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

When transmitting an audio signal, e.g. during radio transmission, cable transmission, satellite transmission and when using recording equipment, it is known to convert the analogue audio signal into a digital audio signal with a certain resolution, to transmit it in this form and convert it back into an analogue signal at the reproduction stage. Digital transmission results in a greater signal-to-noise ratio, particularly at the reproduction stage.

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

The bandwidth required for transmission must in practice be kept as small as possible so as to enable a narrow-band channel to be used or to be able to simultaneously transmit as many audio signals as possible via an existing channel. The required bandwidth can be compressed as such by reducing the number of sampled values or the number of bits per sampled value.

However this measure usually results in a deterioration in the reproduction quality. In order to improve reproduction quality, according to a known method (DE OS 35 06 912.0) the digital audio signal is transformed in sections which follow one another in time into a short-term spectrum which in each case represents the spectral components of the signal for the periods of time, e.g. 20 ms. In conformity with psychoacoustic relationships, components which are not perceived by the listener and are therefore irrelevant in terms of communications are generally easier to locate in the short-term spectrum than in the time domain. These components are either weighted to a lesser degree or omitted entirely when transmission takes place. This measure enables a substantial proportion of the normally necessary data to

be omitted during transmission, so that the average bit rate can be substantially reduced. Developments of this method, which is known from the DE-A-3 506 912, are described in the EP-A-0 251 028.

The method described by J. P. Princen and A. B. Bradley in "Analysis/Synthesis Filterbank 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 the signal into sections. This publication contains a description of a transformation in which overlapping blocks with rounded window functions are produced in the windows without additional coefficients in the frequency domain. In this method N values are firstly cut out of the input signal by means of a window function f(n) of length L and then transformed into N/2 significant coefficients in the frequency domain. The inverse transformation calculates from the N/2 coefficient N sampled values, which are in turn weighted by the window function f(n).

However the output signal of the inverse transformation differs from the input signal of the initial transformation. The input signal can only be exactly reconstructed by adding the output values of successive inverse transformations in the overlap area of N/2 sampled values in each case. In order to enable the input signal to be recovered by 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 these conditions into account, the effective window length of the transformation between N/2 and N sampled values can be

varied.

The choice of window length when applying these methods in transformation encoders has the following effects. A long window which is as rounded as possible permits good frequency selectivity. In this case the errors due to quantization of the coefficients following inverse transformation extend over the entire effective window length. This may have a negative effect on the subjective quality of the encoded signal, particularly when there are substantial jumps in the amplitude of the signal which is to be encoded.

The result of choosing shorter windows is a deterioration of the frequency selectivity. This has a negative effect on the transformation gain to be achieved, particularly in the case of highly correlated input signals. However errors due to quantization of the coefficients can be restricted to the window in question when there are substantial signal jumps, so that they do not have any effect on adjacent windows.

The object of the invention is to improve a method for transmitting a signal according to the preamble of claim 1 so as to obtain optimum frequency selectivity and a high subjective quality of the encoded and decoded signal.

This object is solved in a method of the type initially mentioned by the features presented in the characterising part of claim 1. Developments are listed in the dependent claims.

In one development the window functions of a subsequent window in the area of overlap with the preceding window are formed by reflecting the window functions of the preceding window which are present in this area. This results in an even better adaptation to rapid signal variations. In the case of the window function described above it was also assumed that like window functions are used for all the transformations. If, however, it is assumed that different

window functions, e.g. f(n) and g(n), are used in successive blocks, it becomes apparent that the relations (1) and (2) only relate to the area in which the two halves of the windows overlap:

$$f(N-1-n) = g(n) \qquad 0 \leq n \leq N/2-1 \qquad (3)$$

$$f^2(N/2+n)+g^2(n) = 2 \qquad 0 \leq n \leq N/2-1 \qquad (4)$$

In this case both equations relate just to the second half of the function f and the first half of the function g. The window halves of one block can therefore be selected independently of one another if, together with just the corresponding window halves of the adjacent blocks, they satisfy the equations (3) and (4).

The method according to the invention is explained in the light of the drawings, in which:

Figures 1a - 1c show window functions with different widths,
Figure 2 shows asymmetrical window functions,
Figure 3a shows the variation of an input signal and
Figure 3b shows a window function which is adapted to this input signal.

Figure 1 shows a plurality of windows of a constant width (b) with different window functions f(n). The windows are assembled such that half of adjacent windows overlap. There is thus also an overlap by one half in the case of the sinusoidal window function f(n) in Figure 1a, while in the representation according to Figure 1b there is a partial overlap, which represents less than one half, and in Figure 1c there is just contact between the two sides of the window function f(n).

Figure 2 shows the superimposition of two windows with asymmetrical window functions f(n) and g(n). However the window functions are formed in their common overlapping area such that their resultants add to give 1.

Figure 3a shows an amplitude characteristic A(t) of an input signal. As can be seen, the signal firstly has a constant, low amplitude, which is followed by a signal jump, after which the original amplitude continues. Below this Figure 3b shows window functions of the windows with which this signal can be optimally processed. In the first area 1 these are sinusoidal window functions which permit a high frequency selectivity. However the window 3, in which the signal having the high amplitude falls, has a very narrow window function. The window functions of the adjacent windows 2 and 4, i.e. of the preceding and subsequent windows, have an appropriately adapted window function in the overlapping area, so that the window functions add to give one in these areas. The window functions therefore have an asymmetrical shape in these two adjacent windows 2 and 4. Quantization noise which would occur inside the window 3 in which the signal jump falls is thus restricted to the area of the window functions of the window 3 and thus reduced by half in terms of duration with respect to, for example, a window function with a sinusoidal variation. In spite of the reduced frequency selectivity, the covering effect of the signal jump thus results in a subjective improvement of the audio quality.

The invention is not restricted to audio signals. It may equally relate to video signals. In this case the successive blocks may also follow at, for example, intervals of space instead of time.

Claims

1. Method for transmitting a signal which is divided by window functions (f(n), g(n)) into successive, overlapping windows, that the partial signals contained in the windows are then in each case converted by transformation into a spectrum, the spectra are then encoded, transmitted, decoded following transmission and converted back into partial signals by inverse transformation, and that the windows containing the partial signals are finally assembled in an overlapping manner, the overlapping areas of the windows being weighted such that the resultant of the window functions (f(n), g(n)) yields one, characterised in that the length of the window functions (f(n), g(n)) is selected in accordance with signal variations.

2. Method according to claim 1, characterised in that the windows have constant spacings and the effective length of the window functions (f(n), g(n)) compared with the length of the windows is selected in accordance with signal variations.

3. Method according to claim 1, characterised in that the spacings of the windows are selected according to signal variations.

4. Method according to one or more of claims 1 to 3, characterised in that the length of the window functions (f(n), g(n)) is selected so as to be small when the signal variations are large and large when the signal variations are small.

5. Method according to one or more of claims 1 to 4, characterised in that the window function of a subsequent window in the area of overlap with the preceding window is formed by reflecting the window function of the preceding window which is present in this area.

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6. Method according to one or more of claims 1 to 5, characterised in that a set of different window functions (f(n), g(n)) is defined from which a window function is in each case assigned in accordance with signal variations, and that during transmission a code which clearly determines the assigned window function is transmitted.

7. Method according to one or more of claims 1 to 6, characterised in that the signals are subject to variations with time, and in particular that the signals are audio signals.

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