Sudden switching between signals of different bandwidths causes obvious aural discomfort in human ears. Besides, because updating of states of filters during time-frequency transform or frequency-time transform generally requires the use of a parameter between consecutive frames, when some proper processing is not performed during bandwidth switching, an error may occur during the updating of these states, which causes some phenomena of abrupt energy changes and deterioration of aural quality.

SUMMARY

An objective of embodiments of the present invention is to provide a speech/audio signal processing method and apparatus, so as to improve aural comfort during bandwidth switching of speech/audio signals.

According to an embodiment of the present invention, a speech/audio signal processing method includes:

when a speech/audio signal switches from a wide frequency signal to a narrow frequency signal, obtaining an initial high frequency signal corresponding to a current frame of speech/audio signal;

obtaining a time-domain global gain parameter of the high frequency signal according to a spectrum tilt parameter of the current frame of speech/audio signal and a correlation between a current frame of narrow frequency signal and a historical frame of narrow frequency signal;

correcting the initial high frequency signal by using the time-domain global gain parameter, to obtain a corrected high frequency time-domain signal; and

synthesizing a current frame of narrow frequency time-domain signal and the corrected high frequency time-domain signal and outputting the synthesized signal.

According to another embodiment of the present invention, a speech/audio signal processing method includes:

when a speech/audio signal switches bandwidth, obtaining an initial high frequency signal corresponding to a current frame of speech/audio signal;

obtaining a time-domain global gain parameter of the initial high frequency signal;

performing weighting processing on an energy ratio and the time-domain global gain parameter, and using an obtained weighted value as a predicted global gain parameter, where the energy ratio is a ratio between energy of a historical frame of high frequency time-domain signal and energy of a current frame of initial high frequency signal;

correcting the initial high frequency signal by using the predicted global gain parameter, to obtain a corrected high frequency time-domain signal; and

synthesizing a current frame of narrow frequency time-domain signal and the corrected high frequency time-domain signal and outputting the synthesized signal.

According to another embodiment of the present invention, a speech/audio signal processing apparatus includes:

a predicting unit, configured to: when a speech/audio signal switches from a wide frequency signal to a narrow frequency signal, obtain an initial high frequency signal corresponding to a current frame of speech/audio signal;

a parameter obtaining unit, configured to obtain a time-domain global gain parameter of the high frequency signal according to a spectrum tilt parameter of the current frame of speech/audio signal and a correlation between a current frame of narrow frequency signal and a historical frame of narrow frequency signal;

a correcting unit, configured to correct the initial high frequency signal by using the predicted global gain parameter, to obtain a corrected high frequency time-domain signal; and

a synthesizing unit, configured to synthesize a current frame of narrow frequency time-domain signal and the corrected high frequency time-domain signal and output the synthesized signal.

According to another embodiment of the present invention, a speech/audio signal processing apparatus includes:

an acquiring unit, configured to: when a speech/audio signal switches bandwidth, obtain an initial high frequency signal corresponding to a current frame of speech/audio signal;

a parameter obtaining unit, configured to obtain a time-domain global gain parameter corresponding to the initial high frequency signal;

a weighting processing unit, configured to perform weighting processing on an energy ratio and the time-domain global gain parameter, and use an obtained weighted value as a predicted global gain parameter, where the energy ratio is a ratio between energy of a historical frame of high frequency time-domain signal and energy of a current frame of initial high frequency signal;

a correcting unit, configured to correct the initial high frequency signal by using the predicted global gain parameter, to obtain a corrected high frequency time-domain signal; and

Sudden switching between signals of different bandwidths causes obvious aural discomfort in human ears. Besides, because updating of states of filters during time-frequency transform or frequency-time transform generally requires the use of a parameter between consecutive frames, when some proper processing is not performed during bandwidth switching, an error may occur during the updating of these states, which causes some phenomena of abrupt energy changes and deterioration of aural quality.

SUMMARY

An objective of the present invention is to provide a speech/audio signal processing method and apparatus, so as to improve aural comfort during bandwidth switching of speech/audio signals.

