

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 slots. 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.

However, due to the formation of time slots, the frequency resolution is impaired since the spectral components caused by the rise and fall of the signal at the beginning and end of the window are additionally added to the spectrum of the original signal. An improvement of the frequency resolution can be achieved by means of a low steepness of the flank of the window function as well as by an extension of the flank region within

the window. Necessarily with these measures, an overlapping of adjacent time slots is required.

If the flank region is extended so much that the window function no longer has a constant value in any region, then adjacent time slots 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 adjacent time slots, 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. The aliasing components resulting from the under-sampling are cancelled out by the use of the method described in this literature after the inverse transformation during the bringing back together again of the time slots.

It has been demonstrated that, for amplitude oscillations within a time segment, especially for signals just starting from rest in the course of a block, these signals are superimposed with noticeable interferences after the transmission. The cause for their being noticeable lies in that the interferences even occur before the start of the signals and thus are not sufficiently hidden.

These interferences may be the result of e.g. quantisation noise which superimposes itself on the impulsive spectrum. After the inverse transformation, the noise components then appear within the whole block in the time domain.

For the reduction of these interferences, the signals in the block in which the change in level occurs, could be subjected to a compression and then after the inverse transformation to an expansion. However, if the boosting of the level for the carrying out of the compression does not thereby extend over the whole block, then the signal components are coupled with aliasing components in another region of the block which cannot be eliminated by the expansion.

The object of the invention is, in a method of the type mentioned hereinabove, to combine an improvement of the

precision of the analysis of the signals which are to be transformed with an improvement of the signal to noise ratio for very large changes of the signal.

This object is achieved in a method as stated in the first part of the claim 1, by the features given in the characterising part.

Thus, the method in accordance with the invention uses overlapping blocks having window functions that result in a high precision of the analysis for signals which are subjected to only small changes in level. If now, a jump in level is detected which exceeds a predetermined threshold, the window functions are changed.

These changed window functions have no or only a small overlap. The impulsive spectra occurring after the transformation are then no longer under-sampled but rather are fully sampled. Thus, no aliasing components can arise here. In so doing, it is accepted that the doubly occurring spectral values within the region of overlap increase the amount of data to be coded. The signals weighted by the changed window functions may be subjected to a compression with a subsequent complementary expansion. The interferences occurring before the jump in level after the inverse transformation, also known as pre-echoes, are thereby reduced. After the jump in level, the signal is then again processed in overlapping blocks.

Developments and advantageous embodiments are apparent from the claims, the description and the drawing which explains an embodiment.

In the drawing:

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

Fig. 2 a graph of the useful signal and aliasing components when using the method in accordance with the literature "Princen & Bradley",

Fig. 3 a diagram of the time sequence of window functions during a jump in the signal,

Fig. 4 a diagram similar to Fig. 3 but with other window functions,

Fig. 5 window functions suitable for the use of the "Time Domain Aliasing Cancellation" method,

Fig. 6a-d the graph of a signal starting during the course of a block, both originally as well as after the inverse transformation,

Fig. 7 the division into sub blocks,

Fig. 8 a window function which is superimposed on the blocks,

Fig. 9 the energy waveform within a block, with and without compression,

Fig. 10 a graph of the non-boosted and of the boosted signals for a signal jump with non-overlapping windows for the calculation of the energy of the sub blocks,

Fig. 11 a graph similar to Fig. 10 but having 50% overlapping windows for the calculation of the energy of the sub blocks,

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, a decision is made as regards a jump in level exceeding a predetermined threshold. If there is no jump in level, then one proceeds to method step 4.

In the method step 4, 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 5, the signal segments contained in the blocks are weighted with 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.

The method step 6 forms the transformation of the signal which up to now was time discrete, into a frequency discrete signal. Spectral values, which each comprise a real and an imaginary part, appear now instead of the amplitude values.

Thereafter, in the method step 7, 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. 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 8, 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 9, there now follows a transformation which is the inverse of that of the method step 6 but which, for a preceding data reduction, is subjected to an altered signal that has been freed from psycho-acoustically redundant components. 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 10, a weighting of the blocks by synthesis windows is undertaken. The synthesis window functions are so arranged that they balance out once more the signal distortions which have arisen due to the weighting by the analysis windows in the method step 5. 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. This last criterium incorporates the compensation of the aliasing components.

In the method step 11, the 50% overlapping blocks are added whereby the aliasing components in the two superimposed overlapping blocks appear respectively with inverted signs so that, with the addition, there is a compensation to zero.

