is the maximum value, but every sub-band is allocated some bits. Accordingly, it is considered that degradation in the quality of audio signals can be suppressed and the effective use of radio waves and frequency bands can be realized.

Citation List

Patent Literature

[0013]

PTL 1: Japanese Unexamined Patent Application Publication (Translation of PCT Application) No. 2013-534328

PTL 2: International Publication No. 2005/027095

Non Patent Literature

[0014]

NPL 1: ITU-T Standard G.719, 2008

Summary

[0015]

However, the effective use of radio waves and frequency bands needs to be further promoted. Here, in a case of coding an audio signal having a sampling frequency of about 32 kHz at a low bit rate of 20 kbps/s or less by using the above-described method adopted by Standard G.719, it is not possible to reserve a unit (a number of bits) used in quantization of all sub-bands, which is a problem.

[0016]

Fig. 9 is a diagram illustrating an example of the result of bit allocation to each sub-band in a case of using the coding method specified in Standard G.719 at 20 kbit/s. As illustrated, bit allocation fails not only in a high-frequency range but also, depending on the situation, in a low-frequency range, which is essential for hearing. Consequently, coding of spectra in the corresponding sub-bands is not possible, resulting in significant degradation in the quality of audio signals.

[0017]

A method for dynamically changing a bit allocation method may be employed in various embodiments (PTL 1).

[0018]

However, the bit allocation method is changed while a single coding method (quantization method) is used without changing the coding method (quantization method), and therefore, this approach to degradation in the quality of audio signals has a limited effect.

[0019]

The present disclosure provides a coding technique and a decoding technique for realizing high-quality audio signals while reducing the overall bit rate.

[0020]

An audio signal coding apparatus according to an aspect the present disclosure includes a time-frequency transformer that generates a spectrum by performing a transform on an input audio signal into a frequency domain, divides the spectrum into sub-bands, which are predetermined frequency bands, and outputs sub-band spectra; a sub-band energy quantizer that obtains, for each of the sub-bands, quantized sub-band energy; a tonality calculator that analyzes tonality of the sub-band spectra and outputs an analysis result; a bit allocator that selects a second sub-band on which quantization is performed by a second quantizer from among the sub-bands on the basis of the analysis result of the tonality and the quantized sub-band energy, and determines a first number of bits to be allocated to a first sub-band, among the sub-bands, on which quantization is performed by a first quantizer; and a multiplexer that multiplexes into information coded information output from the first quantizer and from the second quantizer, the quantized sub-band energy, and the analysis result of the tonality, and outputs the multiplexed information. The first quantizer pulse-codes a sub-band spectrum among the sub-band spectra that is included in the first sub-band by using the first number of bits, and the second quantizer codes a sub-band spectrum among the sub-band spectra that is included in the second sub-band by using a pitch filter.

[0021]

Note that general or specific embodiments may be implemented as a system, a method, an integrated circuit, or a computer program, or any selective combination of a system, an apparatus, a method, an integrated circuit, and a computer program.

In an aspect of the present disclosure, an audio signal coding method comprising: generating a spectrum by performing a transform on an input audio signal into a frequency domain; dividing the spectrum into sub-bands, which are predetermined frequency bands, and outputting sub-band spectra; obtaining, for each of the sub-bands, quantized sub-band energy; analyzing tonality of the sub-band spectra and outputting an analysis result; selecting a second sub-band from among the sub-bands on the basis of the analysis result of the tonality and the quantized sub-

band energy; determining a first number of bits to be allocated to a first sub-band among the sub-bands; generating first coded information by coding a sub-band spectrum among the sub-band spectra that is included in the first sub-band by using the first number of bits; generating second coded information by coding a sub-band spectrum among the sub-band spectra that is included in the second sub-band by using a pitch filter; and multiplexing together and outputting the first coded information and the second coded information.

[0022]

With the coding apparatus, decoding apparatus, and so on according to the present disclosure, it is possible to code and decode high-quality audio signals while reducing the overall bit rate.

Brief Description of Drawings

[0023]

[Fig. 1] Fig. 1 is a block diagram of a coding apparatus according to a first