

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. The advantage of a high frequency resolution consists in that with narrow band signal components only a few items of data are needed for their coding whereby a very effective bit allocation

is achieved and the average quantity of data occurring during the transmission is considerably reduced.

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 follows 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 this measure 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 subsampling can be

compensated after the weighting by the synthesis window when using similar window functions.

It has been discovered that the frequency resolution can be increased by the selection of larger overlapping regions if the signal is simultaneously weighted by suitable analysis and synthesis windows. In order to reduce the increasing data rate caused by the greater number of sampling values back to its original value, the subsampling must be effected with a still higher factor whereby however, further aliasing components appear.

The object of the invention is, for a method for the transmission of a signal, to provide generally valid measures which will allow a reduction of the sampling values and thus of the data rate in the case of multiply overlapping blocks with a simultaneous compensation of the aliasing components.

This object is achieved by the method specified in the claim 1.

The method makes it possible to provide the general conditions for the transformations and to use these in a practical form for overlappings which correspond to a power of two i.e. for a twofold, fourfold, eightfold etc overlapping. Embodiments of the method also provide the general conditions for determining different analysis and synthesis window and for their application in a practical way.

By the utilisation of the method steps in accordance with the invention, one is able to exploit the analysis properties of softer analysis window functions which become better with increasing overlapping. By means of appropriate subsampling in the frequency domain, the outlay is reduced to the original data rate of the time signal through multiple transmission of partial window regions.

Further developments and advantageous embodiments of the invention are apparent from the claims, the further description and the drawing. The invention will be explained hereinafter.

In the drawing:

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

Fig. 2 a graphical diagram of the segmenting of the continuous time signal,

Fig. 3 the formation of blocks from the segmented original signal,

Fig. 4 an even and odd block signal portion,

Fig. 5 the periodisation of an even block signal portion,

Fig. 6 a subsampling in the spectrum by the factor 2 and displacement from the origin of the co-ordinate system by $TB/2$ in the time domain when using a cosine transformation,

Fig. 7 a subsampling in the spectrum by the factor 2 and displacement from the origin of the co-ordinate system by $TB/4$ in the time domain when using a cosine transformation,

Fig. 8 signal components with overlapping addition for different displacements of the starting point of the block from the origin of the co-ordinate system,

Fig. 9 a subsampling in the spectrum by the factor 2, displacement from the origin of the co-ordinate system by $TB/4$ in the time domain and displacement by half a sampling period while sampling in the frequency domain when using a cosine transformation,

Fig. 10 signal components after transformation, subsampling, transmission and inverse transformation for different transformations,

Fig. 11 a signal reconstruction for 50% overlapping,

Fig. 12 a segmenting of analysis and synthesis windows,

Fig. 13 components of the analysis and synthesis windows in the respective block region,

Fig. 14 signal components after transformation, subsampling, transmission and inverse transformation in the synthesis window region for different transformations with fourfold overlapping of the blocks,

Fig. 15 an aliasing compensation using rectangular windows for the analysis and synthesis,

Fig. 16 a compensation scheme for the alternating use of sine and cosine transformations,

Fig. 17 a twofold overlapping in the time domain,

Fig. 18 a fourfold overlapping in the time domain,

Fig. 19 an eightfold overlapping in the time domain,

Fig. 20 a twofold overlapping in the frequency domain,

Fig. 21 a fourfold overlapping in the frequency domain,

Fig. 22 an eightfold overlapping in the frequency domain.

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

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 the sampling values 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 during the subsequent transformation.

The method step 5 forms the transformation of the signal which up to now was time discrete, into a frequency discrete signal. Spectral values which in each case comprise a real and an imaginary part now appear instead of amplitude values.

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. A sub-sampling is also carried out simultaneously in connection with the conversion of the spectral values. As a result, the number of the 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 correspondence 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 centre of the overlapping region multiplied by the synthesis window for the block n and the analysis window mirrored in the centre 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.

For the further explanation, the cosine or the sine transformation will serve as the basis of the multiply overlapping transformations. Furthermore, we will assume the use of time continuous signals throughout the description of the method. The transition to time discrete signals can be separated from any further considerations and be undertaken in accordance with the generally known manner of procedure.

