CLAIMS

1. A method for processing a speech/audio signal, wherein the method comprises:

receiving a bitstream, and decoding the bitstream, to obtain the speech/audio signal;

determining a first speech/audio signal according to the speech/audio signal, wherein the first speech/audio signal is in the speech/audio signal and noise component of the first speech/audio signal needs to be reconstructed;

determining a sign of each sample value in the first speech/audio signal and an amplitude value of each sample value in the first speech/audio signal;

determining an adaptive normalization length;

determining an adjusted amplitude value of each sample value according to the adaptive normalization length and the amplitude value of each sample value; and

determining a second speech/audio signal according to the sign of each sample value and the adjusted amplitude value of each sample value, wherein the second speech/audio signal is a signal obtained by reconstructing the noise component for the first speech/audio signal.

2. The method according to claim 1, wherein the determining an adjusted amplitude value of each sample value according to the adaptive normalization length and the amplitude value of each sample value comprises:

calculating, according to the amplitude value of each sample value and the adaptive normalization length, an average amplitude value corresponding to each sample value, and determining, according to the average amplitude value corresponding to each sample value, an amplitude disturbance value corresponding to each sample value; and

calculating the adjusted amplitude value of each sample value according to the amplitude value of each sample value and according to the amplitude disturbance value corresponding to each sample value.

3. The method according to claim 2, wherein the calculating, according to the amplitude value of each sample value and the adaptive normalization length, an average amplitude value corresponding to each sample value comprises:

determining, for each sample value and according to the adaptive normalization length, a subband to which the sample value belongs; and

calculating an average value of amplitude values of all sample values in the subband to which the sample value belongs, and using the average value obtained by means of calculation as the average amplitude value corresponding to the sample value.

4. The method according to claim 3, wherein the determining, for each sample value and

according to the adaptive normalization length, a subband to which the sample value belongs comprises:

performing subband grouping on all sample values in a preset order according to the adaptive normalization length; and for each sample value, determining a subband comprising the sample value as the subband to which the sample value belongs; or

for each sample value, determining a subband consisting of m sample values before the sample value, the sample value, and n sample values after the sample value as the subband to which the sample value belongs, wherein m and n depend on the adaptive normalization length, m is an integer not less than 0, and n is an integer not less than 0.

5. The method according to any one of claims 2 to 4, wherein the calculating the adjusted amplitude value of each sample value according to the amplitude value of each sample value and according to the amplitude disturbance value corresponding to each sample value comprises:

subtracting the amplitude disturbance value corresponding to each sample value from the amplitude value of each sample value, to obtain a difference between the amplitude value of each sample value and the amplitude disturbance value corresponding to each sample value, and using the obtained difference as the adjusted amplitude value of each sample value.

6. The method according to any one of claims 1 to 5, wherein the determining an adaptive normalization length comprises:

dividing a low frequency band signal in the speech/audio signal into N subbands, wherein N is a natural number;

calculating a peak-to-average ratio of each subband, and determining a quantity of subbands whose peak-to-average ratios are greater than a preset peak-to-average ratio threshold; and

calculating the adaptive normalization length according to a signal type of a high frequency band signal in the speech/audio signal and the quantity of the subbands.

7. The method according to claim 6, wherein the calculating the adaptive normalization length according to a signal type of a high frequency band signal in the speech/audio signal and the quantity of the subbands comprises:

calculating the adaptive normalization length according to a formula $L = K + \alpha \times M$, wherein

L is the adaptive normalization length; K is a numerical value corresponding to the signal type of the high frequency band signal in the speech/audio signal, and different signal types of high frequency band signals correspond to different numerical values K; M is the quantity of the subbands whose peak-to-average ratios are greater than the preset peak-to-average ratio threshold; and α is a constant less than 1.

