be-encoded time domain signal into a low band signal and a high band signal;

encoding the low band signal to obtain a low frequency encoding parameter;

calculating a voiced degree factor according to the low frequency encoding parameter, and predicting a high band excitation signal according to the low frequency encoding parameter, wherein the voiced degree factor is used to indicate a degree of a voiced characteristic presented by the high band signal;

weighting the high band excitation signal and random noise by using the voiced degree factor, so as to obtain a synthesized excitation signal; and

obtaining a high frequency encoding parameter based on the synthesized excitation signal and the high band signal;

wherein the low frequency encoding parameter comprises a pitch period, and the weighting the high band excitation signal and random noise by using the voiced degree factor, so as to obtain a synthesized excitation signal comprises:

modifying the voiced degree factor by using the pitch period; and

weighting the high band excitation signal and the random noise by using a modified voiced degree factor, so as to obtain the synthesized excitation signal.

4.An audio signal encoding method, comprising:

dividing a to-be-encoded time domain signal into a low band signal and a high band signal;

encoding the low band signal to obtain a low frequency encoding parameter;

calculating a voiced degree factor according to the low frequency encoding parameter, and predicting a high band excitation signal according to the low frequency encoding parameter, wherein the voiced degree factor is used to indicate a degree of a voiced characteristic presented by the high band signal;

weighting the high band excitation signal and random noise by using the voiced degree factor, so as to obtain a synthesized excitation signal; and

obtaining a high frequency encoding parameter based on the synthesized excitation signal and the high band signal ;

wherein the low frequency encoding parameter comprises an algebraic codebook, an algebraic codebook gain, an adaptive codebook, an adaptive codebook gain, and a pitch period, and the predicting a high band excitation signal according to the low frequency encoding parameter comprises:

modifying the voiced degree factor by using the pitch period; and

weighting the algebraic codebook and the random noise by using a modified voiced degree factor, so as to obtain a weighting result, and adding a product of the weighting result and the algebraic codebook gain and a product of the adaptive codebook and the adaptive codebook gain, so as to predict the high band excitation signal.

5. The method according to claim 4, wherein the modifying the voiced degree factor by using the pitch period is performed according to the following formula:

$$voice_fac_A = voice_fac * \gamma$$

$$\gamma = \begin{cases} -a1 * T0 + b1 & T0 \le threshold_\min \\ a2 * T0 + b2 & threshold_\min \le T0 \le threshold_\max \\ 1 & T0 \ge threshold_\max \end{cases}$$

wherein voice_fac is the voiced degree factor, T0 is the pitch period, a1, a2, and $b1 > 0$, $b2 \ge 0$, threshold_min and threshold_max are respectively a preset minimum value and a preset maximum value of the pitch period, and voice_fac_A is the modified voiced degree factor;

6. The method according to claim3, wherein the modifying the voiced degree factor by using the pitch period is performed according to the following formula:

$$voice_fac_A = voice_fac * \gamma$$

$$\gamma = \begin{cases} -a1 * T0 + b1 & T0 \le threshold_\min \\ a2 * T0 + b2 & threshold_\min \le T0 \le threshold_\max \\ 1 & T0 \ge threshold_\max \end{cases}$$

wherein voice_fac is the voiced degree factor, T0 is the pitch period, a1, a2, and $b1 > 0$, $b2 \ge 0$, threshold_min and threshold_max are respectively a preset minimum value and a preset maximum value of the pitch period, and voice_fac_A is the modified voiced degree factor.