The cosine transformation is defined as an integral transformation by means of the following pair of equations:

$$F_c(f) = A_c \int_0^{\infty} f(t) \cos(2 \pi f t) dt \quad (1)$$

$$f(t) = B_c \int_0^{\infty} F(f) \cos(2 \pi f t) df \quad (2)$$

The same applies for the sine transformation:

$$F_s(f) = A_s \int_0^{\infty} f(t) \sin(2 \pi f t) dt \quad (3)$$

$$f(t) = B_s \int_0^{\infty} F(f) \sin(2 \pi f t) df \quad (4)$$

The constants A_c , B_c , A_s and B_s serve for the purposes of

normalisation and do not warrant further consideration here. In accordance with the equations (1) to (4), the time function $f(t)$ for the cosine or sine transformation is different from zero only for $t > 0$. This can always be achieved for a transformation involving finite block lengths by appropriate choice of the origin of the co-ordinate system.

The cosine and the sine transformation can be traced back to the Fourier transformation in that the even signal portion of a Fourier transformation is supplied for the cosine transformation and the odd signal portion for the sine transformation. By using this conversion, all of the known theorems used for the Fourier transformation can also be made use of for the sine and cosine transformation.

The continuous time signal is subdivided into segments which correspond to the block regions that are supplied after the transformation, subsampling, transmission and inverse transformation for the overlapping addition. If one designates the block length as T_B , then the segments with 50% overlapping have the length $T_B/2$ as Fig. 2 depicts. The segments in time sequence are referenced N_i .

If blocks having 50% overlapping are cut out from the thus described continuous signal by means of a simple rectangular window, then the blocks which are apparent from Fig. 3 occur for a cycle comprising a transformation, subsampling, overlapping and inverse transformation.

If one selects the origin of the co-ordinate system such that it coincides with the start of the transformation block, then the even or odd signal portion of such a transformation block multiplied by the factor two appears in this scheme as is shown in Fig. 4.

The reflected signal components corresponding to the signal components N_i are indicated by S_i . The negative sign for the odd signal portion indicates that the reflected signal components appear negative relative to the corresponding original components.

The sampling of the Fourier transforming of the even or of the odd signal portion leads to the corresponding signal components being made periodical in the time domain. Should no aliasing interference arise in the time domain then the signal components that have been made periodic should not be allowed to overlap.

This means that in the limiting case a sampling period of $F_0 = 1/(2TB)$ must be used in the spectrum. With the aid of the scheme e.g. for the cosine transformation, this periodisation leads to the representation shown in Fig. 5.

The position of the beginning of the block in the co-ordinate system used for the transformation is subject to a degree of freedom which until now was established relatively arbitrarily. Previously, the beginning of the block and the zero point of the co-ordinate system coincided. The Figures 6 and 7 show, on the basis of the example of the cosine transformation, a displacement of the beginning of the block to positive times t in the co-ordinate system and the consequences of a subsampling by the factor 2 in the spectrum i.e. a sampling period of $F_0 = 1/TB$. The various signal components in the time domain after transformation, subsampling, transmission and inverse transformation for a displacement of the transformation block from the origin of the co-ordinate system of $TB/2$ are depicted in Fig. 6. Fig. 7 shows the signal components for a displacement of $TB/4$.

It is apparent from these two examples what effect this important degree of freedom has on the components which are located in the zone that is faded out from the periodically recurring signal portions by the synthesis window in the receiver. This time zone has the duration TB in both illustrations and the position identified in the two pictures. The synthesis window region contains the original signal segments $N1$ and $N2$ on the one hand and the aliasing components on the other. In the case of the twofold block overlapping, the components $S1$ and $S2$. The Figures 6 and 7 show that the displacement of the beginning of the block relative to the origin of the co-ordinate system determines on the one hand, at which location in the block region the signal components occurring due to the mirroring appear but, on the other hand, it has no effect upon signal components having a non-inverted temporal position.

Since the method of aliasing compensation requires that the aliasing components be compensated by addition of the overlapping regions of the blocks, only signal portions which originate from one block half of the original signal may be located in this block half for a twofold overlapping. This requirement is not fulfilled by a time displacement of the magnitude $TB/2$ since here, reflected signal portions from the second half of the block fall in the first half of the block and vice versa. In this case, for a block overlapping, n components which originate from the overlapping region $n-1$ as well as $n+1$

appear in the overlapping region. Since these components only appear once in the respective overlapping region, they cannot compensate each other. Fig. 8 shows for successive blocks which signal components are basically involved in the addition in the overlapping region for the displacements $TB/2$ and $TB/4$.