8. The method according to any one of claims 1 to 5, wherein the determining an adaptive

normalization length comprises:

calculating a peak-to-average ratio of a low frequency band signal in the speech/audio signal and a peak-to-average ratio of a high frequency band signal in the speech/audio signal; and when an absolute value of a difference between the peak-to-average ratio of the low frequency band signal and the peak-to-average ratio of the high frequency band signal is less than a preset difference threshold, determining the adaptive normalization length as a preset first length value, or when an absolute value of a difference between the peak-to-average ratio of the low frequency band signal and the peak-to-average ratio of the high frequency band signal is not less than a preset difference threshold, determining the adaptive normalization length as a preset second length value, wherein the first length value is greater than the second length value; or

calculating a peak-to-average ratio of a low frequency band signal in the speech/audio signal and a peak-to-average ratio of a high frequency band signal in the speech/audio signal; and when the peak-to-average ratio of the low frequency band signal is less than the peak-to-average ratio of the high frequency band signal, determining the adaptive normalization length as a preset first length value, or when the peak-to-average ratio of the low frequency band signal is not less than the peak-to-average ratio of the high frequency band signal, determining the adaptive normalization length as a preset second length value; or

determining the adaptive normalization length according to a signal type of a high frequency band signal in the speech/audio signal, wherein different signal types of high frequency band signals correspond to different adaptive normalization lengths.

9. The method according to any one of claims 1 to 8, wherein the determining a second speech/audio signal according to the sign of each sample value and the adjusted amplitude value of each sample value comprises:

determining a new value of each sample value according to the sign and the adjusted amplitude value of each sample value, to obtain the second speech/audio signal; or

calculating a modification factor; performing modification processing on an adjusted amplitude value, which is greater than 0, in the adjusted amplitude values of the sample values according to the modification factor; and determining a new value of each sample value according to the sign of each sample value and an adjusted amplitude value that is obtained after the modification processing, to obtain the second speech/audio signal.

10. The method according to claim 9, wherein the calculating a modification factor comprises:

calculating the modification factor by using a formula $\beta = a/L$, wherein β is the modification factor, L is the adaptive normalization length, and a is a constant greater than 1.

11. The method according to claim 10, wherein the performing modification processing on an

adjusted amplitude value, which is greater than 0, in the adjusted amplitude values of the sample values according to the modification factor comprises:

performing modification processing on the adjusted amplitude value, which is greater than 0, in the adjusted amplitude values of the sample values by using the following formula:

$$Y = y \times (b - \beta);$$

wherein Y is the adjusted amplitude value obtained after the modification processing; y is the adjusted amplitude value, which is greater than 0, in the adjusted amplitude values of the sample values; and b is a constant, and $0 < b < 2$.

12. An apparatus for reconstructing a noise component of a speech/audio signal, comprising:

a bitstream processing unit, configured to receive a bitstream and decode the bitstream, to obtain the speech/audio signal;

a signal determining unit, configured to determine a first speech/audio signal according to the speech/audio signal obtained by the bitstream processing unit, wherein the first speech/audio signal is in the speech/audio signal obtained by means of decoding and noise component of the first speech/audio signal needs to be reconstructed;

a first determining unit, configured to determine a sign of each sample value in the first speech/audio signal determined by the signal determining unit and an amplitude value of each sample value in the first speech/audio signal determined by the signal determining unit;

a second determining unit, configured to determine an adaptive normalization length;

a third determining unit, configured to determine an adjusted amplitude value of each sample value according to the adaptive normalization length determined by the second determining unit and the amplitude value that is of each sample value and is determined by the first determining unit; and

a fourth determining unit, configured to determine a second speech/audio signal according to the sign that is of each sample value and is determined by the first determining unit and the adjusted amplitude value that is of each sample value and is determined by the third determining unit, wherein the second speech/audio signal is a signal obtained by reconstructing the noise component for the first speech/audio signal.

13. The apparatus according to claim 12, wherein the third determining unit comprises:

a determining subunit, configured to calculate, according to the amplitude value of each sample value and the adaptive normalization length, an average amplitude value corresponding to each sample value, and determine, according to the average amplitude value corresponding to each sample value, an amplitude disturbance value corresponding to each sample value; and

an adjusted amplitude value calculation subunit, configured to calculate the adjusted amplitude value of each sample value according to the amplitude value of each sample value and according to

the amplitude disturbance value corresponding to each sample value.

14. The apparatus according to claim 13, wherein the determining subunit comprises:

a determining module, configured to determine, for each sample value and according to the adaptive normalization length, a subband to which the sample value belongs; and

a calculation module, configured to calculate an average value of amplitude values of all sample values in the subband to which the sample value belongs, and use the average value obtained by means of calculation as the average amplitude value corresponding to the sample value.

