CLAIMS

1. An audio encoding method, applied to an electronic device, the method comprising:

performing down-sampling processing on an audio signal to obtain a low-frequency signal of the audio signal;

performing low-frequency feature extraction processing on the low-frequency signal to obtain low-frequency features of the audio signal;

performing high-frequency analysis processing on the audio signal to obtain high-frequency features of the audio signal,

a feature dimension of the high-frequency features being lower than a feature dimension of the low-frequency features; and

performing encoding processing on the low-frequency features to obtain a low-frequency code stream of the audio signal, and performing encoding processing on the high-frequency features to obtain a high-frequency code stream of the audio signal.

2. The method according to claim 1, wherein

the audio signal comprises a plurality of first sample points obtained through sampling; and

the performing down-sampling processing on an audio signal to obtain a low-frequency signal of the audio signal comprises:

performing down-sampling processing on each of the first sampling points comprised in the audio signal through a down-sampling filter, to obtain the low-frequency signal of the audio signal.

3. The method according to claim 2, wherein the performing down-sampling processing on each of the first sampling points comprised in the audio signal through a down-sampling filter, to obtain the low-frequency signal of the audio signal comprises:

performing the following processing through the down-sampling filter:

performing digital signal-based filtering processing on the first sampling point comprised in the audio signal, to obtain a filtered audio signal; and

performing digital signal-based down-sampling processing on the filtered audio signal, to obtain the low-frequency signal of the audio signal.

4. The method according to any one of claims 1 to 3, wherein the performing low-frequency

feature extraction processing on the low-frequency signal to obtain low-frequency features of the audio signal comprises:

performing convolution processing on the low-frequency signal to obtain a convolution feature of the low-frequency signal;

performing pooling processing on the convolution feature to obtain a pooling feature of the low-frequency signal;

performing down-sampling processing on the pooling feature to obtain a down-sampling feature of the low-frequency signal; and

performing convolution processing on the down-sampling feature to obtain the low-frequency features of the audio signal.

5. The method according to claim 4, wherein

the down-sampling processing is implemented through a plurality of cascaded encoding layers; and

the performing down-sampling processing on the pooling feature to obtain a down-sampling feature of the low-frequency signal comprises:

performing down-sampling processing on the pooling feature through a first encoding layer in the plurality of cascaded encoding layers;

outputting a down-sampling result of the first encoding layer to a subsequent cascaded encoding layer, and further performing down-sampling processing and the outputting of the down-sampling result through the subsequent cascaded encoding layer until the down-sampling result is outputted to a last encoding layer; and

determining a down-sampling result outputted by the last encoding layer as the down-sampling feature of the low-frequency signal.

6. The method according to any one of claims 1 to 5, wherein the performing high-frequency analysis processing on the audio signal to obtain high-frequency features of the audio signal comprises:

performing band extension processing on the audio signal to obtain the high-frequency features of the audio signal.

7. The method according to claim 6, wherein the performing band extension processing on the audio signal to obtain the high-frequency features of the audio signal comprises:

performing frequency domain transformation processing on a plurality of second sample points comprised in the audio signal, to obtain transformation coefficients respectively corresponding to the plurality of second sample points;

dividing high-frequency transformation coefficients in the transformation coefficients respectively corresponding to the plurality of second sample points into a plurality of sub-bands;

performing averaging processing on the transformation coefficient comprised in each of the sub-bands, to obtain average energy corresponding to each sub-band, and determining the average energy as a sub-band spectral envelope corresponding to each sub-band; and

determining the sub-band spectral envelopes respectively corresponding to the plurality of sub-bands as the high-frequency features of the audio signal.

8. The method according to claim 7, wherein the performing frequency domain transformation processing on a plurality of second sample points comprised in the audio signal, to obtain transformation coefficients respectively corresponding to the plurality of second sample points comprises:

obtaining a plurality of third sample points comprised in a reference audio signal, the reference audio signal being an audio signal adjacent to the audio signal; and

performing, based on the plurality of third sample points comprised in the reference audio signal and the plurality of second sample points comprised in the audio signal, discrete cosine transformation processing on the plurality of second sample points comprised in the audio signal, to obtain transformation coefficients respectively corresponding to the plurality of second sample points.

9. The method according to claim 7, wherein the performing averaging processing on the transformation coefficient comprised in each of the sub-bands to obtain average energy corresponding to each sub-band comprises:

determining a quadratic sum of transformation coefficients corresponding to the second sample points comprised in each sub-band; and

determining, as the average energy corresponding to each sub-band, a ratio of the quadratic sum to a quantity of second sample points comprised in the sub-band.