Since only a time displacement of $TB/4$ contains the fundamentally correct components, such that, after an addition in the overlapping region the useful signal occurs on the one hand and the aliasing components are self compensating on the other, this displacement of the beginning of the block from the origin of the co-ordinate system is used for the cosine or sine transformation.

A further degree of freedom is provided by the sampling scheme in the frequency domain. Herein, it is possible to either let the sampling begin at the frequency $f=0$ or else to introduce a sampling displacement which corresponds to half the sampling period for the sampling in the frequency domain. Other displacement magnitudes lead to a doubling of the data rate due to an unsymmetrical sampling of the symmetrical picture function of the cosine or sine transformation in the Fourier spectral range. After the transformation, subsampling, transmission and inverse transformation, the displacement by half the sampling period for the sampling in the spectrum causes an alternating sign change in the time domain during the periodisation of the even or of the odd block portions. Fig. 9 shows this with the scheme used for the cosine transformation.

For the cosine transformation, it can be gathered from Figures 7 and 9 which signal components are located in the region that is blocked out by the synthesis window after the transformation, subsampling, transmission and inverse transformation. If one takes into account that for the sine transformation only the mirrored components have a negative sign as compared with those for a cosine transformation, then for this too, one can indicate the signal components in the synthesis window region. The signal components which are present in the synthesis window region after the transformation, subsampling, transmission and inverse transformation are indicated in Fig. 10 for the four possible variants of transformation, cosine transformation with and without sampling displacement in the spectrum and sine transformation with and without sampling displacement in the spectrum.

The mechanics of the aliasing compensation for a plurality of successive blocks when using a displacement of $TB/4$ is now shown in Fig. 11. Of the possible variants for the transformation,

cosine and sine transformation without sampling displacement are used alternately here for the blocks succeeding each other in time. The aliasing components thereby receive firstly a positive and then a negative sign.

Rectangular windows were used as the analysis and synthesis window functions in the example indicated above. However, these windows exhibit a particularly poor behaviour as regards selectivity in the spectrum region. Better results are achieved if "soft" edged window functions are used for the analysis and synthesis.

For the use of window functions which deviate from the rectangular window function, two conditions still have to be observed:

1. the useful signal has to be correctly reconstructed by means of the superimposition,
2. the aliasing interference must still be mutually compensating.

In order to be able to set up the equations which are necessary for the maintenance of the two conditions mentioned above, a scheme must likewise be utilised that corresponds to the previously described scheme and extends it.

If one segments the analysis window and the synthesis window in the same manner as the signal, there results a diagram in accordance with Fig. 12.

In correspondence with the signal components, the components of the analysis or of the synthesis window which are regularly aligned in time are designated as a_{ni} or s_{ni} . Components of the analysis window which are inverted in time due to the mirroring are designated a_{si} . If now, in Fig. 11, one uses the window segments which were involved in the reconstruction of the current signal components, there then ensues a diagram according to Fig. 13. The equations, which analysis and synthesis window function must observe in order that after the addition in the overlapping region the signal itself is reconstructed on the one hand and the aliasing interference is compensated on the other, can be read out from this diagram with the aid of Fig. 11.

Two equations for the block overlapping regions ensue from Fig. 13:

$$a_{n1} * s_{n1} + a_{n2} * s_{n2} = 1 \quad (5) \text{ (reconstruction requirement)}$$

$$-a_{s1} * s_{n1} + a_{s2} * s_{n2} = 0 \quad (6) \text{ (compensation requirement)}$$

If one uses identical window functions for all of the transformations succeeding each other in time and if one references the analysis window functions with $a(x)$ wherein x is to represent a normalised time which has the value 0 at the beginning of the block and the value 1 at the end of the block and also the synthesis window functions with $s(x)$, then one derives from the equations (5) and (6) for the overlapping region $0 \leq x \leq 0.5$:

$$a(x) * s(x) + a(x+0.5) * s(x+0.5) = 1 \quad (7)$$

$$a(0.5-x) * s(x) - a(1-x) * (0.5+x) = 0 \quad (8)$$

These two equations must be fulfilled by the window functions. If one chooses the special case that the window functions are symmetrical about the middle of the block and assumes that the analysis window function is given then there arises from the equations (7) and (8) a construction requirement for the synthesis window function:

$$s(x) = \frac{a(x)}{a^2(x) + a^2(x+0.5)} \quad (9)$$

The method discussed with the example of twofold overlapping will now be systematically extended to greater block overlapping namely the fourfold and eightfold overlapping.