15. The apparatus according to claim 14, wherein the determining module is specifically configured to:

perform subband grouping on all sample values in a preset order according to the adaptive normalization length; and for each sample value, determine a subband comprising the sample value as the subband to which the sample value belongs; or

for each sample value, determine a subband consisting of m sample values before the sample value, the sample value, and n sample values after the sample value as the subband to which the sample value belongs, wherein m and n depend on the adaptive normalization length, m is an integer not less than 0, and n is an integer not less than 0.

16. The apparatus according to any one of claims 13 to 15, wherein the adjusted amplitude value calculation subunit is specifically configured to:

subtract the amplitude disturbance value corresponding to each sample value from the amplitude value of each sample value, to obtain a difference between the amplitude value of each sample value and the amplitude disturbance value corresponding to each sample value, and use the obtained difference as the adjusted amplitude value of each sample value.

17. The apparatus according to any one of claims 12 to 16, wherein the second determining unit comprises:

a division subunit, configured to divide a low frequency band signal in the speech/audio signal into N subbands, wherein N is a natural number;

a quantity determining subunit, configured to calculate a peak-to-average ratio of each subband, and determine a quantity of subbands whose peak-to-average ratios are greater than a preset peak-to-average ratio threshold; and

a length calculation subunit, configured to calculate the adaptive normalization length according to a signal type of a high frequency band signal in the speech/audio signal and the quantity of the subbands.

18. The apparatus according to claim 17, wherein the length calculation subunit is specifically configured to:

calculate the adaptive normalization length according to a formula $L = K + \alpha \times M$, wherein L is the adaptive normalization length; K is a numerical value corresponding to the signal type of the high frequency band signal in the speech/audio signal, and different signal types of high frequency band signals correspond to different numerical values K; M is the quantity of the subbands whose peak-to-average ratios are greater than the preset peak-to-average ratio threshold; and α is a constant less than 1.

19. The apparatus according to any one of claims 12 to 16, wherein the second determining unit is specifically configured to:

calculate a peak-to-average ratio of a low frequency band signal in the speech/audio signal and a peak-to-average ratio of a high frequency band signal in the speech/audio signal; and when an absolute value of a difference between the peak-to-average ratio of the low frequency band signal and the peak-to-average ratio of the high frequency band signal is less than a preset difference threshold, determine the adaptive normalization length as a preset first length value, or when an absolute value of a difference between the peak-to-average ratio of the low frequency band signal and the peak-to-average ratio of the high frequency band signal is not less than a preset difference threshold, determine the adaptive normalization length as a preset second length value, wherein the first length value is greater than the second length value; or

calculate a peak-to-average ratio of a low frequency band signal in the speech/audio signal and a peak-to-average ratio of a high frequency band signal in the speech/audio signal; and when the peak-to-average ratio of the low frequency band signal is less than the peak-to-average ratio of the high frequency band signal, determine the adaptive normalization length as a preset first length value, or when the peak-to-average ratio of the low frequency band signal is not less than the peak-to-average ratio of the high frequency band signal, determine the adaptive normalization length as a preset second length value; or

determine the adaptive normalization length according to a signal type of a high frequency band signal in the speech/audio signal, wherein different signal types of high frequency band signals correspond to different adaptive normalization lengths.

20. The apparatus according to any one of claims 12 to 19, wherein the fourth determining unit is specifically configured to:

determine a new value of each sample value according to the sign and the adjusted amplitude value of each sample value, to obtain the second speech/audio signal; or

calculate a modification factor; perform modification processing on an adjusted amplitude value, which is greater than 0, in the adjusted amplitude values of the sample values according to the modification factor; and determine a new value of each sample value according to the sign of

~~each sample value and an adjusted amplitude value that is obtained after the modification processing, to obtain the second speech/audio signal.~~

~~21. The apparatus according to claim 20, wherein the fourth determining unit is specifically configured to calculate the modification factor by using a formula $\beta = a/L$, wherein β is the modification factor, L is the adaptive normalization length, and a is a constant greater than 1.~~

~~22. The apparatus according to claim 21, wherein the fourth determining unit is specifically configured to:~~

~~perform modification processing on the adjusted amplitude value, which is greater than 0, in the adjusted amplitude values of the sample values by using the following formula:~~

~~$Y = y \times (b - \beta)$;~~

~~wherein Y is the adjusted amplitude value obtained after the modification processing; y is the adjusted amplitude value, which is greater than 0, in the adjusted amplitude values of the sample values; and b is a constant, and $0 < b < 2$.~~

