

The invention relates to a method for the transmission of a signal in accordance with the first part of claim 1.

For the transmission of an audio signal e.g. for radio broadcast transmission, cable transmission, satellite transmission and in recording devices, it is known to convert the analogue signal into a digital signal having a certain resolution, to transmit it in this form and to convert it again into an analogue signal for the reproduction. Due to the digital transmission, a greater signal to noise ratio is achieved particularly during the reproduction.

The bandwidth required for the transmission of such a signal is determined, in essence, by the number of sampling values that have to be transmitted per unit of time as well as by the resolution.

In practice, there exists the demand to keep the bandwidth necessary for the transmission as small as possible in order to manage with a narrow band channel or to be able to simultaneously transmit as many audio signals as possible over a wideband channel. The bandwidth required can be reduced by a reduction of the sampling values or of the number of the bits per sampling value. In general however, this measure results in a worsening of the quality of reproduction.

For the improvement of the quality of reproduction, a method is known from the DE-OS 35 06 912 in which the digital audio signal is broken down into successive intervals of time and transformed into an impulsive spectrum which represents the spectral components of the signal for the respective time segments. On the grounds of a psycho-acoustic like law, components, that cannot be perceived by the listener i.e. which are irrelevant in a broadcasting sense, can generally be better located in the impulsive spectrum than in the time domain. These components are less weighted or completely removed for the transmission. Hereby, a considerable portion of the otherwise necessary data can be dropped during the transmission so that the average bit rate can be substantially reduced.

For the formation of the time segments, the signal is initially weighted in the time domain by an analysis window and, after the transformation, coding, transmission, decoding and inverse transformation, it is then weighted by a synthesis window. The form of the analysis window influences the frequency resolution as well as the amount of data occurring during the transmission. Thus, for windows having "hard" flanks, such as a rectangular window has for example, the frequency resolution is poor.

Namely, the spectral components caused by the extreme rise and fall of the signal at the beginning and end of the window are additionally added to the spectrum of the original signal in the weighted segment. Nevertheless, the time segments could be brought back together again without overlapping.

In the method described in the DE-OS 35 06 912, a window function having "soft" flanks has already been selected. Here, a cosine squared function succeeds the beginning and end of the analysis window and a sine squared function the corresponding regions of the synthesis window. The central region of both windows has a constant value. An improved frequency resolution is already obtained with a form of the window function of this type. However, in the region of the "soft" flanks, an overlapping of the successive time segments is necessary which leads to an increase of the average bit rate due to the double transmission of the signals contained in this region.

A further improvement of the frequency resolution could be achieved by means of a still lower steepness of the flank of the window function of the analysis window as well as by an extension of the flank region within the window. Necessarily with these measures however, a greater overlapping of adjacent time segments is required.

If the flank region is extended so much that the window functions no longer has a constant value in any region, then adjacent time segments must overlap by 50%. The number of sampling values and in correspondence therewith, the amount of data, is thereby doubled.

For a 50% overlapping of successive time segments, it is known from the publication "Analysis/Synthesis Filter Bank Design Based on Time Domain Aliasing Cancellation", J. P. Princen and A.B. Bradley, IEEE Transactions, ASSP-34, No 5, Oct 1986, pp 1153 - 1161: and "Subband/Transform Coding Using Filter Bank Design Based on Time Domain Aliasing Cancellation", J. P. Princen, A.W. Johnson and A.B. Bradley, IEEE Int. Conference on Acoustics, Speech and Signal Processing, 1987, pp 2161 - 2164, to reduce the amount of data back to the original value in that only every second sampling value is coded. This proposal assumes similar window functions for the analysis and synthesis windows. The aliasing components resulting from the under-sampling can be compensated after the weighting by the synthesis window when using similar window functions.

It can be useful for improving the frequency resolution to use a specially shaped window for the analysis in order to realise a

low starting slope of the window function for example. The advantage of such a window function lies in that a very high frequency resolution is achieved for narrow band signal components which then leads to a very effective bit allocation having a low bit rate during the coding.