The degrees of freedom, displacement of the zero point of the block in the time co-ordinate system and displacement during the sampling in the spectral region, are determined in accordance with the above considerations. This leads to a displacement of the zero point of the block relative to the origin of the co-ordinate system by an eighth of the block length T_B . The displacement zero and the displacement by a half sampling period during the sampling in the spectrum are also permissible in this case. There thus also results for the fourfold block overlapping, four modifications of the transformation. The signal components which are present in the region of the synthesis window after the transformation, subsampling, transmission and inverse transformation are shown in Fig. 14 for

all valid transformations, initially for usage of rectangular window functions for the analysis and synthesis. Analogously with the case of the twofold overlapping, a segmenting of the signal with the extension in time of the common overlapping region of TB/4 was effected here.

On the assumption of aliasing compensation with the use of rectangular windows for the analysis and synthesis, the following time sequence of transformations occurs:

1. Sine transformation with subsampling by the factor 4, displacement by TB/8 in the time domain and displacement by half a sampling period during sampling in the spectrum;
2. Sine transformation with subsampling by the factor 4, displacement by TB/8 in the time domain and no displacement during sampling in the spectrum;
3. Cosine transformation with subsampling by the factor 4, displacement by TB/8 in the time domain and displacement by half a sampling period during sampling in the spectrum;
4. Cosine transformation with subsampling by the factor 4, displacement by TB/8 in the time domain and no displacement during sampling in the spectrum;

If one utilises this time sequence of transformations then, with the scheme that was used previously, there arises the diagram in accordance with Fig. 15.

If, here again, one also carries out a corresponding segmenting for the analysis and synthesis windows with the designations a_{n1} , a_{s1} and s_{n1} for the individual segments in the original time orientation and in the inverted time orientation, then, here too, one can state the equations which the window functions generally have to fulfil:

$$a_{n1}s_{n1} + a_{n2}s_{n2} + a_{n3}s_{n3} + a_{n4}s_{n4} = 1 \quad (10)$$

$$a_{s1}s_{n1} + a_{s2}s_{n2} - a_{s3}s_{n3} - a_{s4}s_{n4} = 0 \quad (11)$$

$$a_{n1}s_{n3} = a_{n2}s_{n4} \quad (12)$$

$$a_{n3}s_{n1} = a_{n4}s_{n2} \quad (13)$$

$$a_{s1}s_{n3} = a_{s2}s_{n4} \quad (14)$$

$$a_{s3}s_{n1} = a_{s4}s_{n2} \quad (15)$$

If, for the sake of simplicity, one again assumes in this case too that the window functions are functions that are each symmetrical about the centre of the block, then one obtains a construction requirement which the analysis window must observe from the equations (12) to (15). This requirement is given in a

manner of writing in which x again corresponds to the normalised time. This is summed up by the equations (16) and (17):

$$a(0.5-x)*a(x) = a(0.25+x)*a(0.25-x) \quad (16)$$

$$a(1-x)*a(x+0.5) = a(0.75+x)*a(0.75-x) \quad (17)$$

$$\text{for } 0 \leq x \leq 0.25$$

With the provision of an analysis window that adheres to the requirements (16) and (17), a corresponding synthesis window can be calculated by means of the system of equations (18).

$$\begin{vmatrix} 0 & 0 & an1 & -an2 \\ an1 & an2 & an3 & an4 \\ an4 & an3 & an2 & an1 \\ an3 & -an4 & 0 & 0 \end{vmatrix} * \begin{vmatrix} sn1 \\ sn2 \\ sn3 \\ sn4 \end{vmatrix} = \begin{vmatrix} 0 \\ 1 \\ 0 \\ 0 \end{vmatrix} \quad (18)$$

If one does not assume that an aliasing compensation should occur when using rectangular windows but rather that the aliasing components should compensate one another only when utilising a specific window function which is different from a rectangular window, then one can deviate from the above mentioned time sequence of transformations and utilise any arbitrary sequence of transformations. The equations (10) to (15) and thence the constructional requirements for the analysis and synthesis windows are then altered if necessary by another transformation sequence. However, since the signal components for all transformations in the synthesis window region are the same and merely differ in their sign, the structure of the equations (10) to (15) with their segments of analysis and synthesis windows is maintained and only the signs are altered in these six equations due to another time sequence of transformations. However, for the constructional requirement in the equations (16), (17) which the analysis window must adhere to, this means that here, there are basically only two different requirements. Firstly, the signs given in the equations (16), (17) and secondly one with a negative sign at one side of the equality sign in both equations. As an example of such a case, the alternate utilisation of sine and cosine transformation without sampling displacement in the spectrum will be given. Fig. 16 depicts the scheme for this sequence of transformations

on the assumption of rectangular windows for the analysis and synthesis.