1. A method for processing a speech/audio signal, wherein the method comprises:

receiving a bitstream, and decoding the bitstream, to obtain the speech/audio signal;

determining a first speech/audio signal according to the speech/audio signal, wherein –the first speech/audio signal is a signal, whose noise component needs to be reconstructed, in the speech/audio signal;

determining a symbol of each sample value in the first speech/audio signal and an amplitude value of each sample value in the first speech/audio signal;

determining an adaptive normalization length;

determining an adjusted amplitude value of each sample value according to the adaptive normalization length and the amplitude value of each sample value; and

determining a second speech/audio signal according to the symbol of each sample value and the adjusted amplitude value of each sample value, wherein the second speech/audio signal is a signal obtained after the noise component of the first speech/audio signal is reconstructed.

2. The method according to claim 1, wherein the determining an adjusted amplitude value of each sample value according to the adaptive normalization length and the amplitude value of each sample value comprises:

calculating, according to the amplitude value of each sample value and the adaptive normalization length, an average amplitude value corresponding to each sample value, and determining, according to the average amplitude value corresponding to each sample value, an amplitude disturbance value corresponding to each sample value; and

calculating the adjusted amplitude value of each sample value according to the amplitude value

of each sample value and according to the amplitude disturbance value corresponding to each sample value.

3. The method according to claim 2, wherein the calculating, according to the amplitude value of each sample value and the adaptive normalization length, an average amplitude value corresponding to each sample value comprises:

determining, for each sample value and according to the adaptive normalization length, a subband to which the sample value belongs; and

calculating an average value of amplitude values of all sample values in the subband to which the sample value belongs, and using the average value obtained by means of calculation as the average amplitude value corresponding to the sample value.

4. The method according to claim 3, wherein the determining, for each sample value and according to the adaptive normalization length, a subband to which the sample value belongs comprises:

performing subband grouping on all sample values in a preset order according to the adaptive normalization length; and for each sample value, determining a subband comprising the sample value as the subband to which the sample value belongs; or

for each sample value, determining a subband consisting of m sample values before the sample value, the sample value, and n sample values after the sample value as the subband to which the sample value belongs, wherein m and n depend on the adaptive normalization length, m is an integer not less than 0, and n is an integer not less than 0.

5. The method according to any one of claims 2 to 4, wherein the calculating the adjusted amplitude value of each sample value according to the amplitude value of each sample value and according to the amplitude disturbance value corresponding to each sample value comprises:

subtracting the amplitude disturbance value corresponding to each sample value from the amplitude value of each sample value, to obtain a difference between the amplitude value of each sample value and the amplitude disturbance value corresponding to each sample value, and using the obtained difference as the adjusted amplitude value of each sample value.

6. The method according to any one of claims 1 to 5, wherein the determining an adaptive normalization length comprises:

dividing a low frequency band signal in the speech/audio signal into N subbands, wherein N is a natural number;

calculating a peak-to-average ratio of each subband, and determining a quantity of subbands whose peak-to-average ratios are greater than a preset peak-to-average ratio threshold; and

calculating the adaptive normalization length according to a signal type of a high frequency

band signal in the speech/audio signal and the quantity of the subbands.

7. The method according to claim 6, wherein the calculating the adaptive normalization length according to a signal type of a high frequency band signal in the speech/audio signal and the quantity of the subbands comprises:

calculating the adaptive normalization length according to a formula $L = K + \alpha \times M$, wherein L is the adaptive normalization length; K is a numerical value corresponding to the signal type of the high frequency band signal in the speech/audio signal, and different signal types of high frequency band signals correspond to different numerical values K; M is the quantity of the subbands whose peak-to-average ratios are greater than the preset peak-to-average ratio threshold; and α is a constant less than 1.

