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Method for transmitting a signal

The invention relates to a method for transmitting a signal according to the preamble of Claim 1.

When an audio signal is being transmitted, for example in the case of radio transmission, transmission via cable, satellite transmission and in the case of recording devices, it is known to convert the analog audio signal into a digital audio signal with a specific resolution, to transmit it in this form and to convert it back into an analog signal during reproduction. The digital transmission permits a larger signal-to-noise ratio to be obtained, in particular during reproduction.

The bandwidth which is required for the transmission of such a signal is essentially determined by the number of sampled values to be transmitted per time unit and by the resolution.

In practice, there is a need to keep the bandwidth necessary for the transmission as small as possible in order to be able to manage with a narrowband channel or to be able to transmit simultaneously as many audio signals as possible over an existing channel. The required bandwidth can in fact be decreased by reducing the sampled values or the number of bits per sampled value.

However, this measure usually results in a

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degradation during reproduction. In a known method (German Offenlegungsschrift 35 06 912.0), in order to improve the reproduction quality the digital audio signal is transformed in chronologically successive periods into a short-time spectrum which constitutes the spectral components of the signal for each of the time periods, for example 20 ms. Owing to psycho-acoustic laws, components which are not perceived by the listener, which are therefore irrelevant in terms of telecommunications, can be located better in the short-term spectrum than in the time domain. These components are given less weighting, or omitted entirely during the transmission. By virtue of this measure, a considerable part of the data which would otherwise be necessary can be omitted during the transmission so that the average bit rate can be substantially reduced.

The method described in J. P. Princen and A. B. Bradley, "Analysis/Synthesis Filter-bank Design based on Time Domain Aliasing Cancellation", IEEE-Transactions Acoustics, Speech, Signal-Processing, Volume ASSP-34, pages 1153 - 1161, October 1986, is suitable for dividing up the signal into sections. In this application, a transformation is described in which overlapped blocks with rounded-off window functions are produced in the windows without additional coefficients in the frequency range. In this method, N values are initially extracted from the input signal using a window function f(n) of the length N and are subsequently transformed to form N/2 significant coefficients in the frequency range. The inverse transformation is calculated from the N/2 coefficient of N sampled values which are in turn weighted with the window function f(n).

However, the output signal of the inverse transformation differs from the input signal of the original transformation. The precise reconstruction of the input signal is only possible by adding the output values of successive inverse transformations in the overlapping area of in each case N/2 sampled values. So that the

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input signal can be recovered by means of this so-called "overlap add", the window function f(n) must fulfil the following conditions:

$$f(N-1-n) = (f(n) \qquad 0 \leq n \leq N-1 \qquad (1)$$

$$f^2(N/2-1-n)+f^2(n) = 2 \qquad 0 \leq n \leq N/2-1 \qquad (2)$$

The first condition corresponds to a symmetry of f(n). The second condition corresponds to the point symmetry of the square of f(n) in one window half in each case. Taking into account these conditions, the effective window length of the transformation can be varied between N/2 and N sampled values.

The selection of the window length using these methods in transformation encoders leads to the following effects. A large window length with the most rounded-off shape possible permits good frequency selectivity. At the same time, the error due to quantization of the coefficients after the inverse transformation extends over the entire effective window length. This can have negative effects on the subjective quality of the encoded signal, in particular in the case of large jumps in the amplitude of the signal to be encoded.

The selection of the shorter windows brings about a degradation of the frequency selectivity, which, in particular in the case of highly correlated input signals, has negative effects on the transformation gain to be achieved. In contrast, in the case of large signal jumps, errors due to quantization of the coefficients can be restricted to the respective window so that they are prevented from affecting adjacent windows.

The invention is based on the object of improving a method for transmitting a signal according to the preamble of Claim 1 to the extent that an optimum frequency selectivity and high subjective quality of the encoded and decoded signal are achieved.

The object is achieved according to the invention with a method of the type mentioned at the beginning by

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means of the features described in the characterizing part of Claim 1.

The invention is based on the idea that a window switch-over is appropriate even if the useful signal has a frequency jump at the same time as constant energy. Owing to the psycho-acoustic conditions, this may even be necessary under certain circumstances. For this purpose, the useful signal frequency range (for example 0 ... 20 kHz) is divided up into at least 2 separate frequency ranges which are appropriate in terms of psycho-acoustics. The input signal is evaluated separately in the respective frequency ranges and a logical combination of the separate results is subsequently carried out.

In one development of the invention, the frequency change is also used for switching over filter bands for a sub-band encoder (adaptive windowing), as a result of which the transformation gain is ultimately increased.

Furthermore, by means of the detection of frequency jumps, adaptive quantization and encoding can be controlled in the same way as audio encoding methods such as NICAM, MUSICAM, MSC, etc., which permit a variable block length. As a result of the detection of frequency jumps, the time and the rate of occurrence of scale factors, bit assignment factors, etc. can also be calculated so that an improved controllability and observability of the transformation and of the aforementioned audio encoding methods is possible.