According to a first aspect of the present invention, a speech/audio signal processing method includes:

when a speech/audio signal switches from a wide frequency signal to a narrow frequency signal, obtaining an initial high frequency signal corresponding to a current frame of speech/audio signal;

obtaining a time-domain global gain parameter of the high frequency signal according to a spectrum tilt parameter of the current frame of speech/audio signal and a correlation between a current frame of narrow frequency signal and a historical frame of narrow frequency signal;

correcting the initial high frequency signal by using the time-domain global gain parameter, to obtain a corrected high frequency time-domain signal; and

synthesizing a current frame of narrow frequency time-domain signal and the corrected high frequency time-domain signal and outputting the synthesized signal.

In a first possible implementation manner of the first aspect, wherein the obtaining a time-domain global gain parameter of the high frequency signal according to a spectrum tilt parameter of the current frame of speech/audio signal and a correlation between a current frame of narrow frequency signal and a historical frame of narrow frequency signal comprises:

classifying the current frame of speech/audio signal as a first type of signal or a second type of signal according to the spectrum tilt parameter of the current frame of speech/audio signal and the correlation between the current frame of narrow frequency signal and the historical frame of narrow frequency signal;

when the current frame of speech/audio signal is a first type of signal, limiting the

spectrum tilt parameter to less than or equal to a first predetermined value, to obtain a spectrum tilt parameter limit value;

when the current frame of speech/audio signal is a second type of signal, limiting the spectrum tilt parameter to a value in a first range, to obtain a spectrum tilt parameter limit value; and

using the spectrum tilt parameter limit value as the time-domain global gain parameter of the high frequency signal.

With reference to the first possible implementation manner of the first aspect, in a second possible implementation manner, wherein the first type of signal is a fricative signal, and the second type of signal is a non-fricative signal; when the spectrum tilt parameter tilt>5 and a correlation parameter cor is less than a given value, the narrow frequency signal is classified as a fricative signal, the rest being non-fricative signals; the first predetermined value is 8; and the first preset range is [0.5, 1].

With reference to anyone of the first aspect, the first possible implementation manner of the first aspect and the second possible implementation manner of the first aspect, in a third possible implementation manner, wherein the correcting the initial high frequency signal by using the time-domain global gain parameter, to obtain a corrected high frequency time-domain signal comprises:

performing weighting processing on an energy ratio and the time-domain global gain parameter, and using an obtained weighted value as a predicted global gain parameter, wherein the energy ratio is a ratio between energy of a historical frame of high frequency time-domain signal and energy of a current frame of initial high frequency signal; and

correcting the initial high frequency signal by using the predicted global gain parameter.

With reference to anyone of the first aspect, the first possible implementation manner of the first aspect and the second possible implementation manner of the first aspect, in a fourth possible implementation manner, further comprising:

obtaining a time-domain envelope parameter corresponding to the initial high frequency signal, wherein

the correcting the initial high frequency signal by using the time-domain global gain parameter comprises:

correcting the initial high frequency signal by using the time-domain envelope parameter and the time-domain global gain parameter.

According to a second aspect of the present invention, a speech/audio signal

processing method includes:

when a speech/audio signal switches bandwidth, obtaining an initial high frequency signal corresponding to a current frame of speech/audio signal;

obtaining a time-domain global gain parameter of the initial high frequency signal;

performing weighting processing on an energy ratio and the time-domain global gain parameter, and using an obtained weighted value as a predicted global gain parameter, where the energy ratio is a ratio between energy of a historical frame of high frequency time-domain signal and energy of a current frame of initial high frequency signal;

correcting the initial high frequency signal by using the predicted global gain parameter, to obtain a corrected high frequency time-domain signal; and

synthesizing a current frame of narrow frequency time-domain signal and the corrected high frequency time-domain signal and outputting the synthesized signal.

In a first possible implementation manner of the second aspect, wherein the bandwidth switching is switching from a wide frequency signal to a narrow frequency signal, and the obtaining a time-domain global gain parameter of the initial high frequency signal comprises:

obtaining a time-domain global gain parameter of the high frequency signal according to a spectrum tilt parameter of the current frame of speech/audio signal and a correlation between a current frame of narrow frequency signal and a historical frame of narrow frequency signal.