It is known from the lecture paper "Spectral Properties of Audio Signals and Masking with Aspect to Bit Data Reduction", B. Feiten, 86th AES-Convention, March 1989, to use different window functions for the analysis and the synthesis and to apply these to time segments which overlap by 50%. The graphical definition of the sine function which is described does not however lead to a compensation of the aliasing components after the weighting by the synthesis window.

The object of the invention is therefore, to improve the method for the transmission of a signal as is expressed in the first part of the claim 1 to the effect that, with a free choice of an analysis window, a synthesis window can be determined such that by the combination of the two, the aliasing components are compensated to zero.

This object is achieved in a method such as is described in the first part of the claim 1 by the features specified in the characterising part.

The invention thus makes it possible to derive the synthesis window function in a mathematically exact manner from the freely selected analysis window function. The analysis window function can thereby be optimally matched to the requirements for a high frequency resolution without this advantage being partly or completely offset by the aliasing components during the reproduction of the signal. The specified relationship also takes into account unsymmetrical window functions. For symmetrical window functions, one embodiment provides a simplification of the calculation of the synthesis windows.

The invention will be hereinafter explained and contrasted with the state of the art with the help of an embodiment which is illustrated in the drawing.

In the drawing:

Fig. 1 shows a flow diagram including the essential method steps of the invention,

Fig. 2 a diagram of analysis windows and synthesis windows in accordance with a known method,

Fig. 3 a diagram of the aliasing components for the window functions in accordance with Fig. 2,

Fig. 4 a diagram of analysis windows and synthesis windows in accordance with the method according to the invention,

Fig. 5 a diagram of the aliasing components for the window functions in accordance with Fig. 4,

Fig. 6 a table representing the relationship between real and imaginary parts on the one hand and pseudo magnitudes and phases on the other hand for an alternative of the subsampling and

Fig. 7 a table similar to Fig. 6 for another alternative of the subsampling.

In the flow diagram illustrated in Fig. 1, the individual method steps for the carrying out of the method of the invention are illustrated.

An analogue audio signal which, in accordance with method step 1, is converted into a digital signal in which amplitude values digitally coded as sampling values are present, forms the starting value of the method.

In the method step 2, the continuous signal is windowed in that a series of successive sampling values, 1024 sampling values in the present case, are selected.

In the method step 3, blocks are formed from the selected sampling values which overlap in time by 50%. This means, that to some extent, the same sampling values are present in adjacent blocks, although at different locations. Thus, the sampling values present in the first half of a current block correspond to those in the second half of the preceding block.

In the method step 4, the signal segments contained in the blocks are weighted by analysis windows. Hereby, a gentle signal run-in and run-out at the edges of the block is produced which increases the precision of the analysis in the subsequent transformation. A suitable analysis window is illustrated in Fig. 4, which will be discussed further below.

The method step 5 forms the transformation of the signal which up to now was time discrete, into a frequency discrete signal. Spectral values now appear instead of the amplitude values. If a

Fourier transformation is used for the transformation, the transformation values each comprise a real and an imaginary part.

Thereafter, in the method step 6, there follows a conversion of the spectral values into a representation having pseudo values and phases. The spectral values are then ready and prepared for a transmission process such as is described in the DE-OS 35 06 912. For this, several alternatives are possible of which two are illustrated in the tables in accordance with Figs. 6 and 7. In connection with the conversion of the spectral values, a sub-sampling is also simultaneously carried out. As a result, the number of values to be transmitted then agrees once more with the number of the original sampling values. The doubling of the items of data caused by the 50% overlapping of the blocks is thus reversed here.

In the method step referenced 7, a plurality of individual steps are encompassed which comprise the coding, if necessary, a data reduction, transmission and decoding. These method steps may be carried out in accordance with the method in the DE-OS 35 06 912.

In the method step 8, there now follows a transformation which is the inverse of that of the method step 5 but to which, for a preceding data reduction, an altered signal that has been freed from psycho-acoustically redundant components is subjected. Time discrete signals in the form of blocks representing signal segments of a continuous signal are again the result of the inverse transformation. However, only the half of the original sampling values are present in the blocks.