It is advantageous if, in addition to the detection of frequency changes, a detection of amplitude changes is carried out, and the adaptive windowing, quantization and encoding is carried out by logically combining the two results.

The invention is explained in greater detail below in a drawing with reference to one of many possible exemplary embodiments. In the drawing:

Figs 1a-c show window functions with different widths,

Fig. 2 shows asymmetrical window functions and

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Fig. 3a shows the profile of an input signal with amplitude jump,

Fig. 3b shows a window function which is adapted to the input signal in Fig. 3a,

Fig. 3c shows the profile of an input signal with frequency jump,

Fig. 3d shows a window function which is adapted to the input signal in accordance with Fig. 3c,

Fig. 4a shows the block circuit diagram of an encoder part of a transmitter,

Fig. 4b shows the block circuit diagram of a decoder part of a receiver.

Fig. 1 shows a plurality of windows of constant width (b) and with different window functions f(n). The windows are each composed in such a way that the adjacent windows overlap to their mid point. As a result, in Fig. 1a overlapping to the mid point also occurs in the case of the sinusoidal window function f(n), in the illustration according to Fig. 1b partial overlapping, to less than the midpoint, occurs, and in Fig. 1c the two limbs of the window function f(n) only touch.

Fig. 2 shows the superimposition of two windows with asymmetrical window functions f(n) and g(n). The window functions are however configured in their common overlapping area such that their resultants complement one another to form 1.

An amplitude profile A(t) of an input signal is illustrated in Fig. 3a. As can be seen, the signal initially has a uniform low amplitude which is followed by a signal jump, after which the original amplitude continues. Below this, in Fig. 3b window functions of the windows, with which this signal can be processed to an optimum degree, are illustrated. In the first area 1, these window functions are sinusoidal window functions which permit a high degree of frequency selectivity. The window 3, in which the signal with the high amplitude occurs, has, in contrast, a window function which is very

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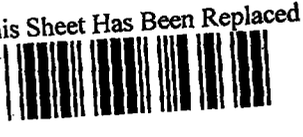

narrow. The window functions of the adjacent windows 2 and 4, that is to say of the preceding window and of the following window have a correspondingly adapted window function in the overlapping area so that in these areas the window functions complement one another to form one. Thus, the window functions in these two adjacent windows 2 and 4 have an asymmetrical shape. Quantization faults which would occurs inside the window 3 and in which the signal jump occurs are thus restricted to the area of the window functions of the window 3 and are thus reduced to half the chronological duration for example in comparison with a window function with a sinusoidal profile. As a result of the covering effect of the signal jump, a subjective improvement of the audio quality is thus obtained despite the reduced frequency selectivity.

The frequency profile F(t) of an input signal is illustrated in Fig. 3c. As can be seen, the signal initially has a constant frequency (for example 10 kHz) which is followed by a frequency jump to a substantially lower frequency (for example 200 Hz) takes place [sic]. The amplitude of the signal remains constant during this process. Window functions of the windows with which this signal can be processed to an optimum degree are illustrated in Fig. 3d.

In the first area 1, the window functions are sinusoidal window functions which permit a high degree of frequency selectivity. The window functions of windows 2 and 3 in which the frequency jump occurs have, in the overlapping area, a window function which is correspondingly adapted in such a way that in this area the window functions complement one another to form 1. Here, the overlapping area is made very narrow. Thus, the window functions in the two adjacent windows 2 and 3 have an asymmetrical shape. Quantization faults which would occur within the window 3 and in which the signal jump occurs are restricted to the area of the window functions of the window 3 and are thus reduced to half the chronological duration for example in comparison with a window function

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with a sinusoidal profile. As a result of the covering effect of the signal jump, a subjective improvement in the audio quality is thus obtained despite the reduced frequency selectivity.

For the simplest detection of frequency jumps, the useful signal with the useful signal frequency range (0 ... 20 kHz) is divided here into two separate frequency ranges (0 ... 1 kHz and 1 kHz ... 20 kHz). Each frequency range is now evaluated separately and subsequently the separate results are logically combined with one another.

In terms of psycho-acoustics, this leads to the following window switch-over. When there is a decrease in the signal energy in the upper frequency range and a simultaneous increase in the lower frequency range, a window switch-over must be performed since it is necessary for psycho-acoustic reasons. The case in the region of the frequency jump according to Fig. 3c then occurs. When there is a decrease in energy in the upper range and no increase in the lower range, no window switch-over is necessary since this case is not critical. Likewise, a window switch-over may be omitted if there is an increase in energy only in the lower range. If an increase in energy takes place only in the upper range, a switch-over must take place in the usual fashion. If there is an increase in energy in both ranges, a switch-over to small window lengths must take place because the signal must contain pulses.