8. The method according to any one of claims 1 to 5, wherein the determining an adaptive normalization length comprises:

calculating a peak-to-average ratio of a low frequency band signal in the speech/audio signal and a peak-to-average ratio of a high frequency band signal in the speech/audio signal; and when an absolute value of a difference between the peak-to-average ratio of the low frequency band signal and the peak-to-average ratio of the high frequency band signal is less than a preset difference threshold, determining the adaptive normalization length as a preset first length value, or when an absolute value of a difference between the peak-to-average ratio of the low frequency band signal and the peak-to-average ratio of the high frequency band signal is not less than a preset difference threshold, determining the adaptive normalization length as a preset second length value, wherein the first length value is greater than the second length value; or

calculating a peak-to-average ratio of a low frequency band signal in the speech/audio signal and a peak-to-average ratio of a high frequency band signal in the speech/audio signal; and when the peak-to-average ratio of the low frequency band signal is less than the peak-to-average ratio of the high frequency band signal, determining the adaptive normalization length as a preset first length value, or when the peak-to-average ratio of the low frequency band signal is not less than the peak-to-average ratio of the high frequency band signal, determining the adaptive normalization length as a preset second length value; or

determining the adaptive normalization length according to a signal type of a high frequency band signal in the speech/audio signal, wherein different signal types of high frequency band signals correspond to different adaptive normalization lengths.

9. The method according to any one of claims 1 to 8, wherein the determining a second speech/audio signal according to the symbol of each sample value and the adjusted amplitude value of each sample value comprises:

determining a new value of each sample value according to the symbol and the adjusted amplitude value of each sample value, to obtain the second speech/audio signal; or

calculating a modification factor; performing modification processing on an adjusted amplitude value, which is greater than 0, in the adjusted amplitude values of the sample values according to the modification factor; and determining a new value of each sample value according to the symbol of each sample value and an adjusted amplitude value that is obtained after the modification processing, to obtain the second speech/audio signal.

10. The method according to claim 9, wherein the calculating a modification factor comprises:

calculating the modification factor by using a formula $\beta = a/L$, wherein β is the modification factor, L is the adaptive normalization length, and a is a constant greater than 1.

11. The method according to claim 9 or 10, wherein the performing modification processing on an adjusted amplitude value, which is greater than 0, in the adjusted amplitude values of the sample values according to the modification factor comprises:

performing modification processing on the adjusted amplitude value, which is greater than 0, in the adjusted amplitude values of the sample values by using the following formula:

$$Y = y \times (b - \beta),$$

wherein Y is the adjusted amplitude value obtained after the modification processing; y is the adjusted amplitude value, which is greater than 0, in the adjusted amplitude values of the sample values; and b is a constant, and $0 < b < 2$.

12. An apparatus for reconstructing a noise component of a speech/audio signal, comprising:

a bitstream processing unit, configured to receive a bitstream and decode the bitstream, to obtain the speech/audio signal;

a signal determining unit, configured to determine a first speech/audio signal according to the speech/audio signal obtained by the bitstream processing unit, wherein the first speech/audio signal is a signal, whose noise component needs to be reconstructed, in the speech/audio signal obtained by means of decoding;

a first determining unit, configured to determine a symbol of each sample value in the first speech/audio signal determined by the signal determining unit and an amplitude value of each sample value in the first speech/audio signal determined by the signal determining unit;

a second determining unit, configured to determine an adaptive normalization length;

a third determining unit, configured to determine an adjusted amplitude value of each sample value according to the adaptive normalization length determined by the second determining unit and the amplitude value that is of each sample value and is determined by the first determining unit; and

a fourth determining unit, configured to determine a second speech/audio signal according to

the symbol that is of each sample value and is determined by the first determining unit and the adjusted amplitude value that is of each sample value and is determined by the third determining unit, wherein the second speech/audio signal is a signal obtained after the noise component of the first speech/audio signal is reconstructed.

13. The apparatus according to claim 12, wherein the third determining unit comprises:

a determining subunit, configured to calculate, according to the amplitude value of each sample value and the adaptive normalization length, an average amplitude value corresponding to each sample value, and determine, according to the average amplitude value corresponding to each sample value, an amplitude disturbance value corresponding to each sample value; and

an adjusted amplitude value calculation subunit, configured to calculate the adjusted amplitude value of each sample value according to the amplitude value of each sample value and according to the amplitude disturbance value corresponding to each sample value.

14. The apparatus according to claim 13, wherein the determining subunit comprises:

a determining module, configured to determine, for each sample value and according to the adaptive normalization length, a subband to which the sample value belongs; and

a calculation module, configured to calculate an average value of amplitude values of all sample values in the subband to which the sample value belongs, and use the average value obtained by means of calculation as the average amplitude value corresponding to the sample value.