With reference to the first possible implementation manner of the first aspect, in a second possible implementation manner, wherein the obtaining a time-domain global gain parameter of the high frequency signal according to a spectrum tilt parameter of a current frame of speech/audio signal and a correlation between a current frame of narrow frequency signal and a historical frame of narrow frequency signal comprises:

classifying the current frame of speech/audio signal as a first type of signal or a second type of signal according to the spectrum tilt parameter of the current frame of speech/audio signal and the correlation between the current frame of narrow frequency signal and the historical frame of narrow frequency signal;

when the current frame of speech/audio signal is a first type of signal, limiting the spectrum tilt parameter to less than or equal to a first predetermined value, to obtain a spectrum tilt parameter limit value;

when the current frame of speech/audio signal is a second type of signal, limiting

the spectrum tilt parameter to a value in a first range, to obtain a spectrum tilt parameter limit value; and

using the spectrum tilt parameter limit value as the time-domain global gain parameter of the high frequency signal.

With reference to the second possible implementation manner of the first aspect, in a third possible implementation manner, wherein the first type of signal is a fricative signal, and the second type of signal is a non-fricative signal; when the spectrum tilt parameter tilt>5 and a correlation parameter cor is less than a given value, the narrow frequency signal is classified as a fricative, the rest being non-fricatives; the first predetermined value is 8; and the first preset range is [0.5, 1].

In a fourth possible implementation manner of the second aspect, wherein the bandwidth switching is switching from a wide frequency signal to a narrow frequency signal, and the obtaining an initial high frequency signal corresponding to a current frame of speech/audio signal comprises:

predicting a high frequency excitation signal according to the current frame of speech/audio signal;

predicting an LPC coefficient of the high frequency signal; and

synthesizing the high frequency excitation signal and the LPC coefficient of the high frequency signal, to obtain the predicted high frequency signal.

In a fifth possible implementation manner of the second aspect, wherein the bandwidth switching is switching from a narrow frequency signal to a wide frequency signal, and the method further comprises:

when narrowband signals of the current frame of speech/audio signal and a previous frame of speech/audio signal have a predetermined correlation, using a value obtained by attenuating, according to a step size, a weighting factor alfa of an energy ratio corresponding to the previous frame of speech/audio signal as a weighting factor of an energy ratio corresponding to the current audio frame, wherein the attenuation is performed frame by frame until alfa is 0.

According to a third aspect of the present invention, a speech/audio signal processing apparatus includes:

a predicting unit, configured to: when a speech/audio signal switches from a wide frequency signal to a narrow frequency signal, obtain an initial high frequency signal corresponding to a current frame of speech/audio signal;

a parameter obtaining unit, configured to obtain a time-domain global gain

parameter of the high frequency signal according to a spectrum tilt parameter of the current frame of speech/audio signal and a correlation between a current frame of narrow frequency signal and a historical frame of narrow frequency signal;

a correcting unit, configured to correct the initial high frequency signal by using the predicted global gain parameter, to obtain a corrected high frequency time-domain signal; and

a synthesizing unit, configured to synthesize a current frame of narrow frequency time-domain signal and the corrected high frequency time-domain signal and output the synthesized signal.

In a first possible implementation manner of the third aspect, wherein the parameter obtaining unit comprises:

a classifying unit, configured to classify the current frame of speech/audio signal as a first type of signal or a second type of signal according to the spectrum tilt parameter of the current frame of speech/audio signal and the correlation between the current frame of speech/audio signal and the historical frame of narrow frequency signal;

a first limiting unit, configured to: when the current frame of speech/audio signal is a first type of signal, limit the spectrum tilt parameter to less than or equal to a first predetermined value, to obtain a spectrum tilt parameter limit value, and use the spectrum tilt parameter limit value as the time-domain global gain parameter of the high frequency signal; and

a second limiting unit, configured to: when the current frame of speech/audio signal is a second type of signal, limit the spectrum tilt parameter to a value in a first range, to obtain a spectrum tilt parameter limit value, and use the spectrum tilt parameter limit value as the time-domain global gain parameter of the high frequency signal.

With reference to the first possible implementation manner of the third aspect, in a second possible implementation manner, wherein the first type of signal is a fricative signal, and the second type of signal is a non-fricative signal; when the spectrum tilt parameter tilt>5 and a correlation parameter cor is less than a given value, the narrow frequency signal is classified as a fricative, the rest being non-fricatives; the first predetermined value is 8; and the first preset range is [0.5, 1].