In the following method step 9, a weighting of the blocks by synthesis windows is undertaken. The synthesis window functions are so formed that they balance out once more the signal distortions which have arisen due to the weighting by the analysis windows in the method step 4. The synthesis window functions used here fulfil two criteria. Firstly, in the overlapping region, they form the complement of unity with the corresponding analysis windows. Secondly, the difference between the analysis window mirrored in the middle of the overlapping region multiplied by the synthesis window for the block n and the analysis window mirrored in the middle of the overlapping region multiplied by the synthesis window for the block $n+1$ is equal to zero in the overlapping region. This last criterion incorporates the compensation of the aliasing components.

In the method step 10, the 50% overlapping blocks are added whereby the aliasing components, in the two blocks that are to be overlapping, appear respectively with inverted signs so that, with the addition, there is a compensation to zero.

The method step 11 represents the formation of continuous sampling values by making the blocks having the windowed signal segments contiguous.

Finally, in the last method step referenced 12, a conversion of the digitally coded sampling values into an analogue signal is undertaken, from which objectively of course, components are missing but which subjectively is found to be identical with the original signal.

Fig. 2 shows the diagram of window functions, namely, an analysis window and a synthesis window such as is described in the reference "Spectral Properties of Audio Signals and Masking with Aspect to Bit Data Reduction", B. Feiten, 86th AES-Convention, March 1989. However, if the window functions specified in this diagram are used in the method steps 4 and 9 in accordance with the illustration in Fig. 1, then the aliasing components illustrated in Fig. 3 arise. This is not desirable since the improved representational accuracy achieved by the increased sharpness of the analysis is, in practice, once more partly nullified.

Fig. 4 shows window functions, namely an analysis and a synthesis window in which the synthesis window function was calculated from the freely chosen analysis window function according to the invention in accordance with the equations:

$$s_n(t) = \frac{a_{n-1}(T_B/2 - t)}{a_n(T_B/2 - t) * a_{n+1}(t) + a_n(t) * a_{n+1}(T_B/2 - t)} ; 0 \leq t \leq T_B/2$$

$$s_{n+1}(t) = \frac{a_n(T_B/2 - t)}{a_n(T_B/2 - t) * a_{n+1}(t) + a_n(t) * a_{n+1}(T_B/2 - t)} ; 0 \leq t \leq T_B/2$$

In these equations:

$a_n(t)$ represents the analysis window function for the block n ,

$s_n(t)$ the synthesis window function for the block n ,
 $a_{n+1}(t)$ the analysis window function for the block $n+1$,
 $s_{n+1}(t)$ the synthesis window function for the block $n+1$ and
 T_B the block time

The appertaining diagram of the aliasing components in Fig. 5 shows that the aliasing components are compensated to zero here, i.e. the improved representational accuracy of the reproduction of the signal is achieved in full.

While the equations given above generally also take into account unsymmetrical windows, the simplified equation

$$s(t) = \frac{a(T_B/2 - t)}{a(T_B/2 - t) * a(t) + a(t) * a(T_B/2 - t)} ; 0 \leq t \leq T_B/2$$

can also be used for symmetrical windows as is illustrated in Fig. 4. In this equation

$a(t)$ means the analysis window function,
 $s(t)$ the synthesis window function and
 T_B the block time

The subsampling undertaken in the method step 6 may be carried out such that the real parts of the spectral values having an even frequency index and the imaginary parts of the spectral values having an odd frequency index in the m th block are used for the transmission. The real parts of the spectral values having an odd frequency index and the imaginary parts of the spectral values having an even frequency index are used for the transmission in the $(m+1)$ th block. Consequently, one only needs $N/4$ real parts and $N/4$ imaginary parts for the transmission for a block having N values with approximately 50% overlapping.

If the spectra are processed with a codec, which requires a magnitude and phase representation, then one requires magnitude and phase values of the spectral values. To this end, the real and imaginary parts have to be converted into corresponding magnitudes and phases.