The method step 12 represents the formation of continuous sampling values by making the blocks having the windowed signal segments contiguous.

Finally, in the last method step referenced 13, 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.

On the other hand, if in accordance with the decision in method step 3, a level jump is present, then one continues with the method step 14.

In the method step 14, blocks are formed from the selected sampling values which do not overlap or overlap by substantially less than 50%.

In the method step 15, the signal segments in the blocks are weighted by analysis windows which have a steep edged waveform.

In the method step 16, a compression takes place.

The method step 17, which corresponds to the method step 6, forms the transformation of the up till now time discrete signal into a frequency discrete signal. Spectral values, which each comprise a real and an imaginary part, appear now instead of the amplitude values.

Thereafter in the method step 18, 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. In contrast to the method step 7 however, a sub-sampling is not effected.

In the method step referenced 19, which corresponds to the method step 8, 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 20, analogously to the method step 9, there now follows a transformation which is the inverse of that of the method step 17 or 6 but which, for a preceding data reduction, is subjected to an altered signal that has been freed from psycho-acoustically redundant components. 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 21, an expansion is carried out.

In the method step 22, a weighting of the blocks by synthesis windows is undertaken. The synthesis windows functions are so arranged that they balance out once more the signal distortions which have arisen due to the weighting by the analysis windows in the method step 15.

In the method step 23, the blocks are added in so far as they overlap.

Thereafter, the whole of the already described method steps 12 and 13 are carried out.

Fig. 2 illustrates the useful signal and the aliasing components for a transformation block as it would appear by usage of the "Time Domain Aliasing Cancellation" method in accordance with Princen & Bradley. The aliasing components result from the mirroring of the useful signal from one block half about the line of symmetry $t_b/4$ or $3t_b/4$. By means of an overlapping addition with the 50% overlapping adjacent blocks, the aliasing components immediately cancel each other out as they have correspondingly inverted signs here. Thus, after transformation and inverse transformation, the correct useful signal is obtained once more.

If a block was specially subjected to a compression and then processed in accordance with the "Time Domain Aliasing Cancellation" method, then the compression process can no longer be reversed by an expansion. For example, if the signals in the first quarter of the block are boosted then these boosted signals appear in the second quarter of the block as aliasing components and are coupled additively to the useful signal. These aliasing components which have been altered by the

compressor, can no longer be compensated by the aliasing components of the adjacent blocks.

Thus, before carrying out a compression upon detection of a signal jump, one switches over to another window which does not produce any greatly overlapping blocks and the "Time Domain Aliasing Cancellation" method is no longer used on these blocks. The time sequence of such blocks is illustrated in the following Figures 3 and 4.

Fig. 3 shows the time sequence of blocks with a diagram of the window functions used for the weighting upon detection of a jump in the signal wherein non-overlapping blocks occur during the jump in the signal.

A block is illustrated in line 1 which overlaps the previous and the following block by 50%. Aliasing components appear in both halves of the block.

A start block for a 0% overlapping with the following block is illustrated in line 2. Aliasing components only appear in the first half of the block since the last quarter of the block is identical to zero.

Line 3 shows a block of equal block length weighted by a rectangular window function but without any overlapping. A compression and expansion of the signal extending over a part of the block length can be carried out in this block.

Line 4 shows an end block for a 0% overlapping with the previous block. Aliasing components only appear in the second half of the block since the first quarter of the block is identical to zero.

A block corresponding to the block illustrated in line 1 is again shown in line 5. This block has a 50% overlapping with the previous and succeeding block.

Fig. 4 shows the time sequence of blocks with a diagram of the window functions used for the weighting upon detection of a jump in the signal wherein blocks overlapping by 6.25% occur during the jump in the signal.

A block is illustrated in line 1 which overlaps the previous and the following block by 50%. Aliasing components appear in both halves of the block.

A start block for a 6.25% overlapping with the following block is illustrated in line 2. Aliasing components only appear in the

first half of the block since the last quarter of the block is identical to zero.

Line 3 shows a block of equal block length weighted by a rectangular window function but without any overlapping. A compression and expansion of the signal extending over a part of the block length can be carried out in this block.

Line 4 shows an end block for a 6.25% overlapping with the previous block. Aliasing components only appear in the second half of the block since the first quarter of the block is identical to zero.

A block corresponding to the block illustrated in line 1 is again shown in line 5. This block has a 50% overlapping with the previous and succeeding block.

The windows used here in the region of a jump in level have an average constant waveform and flanks corresponding to a cosine function. Due to the overlapping present in the region of the flanks, there occurs, after the transformation, an approximately 12.5% greater number of spectral components than with the non-overlapping rectangular blocks in accordance with Fig. 1.