Analysis filters, such as are already used in sub-bandcoders, are suitable for detecting frequency changes. However, simpler filters (for example minimum-phase filters or filter banks without a reconstruction facility) are also conceivable.

The block circuit diagram of an encoder part in a transmitter is shown in Fig. 4a. The OBT (Overlapping-Block Transform), which belongs to the class of semi-overlapping transformations, is realized as the transformation in the encoder realized here. When there is an

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advance of one block of N sampled values in the time domain and a transformation of two N sampled values, in turn only N sampled values are also obtained in the image range so that no additional coefficients have to be transmitted. The OBT also fulfils the requirement that the transformation coefficients correspond to the spectrum of the input signal. As a result of the use of non-rectangular windows which is possible with the OBT, the block effects are also reduced and the frequency selectivity improved. As a result of the adaptive windowing, i.e. switching over the transformation length and the shape of the window as well as window length, which is possible with the OBT, the upper limit for N is not determined here by the covering time. Thus, the adaptive windowing brings about an additional improvement as the result of the fact that the length of the transformation length [sic] permits a good frequency resolution and a higher transformation gain to be achieved and, with corresponding window adaptation, permits the pre-echoes to be suppressed by switching over to shorter transformation lengths. A detection of signal jumps (combination of the detection of amplitude jumps and frequency jumps) is provided for the window adaptation in order to be able to detect changes in terms of pulse, amplitude, phase and frequency in a block in advance. The detection of signal jumps produces a window signal - referred to below as window identification - with which the windowing, transformation and adaptive quantization and encoding in the encoder part of the transmitter is controlled. The advance analysis of the input signal prevents a data block being transformed and encoded repeatedly with different transformation lengths and ensures that the decision as to which of the encoded values is used for transmission is not made, by means of comparison, until after the latter has occurred.

Together with the adaptive bit assignment and adaptive quantization which exploit the overlapping properties of the human ear in the frequency range, the

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encoder which is realized makes it possible to take into account psycho-acoustic conditions to an extremely large degree.

By means of the window identification, it is possible to carry out a corresponding adaptive decoding and inverse OBT in the decoder (see Fig. 4b), so that only the signals which are relevant in terms of psycho-acoustics are output as output signal. Although the signal X(n)' objectively contains less information than the signal X(n), the difference cannot be perceived by the human ear, so that a signal which is adapted completely correctly to the psycho-acoustic conditions can be perceived.

The invention described above is not at all restricted to the OBT but rather also makes it possible to control all known audio-encoding methods which permit a variable block length, such as NICAM, MUSICAM, MSC. The time and the rate of occurrence of scale factors, assignment etc., can be calculated by means of the detection of frequency jumps.

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Patent claims

1. Method for transmitting a signal which is divided up by windows into successive overlapping blocks, [lacuna] in that the signal elements contained in the blocks are each converted into a spectrum by means of transformation, the spectra are subsequently encoded, transmitted, decoded after the transmission and converted back into signal elements by inverse transformation, and in that, finally, the blocks which contain the signal elements are joined to one another in an overlapping fashion, the overlapping areas of the blocks being weighted in such a way that the resultant of the window functions is one, characterized in that the window functions are selected as a function of frequency changes of the signal.

2. Method according to Claim 1, characterized in that the length of the window functions is selected as a function of the frequency changes.

3. Method according to one of the two preceding claims, characterized in that the windows are at constant intervals and the effective length of the window functions in comparison with the length of the windows is selected as a function of the frequency changes.

4. Method according to Claim 1, characterized in that the intervals between the windows are selected as a function of the frequency changes.

5. Method according to one or more of Claim 1 to 4, characterized in that the length of the window functions is selected to be small in the case of large frequency changes and large in the case of small frequency changes.

6. Method according to one of the preceding claims, characterized in that, in order to detect the frequency changes, the useful signal frequency range is divided up into at least 2 frequency ranges which are appropriate in terms of psycho-acoustics, in that each frequency range is evaluated individually and subsequently a logical combination of the individual results is carried out.

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7. Method according to one of Claims 1 to 6, characterized in that the window functions are selected as a function of the amplitude changes of the signal.

8. Method according to Claim 7, characterized in that the length of the window functions is selected as a function of amplitude changes.

9. Method according to one of the two Claims 7 or 8, characterized in that the window function of a following window in the area of overlap with the preceding window is formed [sic] the mirror image of the window function, present in this area, of the preceding window.

10. Method according to one of Claims 7 to 9, characterized in that the length of the window functions is selected to be small in the case of large amplitude changes and large in the case of small amplitude changes.

11. Method for carrying out the method according to one of the preceding claims, characterized in that, by means of a detection of frequency changes and amplitude changes, a window-identification signal is produced which controls an adaptive windowing, transformation, quantization and encoding in the encoder part of a transmitter.

12. Device according to Claim 11, characterized in that the decoding, the transformation and the windowing can be controlled in the decoder part of the receiver by means of the window identification in the encoder part of the transmitter.

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