The tables illustrated in Figs. 6 and 7 show the formation of pseudo magnitudes and phases from real and imaginary parts for the transformation of the signal into a frequency discrete signal as well as the selection of the values for the subsampling.

With the alternative, for which the table in Fig. 6 is illustrative, a pseudo magnitude and phase representation is formed in that one combines in each case the real parts of the spectral values having the frequency indices n with the adjacent imaginary parts of the spectral values having frequency indices $n+1$ as regards the phase and magnitude. One thereby obtains $N/4$ magnitudes and $N/4$ phases. This representation is depicted in Fig. 6.

In another alternative, as is shown by Fig. 7, two successive blocks are combined in each case. The pseudo magnitude and the pseudo phase can be calculated from the real parts of the m th and the imaginary parts of the $(m+1)$ th blocks having the same frequency index. One thus obtains $N/2$ magnitudes and $N/2$ phases for two blocks.

The formation of the pseudo magnitude and the pseudo phase will be explained for a further alternative where a modified cosine and sine transformation is effected as the transformation.

Initially the 50% overlapping blocks are subjected alternately in a block-by-block manner to a modified cosine transformation in accordance with the equation

$$y(k) = D_k \sum_{n=0}^{N-1} x(n) * \cos \left[\frac{(2n+1+N/2)k \pi}{2N} \right] ; 0 \leq k \leq N-1$$

wherein

y represents the spectral values,
 k the indices of the spectral values,
 D a constant in accordance with the definition

$$D_0 \text{ (for } k=0) = \sqrt{N} ; D_k = \sqrt{2/N} ; 1 \leq k \leq N-1$$

x the time values,
 n the indices for the time values of the sampling and
 N the number of the sampling values within a block

and are subjected to a modified sine transformation in accordance with the equation

$$y(k) = D_k \sum_{n=0}^{N-1} x(n) * \sin \left[\frac{(2n+1+N/2)k \pi}{2N} \right], 1 \leq k \leq N$$

wherein

y represents the spectral values,
k the indices of the spectral values,
D a constant in accordance with the definition

$$D_k = \sqrt{2/N} ; 1 \leq k \leq N-1 ; D_N = \sqrt{N}$$

x the time values,
n the indices for the time values of the sampling and
N the number of the sampling values within a block

and, during the inverse transformation, are subjected to a transformation which is the inverse of the initial transformation.

Both transformations deliver real spectral values. In correspondence therewith, they are not directly suitable for a coding in accordance with magnitude and phase as in the case of the Fourier transformation.

The even numbered ones of the spectral values occurring are selected whereby the number of the spectral values which will be further used is only half as great as that of the original values.

Thereafter, complex values are formed from the spectral values of, in each case, two successive, overlapping blocks in that for example, the values of the first block are multiplied by 1 and those of the second block by j and similarly indexed values of the two blocks are added. There thus result complex values which can be resolved in accordance with magnitude and phase. Since the real and imaginary parts of the complex values do not respectively originate from one and the same block, they are referred to as pseudo magnitude and pseudo phase values.

As regards magnitude and phase, the thus resulting pseudo spectrum has in general a very great similarity to the Fourier spectra of the two overlapping blocks which are being considered insofar as the time duration of the individual blocks does not substantially exceed approximately 20 ms. On the basis of the similarity, methods, which were optimised for the coding of real Fourier spectral values in accordance with magnitude and phase e.g. the method described in the DE-OS 35 06 912, are suitable for the coding of the pseudo spectral values.

CLAIMS

1. A method of transmitting a signal in which it is divided by windows into successive blocks which overlap by at least 50% and the signal portions contained in the blocks are weighted by means of analysis windows, then the part signals contained in the blocks undergo a transformation process allowing subsampling with concentration of the aliasing components, such as Time Domain Aliasing Cancellation, the spectra resulting from the transformation are next coded, transmitted, decoded after the transmission and converted by inverse transformation back into part signals, the blocks containing the part signals are weighted by means of synthesis windows and reassembled overlapping each other, characterised in that the window functions of the synthesis windows are determined in dependence on the window functions of the corresponding analysis windows in the overlapping range according to the equations