In order to take this into consideration during the coding, a block slipping bit allocation, coarse quantisation or a suppression of less relevant spectral values can be effected.

Fig. 5 shows window functions suitable for the use of the "Time Domain Aliasing Cancellation" method, 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

With the use of these equations, the signals weighted by the analysis and the synthesis window complement each other to unity and aliasing components are compensated.

The case taken into account by the method in accordance with the invention, that a signal suddenly starts from rest and this occurrence takes place within a block e.g. in its second half, is dealt with in the following Figures 6a-d. This case is illustrated for the time domain in Fig. 6a. Fig. 6b shows the transformed signal. Due to quantisation errors during the coding, an interference spectrum is superimposed on the spectrum illustrated in Fig. 6b so that the spectrum illustrated in Fig. 6c arises. After an inverse transformation, this interference spectrum does not influence the signal waveform just at the onset of the signal but already at the beginning of the block as Fig. 6d shows. Since the pre-hiding effect is less than the post-hiding effect, it is possible for this interference to be heard. By means of a corresponding compression within the block before the transformation and transmission and by an expansion after the transmission and the inverse transformation, the signal to noise ratio can be considerably improved.

To this end, as is illustrated in Fig. 7, each block 116, 117, ... is divided into sub blocks. Except at the edges of the blocks, these sub blocks 119, 120, 121 ... have equal time dilations such that they respectively overlap one another by half. At the edges of the block, there is an overlapping with sub block 118 of the half time dilation. The average signal powers (energy in the time segment divided by the dilation of the time segment) are determined in these overlapping rectangular sub blocks.

In a subsequent step, as is shown in Fig. 8, the initially rectangular sub blocks 119, 120, 121 ... are weighted by $\cos^2$ window functions 122. The time slots at the edges of the block which only have half the time dilation of the other blocks, are weighted by a $\cos^2$ half windows 123. The overlapping weighting functions complement each other at each time point of the signal block to 1.

It is shown in Fig. 9, how the signals in the sub blocks 119, 120, 121, corresponding to the detected average power, illustrated by the solid lines, are so amplified or attenuated that the average powers in the sub blocks 119, 120, 121 become approximately equal, illustrated by the dashed lines. For the purposes of clearer illustration, the blocks are not shown dashed here. Due to the amplification or attenuation of the signals, the ratio between the useful block energy and the interference energy of the block resulting from the coding is not altered. On the other hand, due to these measures, it is now achieved that the same signal to noise ratio exists in all of the blocks. Thus, with reference to the signal to noise ratio, the same effect is achieved as though one had selected from the very beginning, by windowing, blocks which would correspond to the size of the sub blocks. The previously mentioned disadvantages of short blocks are, however, avoided.

Expediently, the overlapping blocks are dimensioned for a time dilation of ca 2 - 4 ms on psycho-acoustical grounds. For blocks having ca 1000 sampling values at a sampling frequency of 44.1 kHz, this corresponds to the formation of ca 10 - 20 sub blocks. Moreover, it is expedient on psycho-acoustical grounds, to limit the signal amplification to a maximum value of e.g. 40 dB.

It is sufficient, to quantise the amplification factors, whereby the quantisation can be effected relatively coarsely, in order to limit the additional data which is necessary for the quantisation steps. The quantisation may be so effected that smaller quantisation steps are selected for smaller amplification factors than for larger. Thereby, the quantisation is so dimensioned that the average power in the boosted sub block does not exceed the average power in the sub block having the highest detected average power i.e. the reference block. In this manner, it is even possible to achieve a gain in the ratio of the useful block energy relative to the interference energy of the block. In this case, the signal to noise ratio for all the sub blocks is no longer identical but is only approximately identical.

If only the sub blocks in which the compression of the signal takes place are weighted by overlapping window functions, but not the sub blocks which serve for the determination of the average signal power for the calculation of the amplification factors, this may result in excessively high amplification factors for certain signal jumps.

This case is illustrated in Fig. 10 for an ideal rectangular jump. The non-boosted signal waveform is referenced by 126, the

boosted signal waveform by 127. The small letters a0 to a8 stand for the boosting factors, otherwise referred to as amplification factors. The excessive increase appears if the flanks of the jump and the limits of the sub block do not coincide.