15. The apparatus according to claim 14, wherein the determining module is specifically configured to:

perform subband grouping on all sample values in a preset order according to the adaptive normalization length; and for each sample value, determine a subband comprising the sample value as the subband to which the sample value belongs; or

for each sample value, determine a subband consisting of m sample values before the sample value, the sample value, and n sample values after the sample value as the subband to which the sample value belongs, wherein m and n depend on the adaptive normalization length, m is an integer not less than 0, and n is an integer not less than 0.

16. The apparatus according to any one of claims 13 to 15, wherein the adjusted amplitude value calculation subunit is specifically configured to:

subtract the amplitude disturbance value corresponding to each sample value from the amplitude value of each sample value, to obtain a difference between the amplitude value of each sample value and the amplitude disturbance value corresponding to each sample value, and use the obtained difference as the adjusted amplitude value of each sample value.

17. The apparatus according to any one of claims 12 to 16, wherein the second determining

unit comprises:

a division subunit, configured to divide a low frequency band signal in the speech/audio signal into N subbands, wherein N is a natural number;

a quantity determining subunit, configured to calculate a peak-to-average ratio of each subband, and determine a quantity of subbands whose peak-to-average ratios are greater than a preset peak-to-average ratio threshold; and

a length calculation subunit, configured to calculate the adaptive normalization length according to a signal type of a high frequency band signal in the speech/audio signal and the quantity of the subbands.

18. The apparatus according to claim 17, wherein the length calculation subunit is specifically configured to:

calculate the adaptive normalization length according to a formula $L = K + \alpha \times M$, wherein

L is the adaptive normalization length; K is a numerical value corresponding to the signal type of the high frequency band signal in the speech/audio signal, and different signal types of high frequency band signals correspond to different numerical values K; M is the quantity of the subbands whose peak-to-average ratios are greater than the preset peak-to-average ratio threshold; and α is a constant less than 1.

19. The apparatus according to any one of claims 12 to 16, wherein the second determining unit is specifically configured to:

calculate a peak-to-average ratio of a low frequency band signal in the speech/audio signal and a peak-to-average ratio of a high frequency band signal in the speech/audio signal; and when an absolute value of a difference between the peak-to-average ratio of the low frequency band signal and the peak-to-average ratio of the high frequency band signal is less than a preset difference threshold, determine the adaptive normalization length as a preset first length value, or when an absolute value of a difference between the peak-to-average ratio of the low frequency band signal and the peak-to-average ratio of the high frequency band signal is not less than a preset difference threshold, determine the adaptive normalization length as a preset second length value, wherein the first length value is greater than the second length value; or

calculate a peak-to-average ratio of a low frequency band signal in the speech/audio signal and a peak-to-average ratio of a high frequency band signal in the speech/audio signal; and when the peak-to-average ratio of the low frequency band signal is less than the peak-to-average ratio of the high frequency band signal, determine the adaptive normalization length as a preset first length value, or when the peak-to-average ratio of the low frequency band signal is not less than the peak-to-average ratio of the high frequency band signal, determine the adaptive normalization

length as a preset second length value; or

determine the adaptive normalization length according to a signal type of a high frequency band signal in the speech/audio signal, wherein different signal types of high frequency band signals correspond to different adaptive normalization lengths.

20. The apparatus according to any one of claims 12 to 19, wherein the fourth determining unit is specifically configured to:

determine a new value of each sample value according to the symbol and the adjusted amplitude value of each sample value, to obtain the second speech/audio signal; or

calculate a modification factor; perform modification processing on an adjusted amplitude value, which is greater than 0, in the adjusted amplitude values of the sample values according to the modification factor; and determine a new value of each sample value according to the symbol of each sample value and an adjusted amplitude value that is obtained after the modification processing, to obtain the second speech/audio signal.

21. The apparatus according to claim 20, wherein the fourth determining unit is specifically configured to calculate the modification factor by using a formula $\beta = a/L$, wherein β is the modification factor, L is the adaptive normalization length, and a is a constant greater than 1.

22. The apparatus according to claim 20 or 21, wherein the fourth determining unit is specifically configured to:

perform modification processing on the adjusted amplitude value, which is greater than 0, in the adjusted amplitude values of the sample values by using the following formula:

$$Y = y \times (b - \beta);$$

wherein Y is the adjusted amplitude value obtained after the modification processing; y is the adjusted amplitude value, which is greater than 0, in the adjusted amplitude values of the sample values; and b is a constant, and $0 < b < 2$.