7. An audio signal decoding method, comprising:

distinguishing a low frequency encoding parameter and a high frequency encoding parameter in encoded information;

decoding the low frequency encoding parameter to obtain a low band signal;

calculating a voiced degree factor according to the low frequency encoding parameter, and predicting a high band excitation signal according to the low frequency encoding parameter, wherein the voiced degree factor is used to indicate a degree of a voiced characteristic presented by a high band signal;

weighting the high band excitation signal and random noise by using the voiced degree factor, so as to obtain a synthesized excitation signal;

obtaining the high band signal based on the synthesized excitation signal and the high frequency encoding parameter; and

combining the low band signal and the high band signal to obtain a final decoded signal;

wherein the weighting the high band excitation signal and random noise by using the voiced degree factor, so as to obtain a synthesized excitation signal comprises:

performing, on the random noise by using a pre-emphasis factor, a pre-emphasis operation for enhancing a high frequency part of the random noise, so as to obtain pre-emphasis noise;

weighting the high band excitation signal and the pre-emphasis noise by using the voiced degree factor, so as to generate a pre-emphasis excitation signal; and

performing, on the pre-emphasis excitation signal by using a de-emphasis factor, a de-emphasis operation for lowering a high frequency part of the pre-emphasis excitation signal, so as to obtain the synthesized excitation signal.

8. The method according to claim 7, wherein the de-emphasis factor is determined based on the pre-emphasis factor and a proportion of the pre-emphasis noise in the pre-emphasis excitation signal.

9. An audio signal decoding method, comprising:

distinguishing a low frequency encoding parameter and a high frequency encoding parameter in encoded information;

decoding the low frequency encoding parameter to obtain a low band signal;

calculating a voiced degree factor according to the low frequency encoding parameter, and predicting a high band excitation signal according to the low frequency encoding parameter, wherein the voiced degree factor is used to indicate a degree of a voiced characteristic presented by a high band signal;

weighting the high band excitation signal and random noise by using the voiced degree factor, so as to obtain a synthesized excitation signal;

obtaining the high band signal based on the synthesized excitation signal and the high frequency encoding parameter; and

combining the low band signal and the high band signal to obtain a final decoded signal;

wherein the low frequency encoding parameter comprises a pitch period, and the weighting the predicted high band excitation signal and random noise by using the voiced degree factor, so as to obtain a synthesized excitation signal comprises:

modifying the voiced degree factor by using the pitch period; and

weighting the high band excitation signal and the random noise by using a modified voiced degree factor, so as to obtain the synthesized excitation signal.

10. An audio signal decoding method, comprising:

distinguishing a low frequency encoding parameter and a high frequency encoding parameter in encoded information;

decoding the low frequency encoding parameter to obtain a low band signal;

calculating a voiced degree factor according to the low frequency encoding parameter, and predicting a high band excitation signal according to the low

frequency encoding parameter, wherein the voiced degree factor is used to indicate a degree of a voiced characteristic presented by a high band signal;

weighting the high band excitation signal and random noise by using the voiced degree factor, so as to obtain a synthesized excitation signal;

obtaining the high band signal based on the synthesized excitation signal and the high frequency encoding parameter; and

combining the low band signal and the high band signal to obtain a final decoded signal;

wherein the low frequency encoding parameter comprises an algebraic codebook, an algebraic codebook gain, an adaptive codebook, an adaptive codebook gain, and a pitch period, and the predicting a high band excitation signal according to the low frequency encoding parameter comprises:

modifying the voiced degree factor by using the pitch period; and

weighting the algebraic codebook and the random noise by using a modified voiced degree factor, so as to obtain a weighting result, and adding a product of the weighting result and the algebraic codebook gain and a product of the adaptive codebook and the adaptive codebook gain, so as to predict the high band excitation signal.

11. The method according to claim 10, wherein the modifying the voiced degree factor by using the pitch period is performed according to the following formula:

$$voice_fac_A = voice_fac * \gamma$$

$$\gamma = \begin{cases} -a1 * T0 + b1 & T0 \le threshold_\min \\ a2 * T0 + b2 & threshold_\min \le T0 \le threshold_\max \\ 1 & T0 \ge threshold_\max \end{cases}$$

wherein voice_fac is the voiced degree factor, T0 is the pitch period, a1, a2, and

$b1 > 0$, $b2 \geq 0$, threshold_min and threshold_max are respectively a preset minimum value and a preset maximum value of the pitch period, and voice_fac_A is the modified voiced degree factor.