10. An audio decoding method, applied to an electronic device, the method comprising:

performing decoding processing on a low-frequency code stream of an audio signal to obtain low-frequency features corresponding to the low-frequency code stream, and performing

decoding processing on a high-frequency code stream of the audio signal to obtain high-frequency features corresponding to the high-frequency code stream,

the low-frequency code stream being obtained by encoding a low-frequency signal obtained through down-sampling of the audio signal, and a feature dimension of the high-frequency features being lower than a feature dimension of the low-frequency features;

performing low-frequency feature reconstruction processing on the low-frequency features to obtain a low-frequency signal corresponding to the low-frequency features;

performing up-sampling processing on the low-frequency signal to obtain an up-sampling signal of the low-frequency signal; and

performing signal reconstruction processing on the high-frequency features and the up-sampling signal to obtain a synthesized audio signal corresponding to the low-frequency code stream and the high-frequency code stream.

11. The method according to claim 10, wherein the performing up-sampling processing on the low-frequency signal to obtain an up-sampling signal of the low-frequency signal comprises:

performing up-sampling processing on a fourth sampling point comprised in the low-frequency signal through an up-sampling filter, to obtain the up-sampling signal of the low-frequency signal.

12. The method according to claim 11, wherein the performing up-sampling processing on a fourth sampling point comprised in the low-frequency signal through an up-sampling filter, to obtain the up-sampling signal of the low-frequency signal comprises:

performing the following processing through the up-sampling filter:

performing digital signal-based up-sampling processing on the fourth sampling point comprised in the low-frequency signal, to obtain an up-sampled low-frequency signal; and

performing digital signal-based filtering processing on the up-sampled low-frequency signal to obtain an up-sampling signal of the low-frequency signal.

13. The method according to any one of claims 10 to 12, wherein the performing low-frequency feature reconstruction processing on the low-frequency features to obtain a low-frequency signal corresponding to the low-frequency features comprises:

performing convolution processing on the low-frequency features to obtain a convolution feature of the low-frequency features;

performing up-sampling processing on the convolution feature to obtain an up-sampling feature of the low-frequency features;

performing pooling processing on the up-sampling feature to obtain a pooling feature of the low-frequency features; and

performing convolution processing on the pooling feature to obtain the low-frequency signal corresponding to the low-frequency features.

14. The method according to claim 13, wherein

the up-sampling processing is implemented through a plurality of cascaded decoding layers; and

the performing up-sampling processing on the convolution feature to obtain an up-sampling feature of the low-frequency features comprises:

performing up-sampling processing on the convolution feature through a first decoding layer in the plurality of cascaded decoding layers;

outputting an up-sampling result of the first decoding layer to a subsequent cascaded decoding layer, and further performing up-sampling processing and the outputting of the up-sampling result through the subsequent cascaded decoding layer until the up-sampling result is outputted to a last decoding layer; and

determining an up-sampling result outputted by the last decoding layer as the up-sampling feature of the low-frequency features.

15. The method according to any one of claims 10 to 12, wherein the performing signal reconstruction processing on the high-frequency features and the up-sampling signal to obtain a synthesized audio signal corresponding to the low-frequency code stream and the high-frequency code stream comprises:

performing frequency domain transformation processing on a plurality of fifth sample points comprised in the up-sampling signal, to obtain transformation coefficients respectively corresponding to the plurality of fifth sample points;

performing band extension processing on the transformation coefficients respectively corresponding to the plurality of fifth sample points and the high-frequency features, to obtain a high-frequency transformation coefficient of a high-frequency signal; and

determining the synthesized audio signal corresponding to the low-frequency code stream and the high-frequency code stream based on the high-frequency transformation coefficient and

transformation coefficients respectively corresponding to some of the plurality of fifth sample points.

16. The method according to claim 15, wherein the performing band extension processing on the transformation coefficients respectively corresponding to the plurality of fifth sample points and the high-frequency features, to obtain a high-frequency transformation coefficient of a high-frequency signal comprises:

performing spectrum replication processing on at least part of the transformation coefficients in a first half of the transformation coefficients respectively corresponding to the plurality of fifth sample points, to obtain a reference high-frequency transformation coefficient of a reference high-frequency signal; and

performing gain processing on the reference high-frequency transformation coefficient of the reference high-frequency signal based on a sub-band spectral envelope corresponding to the high-frequency features, to obtain the high-frequency transformation coefficient of the high-frequency signal.

17. The method according to claim 16, wherein the performing gain processing on the reference high-frequency transformation coefficient of the reference high-frequency signal based on a sub-band spectral envelope corresponding to the high-frequency features, to obtain the high-frequency transformation coefficient of the high-frequency signal comprises:

dividing the reference high-frequency transformation coefficient of the reference high-frequency signal into a plurality of sub-bands based on the sub-band spectral envelope corresponding to the high-frequency features; and

performing the following processing for each of the plurality of sub-bands:

determining first average energy of a high-frequency sub-band corresponding to the sub-band in the sub-band spectral envelope, and determining second average energy of the sub-band;

determining a gain factor based on a ratio of the first average energy to the second average energy;

multiplying the gain factor by each reference high-frequency transformation coefficient comprised in the sub-band, to obtain a high-frequency transformation coefficient corresponding to the sub-band; and

determining the high-frequency transformation coefficient corresponding to each of the

plurality of sub-bands as the high-frequency transformation coefficients of the high-frequency signal.