The equations for the analysis and synthesis windows then correspond to:

$$an1*sn1 + an2*sn2 + an3*sn3 + an4*sn4 = 1 \quad (19)$$

$$as1*sn1 - as2*sn2 + as3*sn3 - as4*sn4 = 0 \quad (20)$$

$$an1*sn3 = -an2*sn4 \quad (21)$$

$$an3*sn1 = -an4*sn2 \quad (22)$$

$$as1*sn3 = as2*sn4 \quad (23)$$

$$as3*sn1 = as4*sn2 \quad (24)$$

and thus the constructional requirement for the analysis window:

$$a(0.5-x)*a(x) = -a(0.25+x)*a(0.25-x) \quad (25)$$

$$a(1-x)*a(x+0.5) = -a(0.75+x)*a(0.75-x) \quad (26)$$

$$\text{for } 0 \leq x \leq 0.25$$

The considerations made in connection with the fourfold overlapping can be transferred systematically to higher degrees of overlap e.g. the eightfold overlapping. The displacement between the beginning of the block and the zero point of the coordinate system now has the magnitude $TB/16$. Furthermore, only the four variants of the transformation as for the fourfold overlapping can again be used. It is also true here that the time transformation sequence only has an effect on the sign of the signal components but not on their position in time in the synthesis window region after transformation, subsampling, transmission and inverse transformation.

From analogous considerations, there arise 14 equations which the analysis and synthesis windows must observe for the following transformation sequence:

1. Sine transformation with subsampling by the factor 4, displacement by $TB/8$ in the time domain and displacement by half a sampling period during sampling of the spectrum;
2. Sine transformation with subsampling by the factor 4, displacement by $TB/8$ in the time domain and no displacement during sampling of the spectrum;
3. Cosine transformation with subsampling by the factor 4, displacement by $TB/8$ in the time domain and displacement by half a sampling period during sampling of the spectrum;

4. Cosine transformation with subsampling by the factor 4, displacement by TB/8 in the time domain and no displacement during sampling of the spectrum;

Three equations, which are to be understood as the conditions which the analysis window has to fulfil, can be extracted therefrom. On the premise of axially symmetrical analysis and synthesis window functions, the conditions are:

$$a(x)*a(0.25-x) = a(0.125+x)*a(0.125-x) \quad (27)$$

$$\begin{aligned} & a(x)*a(0.5-x) + a(0.125+x)*a(0.375-x) + \\ & + a(0.25+x)*a(0.25-x) + a(0.375+x)*a(0.125-x) = 0 \end{aligned} \quad (28)$$

$$\begin{aligned} & a(x)*a(0.25+x) + a(0.25+x)*a(0.5-x) + \\ & a(0.5-x)*a(0.25-x) = a(0.125+x)*a(0.375+x) + \\ & + a(0.375+x)*a(0.375-x) + a(0.375-x)*a(0.125-x) \end{aligned} \quad (29)$$

$$\text{for } 0 \leq x \leq 0.25$$

If a window function $a(x)$ is found which fulfils the above three conditions, the synthesis window can be calculated from the equation system (30).

$$\begin{vmatrix} 0 & 0 & 0 & 0 & 0 & 0 & a_{n1} & -a_{n2} \\ 0 & 0 & 0 & 0 & a_{n1} & a_{n2} & a_{n3} & a_{n4} \\ 0 & 0 & a_{n1} & -a_{n2} & a_{n3} & -a_{n4} & a_{n5} & -a_{n6} \\ a_{n1} & a_{n2} & a_{n3} & -a_{n4} & a_{n5} & a_{n6} & a_{n7} & a_{n8} \\ a_{n8} & a_{n7} & -a_{n6} & -a_{n5} & a_{n4} & a_{n3} & -a_{n2} & -a_{n1} \\ a_{n3} & -a_{n4} & a_{n5} & -a_{n6} & a_{n7} & -a_{n8} & 0 & 0 \\ a_{n5} & a_{n6} & a_{n7} & a_{n8} & 0 & 0 & 0 & 0 \\ a_{n7} & -a_{n8} & 0 & 0 & 0 & 0 & 0 & 0 \end{vmatrix} * \begin{vmatrix} s_{n1} \\ s_{n2} \\ s_{n3} \\ s_{n4} \\ s_{n5} \\ s_{n6} \\ s_{n7} \\ s_{n8} \end{vmatrix} = \begin{vmatrix} 0 \\ 0 \\ 0 \\ 1 \\ 0 \\ 0 \\ 0 \\ 0 \end{vmatrix}$$

equation (30)