12. The method according to claim9, wherein the modifying the voiced degree factor by using the pitch period is performed according to the following formula:

$$voice_fac_A = voice_fac * \gamma$$

$$\gamma = \begin{cases} -a1 * T0 + b1 & T0 \leq threshold_min \\ a2 * T0 + b2 & threshold_min \leq T0 \leq threshold_max \\ 1 & T0 \geq threshold_max \end{cases}$$

wherein voice_fac is the voiced degree factor, T0 is the pitch period, a1, a2, and $b1 > 0$, $b2 \geq 0$, threshold_min and threshold_max are respectively a preset minimum value and a preset maximum value of the pitch period, and voice_fac_A is the modified voiced degree factor.

13. An audio signal encoding apparatus, comprising:

a division unit, configured to divide a to-be-encoded time domain signal into a low band signal and a high band signal;

a low frequency encoding unit, configured to encode the low band signal to obtain a low frequency encoding parameter;

a calculation unit, configured to calculate a voiced degree factor according to the low frequency encoding parameter, wherein the voiced degree factor is used to indicate a degree of a voiced characteristic presented by the high band signal;

a prediction unit, configured to predict a high band excitation signal according to the low frequency encoding parameter;

a synthesizing unit, configured to weight the high band excitation signal and random noise by using the voiced degree factor, so as to obtain a synthesized

excitation signal; and

a high frequency encoding unit, configured to obtain a high frequency encoding parameter based on the synthesized excitation signal and the high band signal;

wherein the synthesizing unit comprises:

a pre-emphasis component, configured to perform, on the random noise by using a pre-emphasis factor, a pre-emphasis operation for enhancing a high frequency part of the random noise, so as to obtain pre-emphasis noise;

a weighting component, configured to weight the high band excitation signal and the pre-emphasis noise by using the voiced degree factor, so as to generate a pre-emphasis excitation signal; and

a de-emphasis component, configured to perform, on the pre-emphasis excitation signal by using a de-emphasis factor, a de-emphasis operation for lowering a high frequency part of the pre-emphasis excitation signal, so as to obtain the synthesized excitation signal.

14. The apparatus according to claim 13, wherein the de-emphasis factor is determined based on the pre-emphasis factor and a proportion of the pre-emphasis noise in the pre-emphasis excitation signal.

15. An audio signal encoding apparatus, comprising:

a division unit, configured to divide a to-be-encoded time domain signal into a low band signal and a high band signal;

a low frequency encoding unit, configured to encode the low band signal to obtain a low frequency encoding parameter;

a calculation unit, configured to calculate a voiced degree factor according to

the low frequency encoding parameter, wherein the voiced degree factor is used to indicate a degree of a voiced characteristic presented by the high band signal;

a prediction unit, configured to predict a high band excitation signal according to the low frequency encoding parameter;

a synthesizing unit, configured to weight the high band excitation signal and random noise by using the voiced degree factor, so as to obtain a synthesized excitation signal; and

a high frequency encoding unit, configured to obtain a high frequency encoding parameter based on the synthesized excitation signal and the high band signal;

wherein the low frequency encoding parameter comprises a pitch period, and the synthesizing unit comprises:

a first modification component, configured to modify the voiced degree factor by using the pitch period; and

a weighting component, configured to weight the high band excitation signal and the random noise by using a modified voiced degree factor, so as to obtain the synthesized excitation signal.

16. An audio signal encoding apparatus, comprising:

a division unit, configured to divide a to-be-encoded time domain signal into a low band signal and a high band signal;

a low frequency encoding unit, configured to encode the low band signal to obtain a low frequency encoding parameter;

a calculation unit, configured to calculate a voiced degree factor according to the low frequency encoding parameter, wherein the voiced degree factor is used to

indicate a degree of a voiced characteristic presented by the high band signal;

a prediction unit, configured to predict a high band excitation signal according to the low frequency encoding parameter;

a synthesizing unit, configured to weight the high band excitation signal and random noise by using the voiced degree factor, so as to obtain a synthesized excitation signal; and