In order to keep the increase low, the determination of the average signal power is, in accordance with a further development, likewise effected in blocks having a 50% overlapping, here however, using rectangular windows. They correspond directly to the sub blocks in which the signals are amplified. The result of this measure is illustrated in Fig. 11 for the same signal jump. Again, the non-boosted signal waveform is referenced by 126, but this time, the boosted altered signal waveform is referenced by 128.

If one applies the method described up to now to the complete audio signal, then the amplification factors are only correct for the energy rich spectral components since, in essence, these determine the factors. In audio signals, the spectral components up to ca 3 kHz are nearly always the most energy rich.

If the method for the energy rich spectral components up to ca 3 kHz has the greatest representational accuracy, then signal jumps at higher frequencies with lower energy components may lead to greater inaccuracies during the coding which could possibly lead to interferences that can be heard.

The signal may also be subjected to a pre-emphasis before the transmission and coding and to a de-emphasis after the transmission and decoding.

Claims

1. Method for the transmission of an audio signal in which the analogue signal is converted into a digital signal, transmitted and again transformed into an analogue signal, wherein the digital audio signal is divided into successive blocks by windows and converted into a spectrum by a respective transformation, that thereafter the spectra are coded, transmitted, decoded after the transmission and converted again into partial signals by an inverse transformation, that the blocks containing the partial signals are finally fitted together again, wherein the signal is weighted on the basis of the changes in level, characterised in that,

a) for changes in level below a predetermined value, the signal is divided up into blocks that overlap one another by 50% and the signal segments contained in the blocks are weighted by means of analysis windows, that the transformation is carried out in accordance with a method which enables a subscanning with compensation of the aliasing components, in particular the "Time Domain Aliasing Cancellation" method, that after the inverse transformation the blocks containing the partial signals are weighted by means of synthesis windows and are fitted together again in an overlapping manner and that

b) for changes in level above a predetermined value, the signal is divided up into blocks that overlap one another by less than 50% or do not overlap and the signal segments contained in the blocks are fully scanned and that the signals are submitted to a compression before the transformation and to an expansion after the inverse transformation.

2. Method in accordance with Claim 1, characterised in that, for the recognition of the changes in level, the energies, contained in the signal of a time period extending over at least two blocks, are determined and the energy values from an earlier region of the time period are compared with those from a later region.



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3. Method in accordance with Claim 1 or 2, 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 region in accordance with the equations:

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

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

wherein,

$a_n(t)$ 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

and that the blocks not overlapping by 50% are divided up into sub blocks and the signal in the sub blocks is amplified and/or attenuated in dependence on the average signal powers therein during the compression and is treated in a complementary manner during the expansion.

4. Method in accordance with Claim 3, characterised in that, the maximum alteration of the time dependent amplification and/or attenuation is limited.

5. Method in accordance with Claim 3 or 4, characterised in that, the division of the blocks into sub blocks is effected by weighting with window functions which have a constant value over the time period.

6. Method in accordance with Claim 5, characterised in that, the window functions overlap one another and the blocks are weighted at their two ends with window functions of half the temporal width.

7. Method in accordance with Claim 6, characterised in that, the blocks are weighted with further "soft" window functions that



have a continuous waveform, are zero at the limits of the sub blocks and, in the whole block, are complementary to produce a constant weighting value, preferably of 1.

8. Method in accordance with any of the Claims 5 to 7, characterised in that, the window functions, except at the edges of the blocks, are Hanning windows corresponding to $\cos^2$ windows and at the edges of the blocks are $\cos^2$ half windows.

9. Method in accordance with any of the Claims 5 to 8, characterised in that, the signals within the respective sub blocks are amplified or attenuated by a constant factor.

10. Method in accordance with Claim 9, characterised in that, the amplification factors (attenuation factors) are derived from the average powers in the signal segments corresponding to the sub blocks, as detection segments.

11. Method in accordance with any of the Claims 5 to 8, characterised in that, the detection segments are also obtained, like the sub blocks, by weighting the blocks with window functions which have a constant value over the time period.

12. Method in accordance with Claim 11, characterised in that, for preference, the window functions overlap.

13. Method in accordance with any of the Claims 10 to 12, characterised in that, specifications (energy divided by duration of the period) are formed for the average powers in the detection segments.

14. Method in accordance with Claim 13, characterised in that, the specifications for the detection segments of a block are related to the specification of a detection segment, preferably that one having the greatest specification and that the resulting specification ratios determine the amplification factors, preferably proportionally.

15. Method in accordance with Claim 14, characterised in that, quantised amplification factors are selected.



16. Use of the method in accordance with any preceding claim in broadcast transmission, cable transmission, satellite transmission or in recording equipment.



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