With reference to anyone of the third aspect, the first possible implementation manner of the third aspect and the second possible implementation manner of the third aspect, in a third possible implementation manner, further comprising:

a weighting processing unit, configured to perform weighting processing on an

energy ratio and the time-domain global gain parameter, and use an obtained weighted value as a predicted global gain parameter, wherein the energy ratio is a ratio between energy of a historical frame of high frequency time-domain signal and energy of a current frame of initial high frequency signal, wherein

the correcting unit is configured to correct the initial high frequency signal by using the predicted global gain parameter, to obtain the corrected high frequency time-domain signal.

With reference to anyone of the third aspect, the first possible implementation manner of the third aspect and the second possible implementation manner of the third aspect, in a fourth possible implementation manner, wherein

the parameter obtaining unit is further configured to obtain a time-domain envelope parameter corresponding to the initial high frequency signal; and

the correcting unit is configured to correct the initial high frequency signal by using the time-domain envelope parameter and the time-domain global gain parameter.

According to a fourth aspect of the present invention, a speech/audio signal processing apparatus includes:

an acquiring unit, configured to: when a speech/audio signal switches bandwidth, obtain an initial high frequency signal corresponding to a current frame of speech/audio signal;

a parameter obtaining unit, configured to obtain a time-domain global gain parameter corresponding to the initial high frequency signal;

a weighting processing unit, configured to perform weighting processing on an energy ratio and the time-domain global gain parameter, and use an obtained weighted value as a predicted global gain parameter, where the energy ratio is a ratio between energy of a historical frame of high frequency time-domain signal and energy of a current frame of initial high frequency signal;

a correcting unit, configured to correct the initial high frequency signal by using the predicted global gain parameter, to obtain a corrected high frequency time-domain signal; and

a synthesizing unit, configured to synthesize a current frame of narrow frequency time-domain signal and the corrected high frequency time-domain signal output the synthesized signal.

In a first possible implementation manner of the fourth aspect, wherein the bandwidth switching is switching from a wide frequency signal to a narrow frequency signal,

and the parameter obtaining unit comprises:

a global gain parameter obtaining unit, configured to obtain the time-domain global gain parameter of the high frequency signal according to a spectrum tilt parameter of the current frame of speech/audio signal and a correlation between a current frame of speech/audio signal and a historical frame of narrow frequency signal.

With reference to the first possible implementation manner of the fourth aspect, in a second possible implementation manner, wherein the global gain parameter obtaining unit comprises:

a classifying unit, configured to classify the current frame of speech/audio signal as a first type of signal or a second type of signal according to the spectrum tilt parameter of the current frame of speech/audio signal and the correlation between the current frame of speech/audio signal and the historical frame of narrow frequency signal;

a first limiting unit, configured to: when the current frame of speech/audio signal is a first type of signal, limit the spectrum tilt parameter to less than or equal to a first predetermined value, to obtain a spectrum tilt parameter limit value, and use the spectrum tilt parameter limit value as the time-domain global gain parameter of the high frequency signal; and

a second limiting unit, configured to: when the current frame of speech/audio signal is a second type of signal, limit the spectrum tilt parameter to a value in a first range, to obtain a spectrum tilt parameter limit value, and use the spectrum tilt parameter limit value as the time-domain global gain parameter of the high frequency signal.

With reference to the second possible implementation manner of the fourth aspect, in a third possible implementation manner, wherein the first type of signal is a fricative signal, and the second type of signal is a non-fricative signal; when the spectrum tilt parameter tilt>5 and a correlation parameter cor is less than a given value, the narrow frequency signal is classified as a fricative, the rest being non-fricatives; the first predetermined value is 8; and the first preset range is [0.5, 1].

With reference to anyone of the fourth aspect, the first possible implementation manner of the fourth aspect and the second possible implementation manner of the fourth aspect, in a fourth possible implementation manner, wherein the bandwidth switching is switching from a wide frequency signal to a narrow frequency signal, and the apparatus further comprises:

a time-domain envelope obtaining unit, configured to use a series of preset values as a high frequency time-domain envelope parameter of the current frame of speech/audio

signal; and

the correcting unit is configured to correct the initial high frequency signal by using the time-domain envelope parameter and the predicted global gain parameter, to obtain the corrected high frequency time-domain signal.