18. The method according to claim 16, wherein the performing spectrum replication processing on at least part of the transformation coefficients in a first half of the transformation coefficients respectively corresponding to the plurality of fifth sample points, to obtain a reference high-frequency transformation coefficient of a reference high-frequency signal comprises:

performing spectrum replication processing on a second half of the transformation coefficients in a first half of the transformation coefficients respectively corresponding to the plurality of fifth sample points, to obtain a reference high-frequency transformation coefficient of a reference high-frequency signal.

19. The method according to claim 15, wherein the determining the synthesized audio signal corresponding to the low-frequency code stream and the high-frequency code stream based on the high-frequency transformation coefficient and transformation coefficients respectively corresponding to some of the plurality of fifth sample points comprises:

performing combining processing on the high-frequency transformation coefficient and transformation coefficients respectively corresponding to a first half of fifth sample points in the plurality of fifth sample points, to obtain a complete transformation coefficient; and

performing inverse transformation processing of spectrum transformation on the complete transformation coefficient, to obtain the synthesized audio signal corresponding to the low-frequency code stream and the high-frequency code stream.

20. An audio encoding apparatus, comprising:

a down-sampling module, configured to perform down-sampling processing on an audio signal to obtain a low-frequency signal of the audio signal;

a low-frequency extraction module, configured to perform low-frequency feature extraction processing on the low-frequency signal to obtain low-frequency features of the audio signal;

a high-frequency analysis module, configured to perform high-frequency analysis processing on the audio signal to obtain high-frequency features of the audio signal,

a feature dimension of the high-frequency features being lower than a feature dimension of the low-frequency features; and

an encoding module, configured to perform encoding processing on the low-frequency

features to obtain a low-frequency code stream of the audio signal, and perform encoding processing on the high-frequency features to obtain a high-frequency code stream of the audio signal.

21. An audio decoding apparatus, comprising:

a decoding module, configured to perform decoding processing on a low-frequency code stream of an audio signal to obtain low-frequency features corresponding to the low-frequency code stream, and perform decoding processing on a high-frequency code stream of the audio signal to obtain high-frequency features corresponding to the high-frequency code stream,

the low-frequency code stream being obtained by encoding a low-frequency signal obtained through down-sampling of the audio signal, and a feature dimension of the high-frequency features being lower than a feature dimension of the low-frequency features;

a low-frequency reconstruction module, configured to perform low-frequency feature reconstruction processing on the low-frequency features to obtain a low-frequency signal corresponding to the low-frequency features;

an up-sampling module, configured to perform up-sampling processing on the low-frequency signal to obtain an up-sampling signal of the low-frequency signal; and

a signal reconstruction module, configured to perform signal reconstruction processing on the high-frequency features and the up-sampling signal to obtain a synthesized audio signal corresponding to the low-frequency code stream and the high-frequency code stream.

22. An electronic device, comprising:

a memory, configured to store a computer program or a computer-executable instruction; and

a processor, configured to implement the audio encoding method according to any one of claims 1 to 9 or the audio decoding method according to any one of claims 10 to 19 when executing the computer program or the computer-executable instruction stored in the memory.

23. A computer-readable storage medium, having a computer program or a computer-executable instruction stored therein, the computer program or the computer-executable instruction, when executed by a processor, implementing the audio encoding method according to any one of claims 1 to 9 or the audio decoding method according to any one of claims 10 to 19.

24. A computer program product, comprising a computer program or a computer-executable

instruction, the computer program or the computer-executable instruction, when executed by a processor, implementing the audio encoding method according to any one of claims 1 to 9 or the audio decoding method according to any one of claims 10 to 19.

25. A non-transitory computer-readable storage medium storing an audio bitstream that is generated by an audio encoding method, the audio encoding method comprising:

performing down-sampling processing on an audio signal to obtain a low-frequency signal of the audio signal;

performing low-frequency feature extraction processing on the low-frequency signal to obtain low-frequency features of the audio signal;

performing high-frequency analysis processing on the audio signal to obtain high-frequency features of the audio signal,

a feature dimension of the high-frequency features being lower than a feature dimension of the low-frequency features; and

performing encoding processing on the low-frequency features to obtain a low-frequency code stream of the audio signal, and performing encoding processing on the high-frequency features to obtain a high-frequency code stream of the audio signal.