a high frequency encoding unit, configured to obtain a high frequency encoding parameter based on the synthesized excitation signal and the high band signal; wherein the low frequency encoding parameter comprises an algebraic codebook, an algebraic codebook gain, an adaptive codebook, an adaptive codebook gain, and a pitch period, and the prediction unit comprises:

a second modification component, configured to modify the voiced degree factor by using the pitch period; and

a prediction component, configured to weight the algebraic codebook and the random noise by using a modified voiced degree factor, so as to obtain a weighting result, and add a product of the weighting result and the algebraic codebook gain and a product of the adaptive codebook and the adaptive codebook gain, so as to predict the high band excitation signal.

17. The apparatus according to claim 15, wherein modify the voiced degree factor according to the following formula:

$$voice_fac_A = voice_fac * \gamma$$

$$\gamma = \begin{cases} -a1 * T0 + b1 & T0 \le threshold_\min \\ a2 * T0 + b2 & threshold_\min \le T0 \le threshold_\max \\ 1 & T0 \ge threshold_\max \end{cases}$$

wherein voice_fac is the voiced degree factor, T0 is the pitch period, a1, a2, and

$b1 > 0$, $b2 \geq 0$, threshold_min and threshold_max are respectively a preset minimum value and a preset maximum value of the pitch period, and voice_fac_A is the modified voiced degree factor.

18. The apparatus according to claim 16, wherein modify the voiced degree factor according to the following formula:

$$voice_fac_A = voice_fac * \gamma$$

$$\gamma = \begin{cases} -a1 * T0 + b1 & T0 \leq threshold_min \\ a2 * T0 + b2 & threshold_min \leq T0 \leq threshold_max \\ 1 & T0 \geq threshold_max \end{cases}$$

wherein voice_fac is the voiced degree factor, T0 is the pitch period, a1, a2, and $b1 > 0$, $b2 \geq 0$, threshold_min and threshold_max are respectively a preset minimum value and a preset maximum value of the pitch period, and voice_fac_A is the modified voiced degree factor.

19. An audio signal decoding apparatus, comprising:

a distinguishing unit, configured to distinguish a low frequency encoding parameter and a high frequency encoding parameter in encoded information;

a low frequency decoding unit, configured to decode the low frequency encoding parameter to obtain a low band signal;

a calculation unit, configured to calculate a voiced degree factor according to the low frequency encoding parameter, wherein the voiced degree factor is used to indicate a degree of a voiced characteristic presented by a high band signal;

a prediction unit, configured to predict a high band excitation signal according to the low frequency encoding parameter;

a synthesizing unit, configured to weight the high band excitation signal and random noise by using the voiced degree factor, so as to obtain a synthesized

excitation signal;

a high frequency decoding unit, configured to obtain the high band signal based on the synthesized excitation signal and the high frequency encoding parameter; and

a combining unit, configured to combine the low band signal and the high band signal to obtain a final decoded signal;

wherein the synthesizing unit comprises:

a pre-emphasis component, configured to perform, on the random noise by using a pre-emphasis factor, a pre-emphasis operation for enhancing a high frequency part of the random noise, so as to obtain pre-emphasis noise;

a weighting component, configured to weight the high band excitation signal and the pre-emphasis noise by using the voiced degree factor, so as to generate a pre-emphasis excitation signal; and

a de-emphasis component, configured to perform, on the pre-emphasis excitation signal by using a de-emphasis factor, a de-emphasis operation for lowering a high frequency part of the pre-emphasis excitation signal, so as to obtain the synthesized excitation signal.

20. The apparatus according to claim 19, wherein the de-emphasis factor is determined based on the pre-emphasis factor and a proportion of the pre-emphasis noise in the pre-emphasis excitation signal.