With reference to anyone of the fourth aspect, the first possible implementation manner of the fourth aspect and the second possible implementation manner of the fourth aspect, in a fifth possible implementation manner, wherein the acquiring unit comprises:

an excitation signal obtaining unit, configured to predict an excitation signal of the high frequency signal according to the current frame of speech/audio signal;

an LPC coefficient obtaining unit, configured to predict an LPC coefficient of the high frequency signal; and

a synthesizing unit, configured to synthesize the excitation signal of the high frequency signal and the LPC coefficient of the high frequency signal, to obtain the predicted high frequency signal.

With reference to anyone of the fourth aspect, the first possible implementation manner of the fourth aspect and the second possible implementation manner of the fourth aspect, in a sixth possible implementation manner, wherein the bandwidth switching is switching from a narrow frequency signal to a wide frequency signal, and the apparatus further comprises:

a weighting factor setting unit, configured to: when narrowband signals of the current frame of speech/audio signal and a previous frame of speech/audio signal have a predetermined correlation, use a value obtained by attenuating, according to a step size, a weighting factor alfa of an energy ratio corresponding to the previous frame of speech/audio signal as a weighting factor of an energy ratio corresponding to the current audio frame, wherein the attenuation is performed frame by frame until alfa is 0.

a synthesizing unit, configured to synthesize a current frame of narrow frequency time-domain signal and the corrected high frequency time-domain signal output the synthesized signal.

~~In the embodiments of~~ According to the present invention, during switching between a wide frequency band and a narrow frequency band, a high frequency signal is corrected, so as to implement a smooth transition of the high frequency signal between the wide frequency band and the narrow frequency band, thereby effectively eliminating aural discomfort caused by the switching between the wide frequency band and the narrow frequency band; in addition, because a bandwidth switching algorithm and a coding/decoding algorithm of the high frequency signal before switching are in a same signal domain, it not only ensures that no extra delay is added and the algorithm is simple, it also ensures performance of an output signal.

BRIEF DESCRIPTION OF DRAWINGS

To describe the technical solutions in the embodiments of the present invention or in the prior art more clearly, the following briefly introduces the accompanying drawings required for describing the embodiments or the prior art. Apparently, the accompanying drawings in the following description show merely some embodiments of the present invention, and a person of ordinary skill in the art may still derive other drawings from these accompanying drawings without creative efforts.

FIG. 1 is a schematic flowchart of an embodiment of a speech/audio signal processing method according to the present invention;

FIG. 2 is a schematic flowchart of another embodiment of a speech/audio signal processing method according to the present invention;

FIG. 3 is a schematic flowchart of another embodiment of a speech/audio signal processing method according to the present invention;

FIG. 4 is a schematic flowchart of another embodiment of a speech/audio signal processing method according to the present invention;

FIG. 5 is a schematic structural diagram of an embodiment of a speech/audio signal processing apparatus according to the present invention;

FIG. 6 is a schematic structural diagram of an embodiment of a speech/audio signal processing apparatus according to the present invention;

FIG. 7 is a schematic structural diagram of an embodiment of a parameter obtaining unit according to the present invention;

CLAIMS

~~What is claimed is:~~

1. A speech/audio signal processing method, comprising:

when a speech/audio signal switches from a wide frequency signal to a narrow frequency signal, obtaining an initial high frequency signal corresponding to a current frame of speech/audio signal;

obtaining a time-domain global gain parameter of the high frequency signal according to a spectrum tilt parameter of the current frame of speech/audio signal and a correlation between a current frame of narrow frequency signal and a historical frame of narrow frequency signal;

correcting the initial high frequency signal by using the time-domain global gain parameter, to obtain a corrected high frequency time-domain signal; and

synthesizing a current frame of narrow frequency time-domain signal and the corrected high frequency time-domain signal and outputting the synthesized signal.