$$s_n(t) = \frac{a_{n-1}(T_B/2 - t)}{a_n(T_B/2 - t) * a_{n+1}(t) + a_n(t) * a_{n+1}(T_B/2 - t)} ; 0 \leq t \leq T_B/2$$

$$s_{n+1}(t) = \frac{a_n(T_B/2 - t)}{a_n(T_B/2 - t) * a_{n+1}(t) + a_n(t) * a_{n+1}(T_B/2 - t)} ; 0 \leq t \leq T_B/2$$



in which

$a_n(t)$ represents the analysis window function for the block n ,

$s_n(t)$ represents the synthesis window function for the block n ,

$a_{n+1}(t)$ represents the analysis window for the block $n + 1$

$s_{n+1}(t)$ represents the synthesis window of the block $n + 1$, and

T_B represents the block period.

2. A method according to claim 1, characterised in that with symmetrical windows the window function of the synthesis window is determined in dependence on the window function of the analysis window according to the equation

$$s(t) = \frac{a(T_B/2 - t)}{a(T_B/2 - t) * a(t) + a(t) * a(T_B/2 - t)} ; 0 \leq t \leq T_B/2$$

in which

$a(t)$ represents the analysis window function

$s(t)$ represents the synthesis window function and

T_B represents the block period.

3. A method according to claim 1 or 2, characterised in that the subsampling is undertaken in such a way that after the transformation there are selected for the coding and transmission in the m th block the real parts of the spectral values with even frequency index and the imaginary parts of the spectral values with odd frequency index and that there are selected in the $(m + 1)$ th block the imaginary parts of the spectral values with even frequency index and the real parts of the spectral values with odd frequency index.



4. A method according to claim 1 or 2, characterised in that the subsampling is undertaken in such a way that after the transformation there is formed a pseudo magnitude and phase representation, in which in each case the real parts of the spectral values with the frequency indices n are combined, with respect to magnitude and phase, with adjoining imaginary parts of the spectral values with the frequency indices $n + 1$.

5. A method according to claim 1 or 2, characterised in that the subsampling is undertaken in such a way that after the transformation two successive blocks are combined and there is formed a pseudo magnitude and phase representation, in which in each case the real parts of the spectral values of the m th block are selected with the imaginary parts of the spectral values of the $(m + 1)$ th block with the same frequency index, and magnitude and phase are calculated therefrom.

6. A method according to claim 1 or 2, characterised in that as a preferred transformation process for the initial transformation there is undertaken, alternating from block to block, a modified cosine transformation according to the equation

$$y(k) = D_k \sum_{n=0}^{N-1} x(n) * \cos \left[\frac{(2n+1+N/2)k\pi}{2N} \right] ; 0 \leq k \leq N-1$$

in which

y represents the spectral values

k represent the indices of the spectral values,



D represents a constant according to the definition

$$D_0 \text{ (for } k=0) = \sqrt{N} ; D_k = \sqrt{2/N} ; 1 \leq k \leq N-1$$

x represents the time values,

n represents the indices of the time values of the sampling and

N represents the number of sampling values in a block

and a modified sine transformation is undertaken according to the equation

$$y(k) = D_k \sum_{n=0}^{N-1} x(n) * \sin \left[\frac{(2n+1+N/2)k \pi}{2N} \right] ; 1 \leq k \leq N$$

in which

y represents the spectral values,

k represents the indices of the spectral values

D represents a constant according to the definition

$$D_k = \sqrt{2/N} ; 1 \leq k \leq N-1 ; D_N = \sqrt{N}$$

x represents the time values,

n represents the indices of the time values of the



sampling and

N represents the number of sampling values in a block

and for the inverse transformation there is undertaken a transformation which is inverse to the initial transformation.

7. A method according to claim 6 characterised in that preferably from the spectral values occurring there are selected the even ones and thereafter from the spectral values which are further employed there are formed complex values, in which from the spectral values of two successive and mutually-overlapping blocks in each case the values of the first block are multiplied by 1 and those of the second block are multiplied by j and values of both the blocks having the same indices are added.