21. An audio signal decoding apparatus, comprising:

a distinguishing unit, configured to distinguish a low frequency encoding parameter and a high frequency encoding parameter in encoded information;

a low frequency decoding unit, configured to decode the low frequency

encoding parameter to obtain a low band signal;

a calculation unit, configured to calculate a voiced degree factor according to the low frequency encoding parameter, wherein the voiced degree factor is used to indicate a degree of a voiced characteristic presented by a high band signal;

a prediction unit, configured to predict a high band excitation signal according to the low frequency encoding parameter;

a synthesizing unit, configured to weight the high band excitation signal and random noise by using the voiced degree factor, so as to obtain a synthesized excitation signal;

a high frequency decoding unit, configured to obtain the high band signal based on the synthesized excitation signal and the high frequency encoding parameter; and

a combining unit, configured to combine the low band signal and the high band signal to obtain a final decoded signal; wherein the low frequency encoding parameter comprises a pitch period, and the synthesizing unit comprises:

a first modification component, configured to modify the voiced degree factor by using the pitch period; and

a weighting component, configured to weight the high band excitation signal and the random noise by using a modified voiced degree factor, so as to obtain the synthesized excitation signal.

22. An audio signal decoding apparatus, comprising:

a distinguishing unit, configured to distinguish a low frequency encoding parameter and a high frequency encoding parameter in encoded information;

a low frequency decoding unit, configured to decode the low frequency

encoding parameter to obtain a low band signal;

a calculation unit, configured to calculate a voiced degree factor according to the low frequency encoding parameter, wherein the voiced degree factor is used to indicate a degree of a voiced characteristic presented by a high band signal;

a prediction unit, configured to predict a high band excitation signal according to the low frequency encoding parameter;

a synthesizing unit, configured to weight the high band excitation signal and random noise by using the voiced degree factor, so as to obtain a synthesized excitation signal;

a high frequency decoding unit, configured to obtain the high band signal based on the synthesized excitation signal and the high frequency encoding parameter; and

a combining unit, configured to combine the low band signal and the high band signal to obtain a final decoded signal; wherein the low frequency encoding parameter comprises an algebraic codebook, an algebraic codebook gain, an adaptive codebook, an adaptive codebook gain, and a pitch period, and the prediction unit comprises:

a second modification component, configured to modify the voiced degree factor by using the pitch period; and

a prediction component, configured to weight the algebraic codebook and the random noise by using a modified voiced degree factor, so as to obtain a weighting result, and add a product of the weighting result and the algebraic codebook gain and a product of the adaptive codebook and the adaptive codebook gain, so as to predict the high band excitation signal.

23. The apparatus according to claim 21, wherein modify the voiced degree factor according to the following formula:

$$voice_fac_A = voice_fac * \gamma$$

$$\gamma = \begin{cases} -a1 * T0 + b1 & T0 \le threshold_\min \\ a2 * T0 + b2 & threshold_\min \le T0 \le threshold_\max \\ 1 & T0 \ge threshold_\max \end{cases}$$

wherein voice_fac is the voiced degree factor, T0 is the pitch period, a1, a2, and $b1 > 0$, $b2 \ge 0$, threshold_min and threshold_max are respectively a preset minimum value and a preset maximum value of the pitch period, and voice_fac_A is the modified voiced degree factor.

24. The apparatus according to claim 22, wherein modify the voiced degree factor according to the following formula:

$$voice_fac_A = voice_fac * \gamma$$

$$\gamma = \begin{cases} -a1 * T0 + b1 & T0 \le threshold_\min \\ a2 * T0 + b2 & threshold_\min \le T0 \le threshold_\max \\ 1 & T0 \ge threshold_\max \end{cases}$$

wherein voice_fac is the voiced degree factor, T0 is the pitch period, a1, a2, and $b1 > 0$, $b2 \ge 0$, threshold_min and threshold_max are respectively a preset minimum value and a preset maximum value of the pitch period, and voice_fac_A is the modified voiced degree factor.

25.A computer-readable storage medium having a program recorded thereon, wherein the program makes a computer execute the method of any claims 1-6.

26. A computer-readable storage medium having a program recorded thereon, wherein the program makes a computer execute the method of any claims 7-12.