2. The method according to claim 1, wherein the obtaining a time-domain global gain parameter of the high frequency signal according to a spectrum tilt parameter of the current frame of speech/audio signal and a correlation between a current frame of narrow frequency signal and a historical frame of narrow frequency signal comprises:

classifying the current frame of speech/audio signal as a first type of signal or a second type of signal according to the spectrum tilt parameter of the current frame of speech/audio signal and the correlation between the current frame of narrow frequency signal and the historical frame of narrow frequency signal;

when the current frame of speech/audio signal is a first type of signal, limiting the spectrum tilt parameter to less than or equal to a first predetermined value, to obtain a spectrum tilt parameter limit value;

when the current frame of speech/audio signal is a second type of signal, limiting the spectrum tilt parameter to a value in a first range, to obtain a spectrum tilt parameter limit value; and

using the spectrum tilt parameter limit value as the time-domain global gain parameter of the high frequency signal.

3. The method according to claim 2, wherein the first type of signal is a fricative signal, and the second type of signal is a non-fricative signal; when the spectrum tilt parameter tilt>5 and a correlation parameter cor is less than a given value, the narrow frequency signal is

classified as a fricative signal, the rest being non-fricative signals; the first predetermined value is 8; and the first preset range is [0.5, 1].

4. The method according to any one of claims 1 to 3, wherein the correcting the initial high frequency signal by using the time-domain global gain parameter, to obtain a corrected high frequency time-domain signal comprises:

performing weighting processing on an energy ratio and the time-domain global gain parameter, and using an obtained weighted value as a predicted global gain parameter, wherein the energy ratio is a ratio between energy of a historical frame of high frequency time-domain signal and energy of a current frame of initial high frequency signal; and

correcting the initial high frequency signal by using the predicted global gain parameter.

5. The method according to any one of claims 1 to 3, further comprising:

obtaining a time-domain envelope parameter corresponding to the initial high frequency signal, wherein

the correcting the initial high frequency signal by using the time-domain global gain parameter comprises:

correcting the initial high frequency signal by using the time-domain envelope parameter and the time-domain global gain parameter.

6. A speech/audio signal processing method, comprising:

when a speech/audio signal switches bandwidth, obtaining an initial high frequency signal corresponding to a current frame of speech/audio signal;

obtaining a time-domain global gain parameter of the initial high frequency signal;

performing weighting processing on an energy ratio and the time-domain global gain parameter, and using an obtained weighted value as a predicted global gain parameter, wherein the energy ratio is a ratio between energy of a historical frame of high frequency time-domain signal and energy of a current frame of initial high frequency signal;

correcting the initial high frequency signal by using the predicted global gain parameter, to obtain a corrected high frequency time-domain signal; and

synthesizing a current frame of narrow frequency time-domain signal and the corrected high frequency time-domain signal and outputting the synthesized signal.

7. The method according to claim 6, wherein the bandwidth switching is switching from a wide frequency signal to a narrow frequency signal, and the obtaining a time-domain global gain parameter ~~corresponding to~~of the initial high frequency signal comprises:

obtaining a time-domain global gain parameter of the high frequency signal according to a spectrum tilt parameter of the current frame of speech/audio signal and a correlation

when narrowband signals of the current frame of speech/audio signal and a previous frame of speech/audio signal have a predetermined correlation, using a value obtained by attenuating, according to a ~~certain~~ step size, a weighting factor alfa of ~~the~~an energy ratio corresponding to the previous frame of speech/audio signal as a weighting factor of ~~the~~ an energy ratio corresponding to the current audio frame, wherein the attenuation is performed frame by frame until alfa is 0.

12. A speech/audio signal processing apparatus, comprising:

a predicting unit, configured to: when a speech/audio signal switches from a wide frequency signal to a narrow frequency signal, obtain an initial high frequency signal corresponding to a current frame of speech/audio signal;

a parameter obtaining unit, configured to obtain a time-domain global gain parameter of the high frequency signal according to a spectrum tilt parameter of the current frame of speech/audio signal and a correlation between a current frame of narrow frequency signal and a historical frame of narrow frequency signal;

a correcting unit, configured to correct the initial high frequency signal by using the predicted global gain parameter, to obtain a corrected high frequency time-domain signal; and

a synthesizing unit, configured to synthesize a current frame of narrow frequency time-domain signal and the corrected high frequency time-domain signal and output the synthesized signal.

13. The apparatus according to claim 12, wherein the parameter obtaining unit comprises:

a classifying unit, configured to classify the current frame of speech/audio signal as a first type of signal or a second type of signal according to the spectrum tilt parameter of the current frame of speech/audio signal and the correlation between the current frame of speech/audio signal and the historical frame of narrow frequency signal;

a first limiting unit, configured to: when the current frame of speech/audio signal is a first type of signal, limit the spectrum tilt parameter to less than or equal to a first predetermined value, to obtain a spectrum tilt parameter limit value, and use the spectrum tilt parameter limit value as the time-domain global gain parameter of the high frequency signal; and

a second limiting unit, configured to: when the current frame of speech/audio signal is a second type of signal, limit the spectrum tilt parameter to a value in a first range, to obtain a spectrum tilt parameter limit value, and use the spectrum tilt parameter limit value as the

synthesized signal.

18. The apparatus according to claim 17, wherein the bandwidth switching is switching from a wide frequency signal to a narrow frequency signal, and the parameter obtaining unit comprises:

a global gain parameter obtaining unit, configured to obtain the time-domain global gain parameter of the high frequency signal according to a spectrum tilt parameter of the current frame of speech/audio signal and a correlation between a current frame of speech/audio signal and a historical frame of narrow frequency signal.

19. The apparatus according to claim 18, wherein the global gain parameter obtaining unit comprises:

a classifying unit, configured to classify the current frame of speech/audio signal as a first type of signal or a second type of signal according to the spectrum tilt parameter of the current frame of speech/audio signal and the correlation between the current frame of speech/audio signal and the historical frame of narrow frequency signal;

a first limiting unit, configured to: when the current frame of speech/audio signal is a first type of signal, limit the spectrum tilt parameter to less than or equal to a first predetermined value, to obtain a spectrum tilt parameter limit value, and use the spectrum tilt parameter limit value as the time-domain global gain parameter of the high frequency signal; and

a second limiting unit, configured to: when the current frame of speech/audio signal is a second type of signal, limit the spectrum tilt parameter to a value in a first range, to obtain a spectrum tilt parameter limit value, and use the spectrum tilt parameter limit value as the time-domain global gain parameter of the high frequency signal.

20. The apparatus according to claim 19, wherein the first type of signal is a fricative signal, and the second type of signal is a non-fricative signal; when the spectrum tilt parameter tilt>5 and a correlation parameter cor is less than a given value, the narrow frequency signal is classified as a fricative, the rest being non-fricatives; the first predetermined value is 8; and the first preset range is [0.5, 1].

21. The apparatus according to any one of claims 17 to 20, wherein the bandwidth switching is switching from a ~~narrow~~ wide frequency signal to a ~~wide~~narrow frequency signal, and the apparatus further comprises:

a time-domain envelope obtaining unit, configured to use a series of preset values as a high frequency time-domain envelope parameter of the current frame of speech/audio signal; and

the correcting unit is configured to correct the initial high frequency signal by using the time-domain envelope parameter and the predicted global gain parameter, to obtain the corrected high frequency time-domain signal.

22. The apparatus according to any one of claims 17 to 20, wherein the acquiring unit comprises:

an excitation signal obtaining unit, configured to predict an excitation signal of the high frequency signal according to the current frame of speech/audio signal;

an LPC coefficient obtaining unit, configured to predict an LPC coefficient of the high frequency signal; and

a synthesizing unit, configured to synthesize the excitation signal of the high frequency signal and the LPC coefficient of the high frequency signal, to obtain the predicted high frequency signal.

23. The apparatus according to any one of claims 17 to 20, wherein the bandwidth switching is switching from a narrow frequency signal to a wide frequency signal, and the apparatus further comprises:

a weighting factor setting unit, configured to: when narrowband signals of the current frame of speech/audio signal and a previous frame of speech/audio signal have a predetermined correlation, use a value obtained by attenuating, according to a ~~certain~~ step size, a weighting factor alfa of ~~the~~ an energy ratio corresponding to the previous frame of speech/audio signal as a weighting factor of ~~the~~ an energy ratio corresponding to the current audio frame, wherein the attenuation is performed frame by frame until alfa is 0.

24. A computer-readable storage medium having a program recorded thereon; where the program makes the computer execute method of any of claims 1 to 11.