The solutions for the calculation of analysis and synthesis windows of the differently overlapping blocks which have been given in the preceding paragraphs display a symmetrical construction. This is clear and recognisable from the matrix equations (18). The conditions which the analysis window function has to fulfil produce greater limitations for a realistic window design with increasing overlapping.

For a uniform comparison of the different block overlappings, a Kaiser Bessel window was selected as the basis. The Kaiser

Bessel window was modified in each case such that it satisfied the relevant conditions for the overlapping transformation. This always occurred by means of an expansion corresponding to the window conditions with the provision of the Kaiser Bessel window function in a partial region of the window. In the case of the twofold overlapping, the degrees of freedom for the analysis window are still so great that a complete Kaiser Bessel window can be selected here (Fig. 17). In the case of the fourfold overlapping, the Kaiser Bessel window has to be modified due to the limited degrees of freedom as are described by the equations (16) and (17). With the window construction in Fig. 18, a Kaiser Bessel window between the normalised time values $x=1/8$ and $x=7/8$ was provided and the remaining portions of the window appended by means of the equations (16) and (17). Since the limitations for the analysis window become ever larger with increasing overlapping, the Kaiser Bessel window can only be provided between $x=1/4$ and $x=3/4$ in the case of the eightfold overlapping (Fig. 19). The remaining portions of the window then had to be calculated by means of the equations (27) to (29).

The Figures 17 to 22 show the time waveform of the window function and the amount of the Fourier spectrum. The windows were displayed such that the effective width of the window i.e. the window width in which virtually all of the energy of the window lies, is the same for all sizes of overlapping. This width of window corresponds to the window width for the twofold overlapping. The nominal window width doubles for each doubling of the overlapping as the Figures 17 to 19 depict.

Through the choice of the Kaiser Bessel window as the analysis window function, the corresponding synthesis windows have increases greater than one. This behaviour has to be taken into account for an optimisation of the window on the basis of the current application since possible interference components which occur during the coding of the spectrum can be slightly raised thereby. If one compares all three time functions of the analysis window with one another, then the almost identical waveform in the region of the effective window width can be readily appreciated. From this, there results in the spectrum an identical width of the "main slope" of the magnitude waveform of the Fourier transforming of the analysis window function. The effect which the multiple overlapping has in the chosen example is clearly apparent from the Figures 20 to 22. Due to the doubling of the overlapping in each case and the lengthening of the original Kaiser Bessel window that is associated therewith, there results ever smaller window edge values. Due to this effect, there occur in the spectrum ever greater attenuations into which the spectral waveform flows after the "main slope".

This family of windows is to be seen as just an example for the method of operation of the multiple overlapping. The design of the window has to be matched to the case in hand. This means for example, that in the case of an analysis window for the eightfold overlapping, the "main slope" is smaller through an appropriate design if such a large attenuation is not needed after the "main slope".

Claims

1. A method for the transmission of a signal, in which the analog signal is converted into a digital signal, transmitted digitally and converted back into an analog signal (Fig.1), and in which the signal is divided by means of mutually-overlapping time windows into blocks following each other in time, which are each converted into a signal sequence representing the impulsive spectrum, characterised by,

a) assigning the signals to windows and forming time blocks with block length TB with overlapping regions with relative sizes of $N-1/N$ with $N = 2^n$ for integral values of n;

b) subjecting each of the individual blocks displaced by $TB/2 \cdot N$ in the time domain to a sine- or cosine-transformation;

c) subsampling by selection of each Nth value according to the schedule: C1: 0, N, 2N, 3N.... or C2: N/2, 3N/2, 5N/2..... whereby there arise four combinations with reference to transformation and form of subsampling:

- transformation and subsampling form
- K1: Cosine transformation + selection schedule C1
- K2: Cosine transformation + selection schedule C2
- K3: Sine transformation and selection schedule C1
- K4: Sine transformation and selection schedule C2;

d) applying one of the combinations K1....K4 in any permutation to each of the blocks involved in the



overlapping, whereby there arise four value groups with items for the overlapping regions of the blocks, which differ merely by their sign;

- e) selecting the combinations $K_1...K_4$ to be applied according to the criterion that after inverse transformation and summation of the components in the signal portions of the blocks involved in the overlapping all those signal components are compensated which do not originate from the same segment of the original signal;
- f) coding, transmission and decoding;
- g) subjecting each of the individual blocks to an inverse sine- or cosine-transformation;
- h) segmenting the time - continuous signal into successive signal portions N_i for $i = 1, 2, 3, \dots$, which according to the combination $K_1...K_4$ used are contained in addition to the components $N_i...$ the components $S_i...$ corresponding to these components reflected in time;
- i) summing the components $N_i...$ and $S_i...$ in the signal portions N_i of the blocks involved in the overlapping.

2. A method according to claim 1, characterised in that:

- a) before the transformation the blocks are weighted by analysis windows and after the transformation by synthesis windows, which form segments of the same length TB/N as the signal portions;



b) the weighting occurs by multiplication of the signal components $Ni...$ or the components $Si...$ corresponding to these components reflected in time by the components of the analysis window $ani...$ or the components $asi...$ corresponding to these components reflected in time and the components of the synthesis window $sni...$;

c) the analysis and synthesis window functions used in the weighting must fulfil the following conditions in the overlapping regions:

I) the sum of the regularly occurring signal components of the analysis window and synthesis window superimposed in the signal portions of a block amounts to 1;

(II) the sum of the regularly occurring aliasing components of the analysis window and synthesis window superimposed in the signal portions of a block amounts to 0.

3. A method according to claim 2, characterised in that in the case of window functions symmetrical about the centre of the block and with double overlapping the synthesis window function is determined from a predetermined analysis window function according to the following equation:

$$s(x) = \frac{a(x)}{a^2(x) + a^2(x+0,5)}$$

for $0 \leq x \leq 0,5$

in which $s(x)$ is the synthesis window function,
 $a(x)$ is the analysis window function,



x represents a normalised time, having the value 0 at the beginning of a block and the value 1 at the end.

4. A method according to claim 2, characterised in that in the case of window functions symmetrical about the centre of the block and with fourfold overlapping there is first determined an analysis window function, which fulfils the following equations:

$$a(0,5-x)*a(x) = a(0,25+x)*a(0,25-x)$$

$$a(1-x)*a(x+0,5) = a(0,75+x)*a(0,75-x)$$

for $0 \leq x \leq 0,25$

in which $a(x)$ is the analysis function, and then the synthesis window function is determined from the previously determined analysis window function according to the following equation system:

$$\begin{vmatrix} 0 & 0 & an1 & -an2 \\ an1 & an2 & an3 & an4 \\ an4 & an3 & an2 & an1 \\ an3 & -an4 & 0 & 0 \end{vmatrix} * \begin{vmatrix} sn1 \\ sn2 \\ sn3 \\ sn4 \end{vmatrix} = \begin{vmatrix} 0 \\ 1 \\ 0 \\ 0 \end{vmatrix}$$

wherein

sn_i are the components of the synthesis window function, an_i are the components of the analysis window function, x represents a normalised time, having the value 0 at the beginning of a block and the value 1 at the end.



5. A method according to claim 3, characterised in that in the case of window functions symmetrical about the centre of the block and with eightfold overlapping there is first determined an analysis window function, which fulfils the following equations:

$$a(x)*a(0,25-x) = a(0,125+x)*a(0,125-x)$$

$$a(x)*a(0,5-x) + a(0,125+x)*a(0,375-x) + \\ + a(0,25+x)*a(0,25-x) + a(0,375+x)*a(0,125-x) = 0$$

$$a(x)*a(0,25+x) + a(0,25+x)*a(0,5-x) + \\ + a(0,5-x)*a(0,25-x) = a(0,125+x)*a(0,375+x) + \\ + a(0,375+x)*a(0,375-x) + a(0,375-x)*a(0,125-x)$$

for $0 \leq x \leq 0,125$

in which $A(x)$ is the analysis function, and then the synthesis window function is determined from the previously determined analysis window function according to the following equation system:

0	0	0	0	0	0	an1	-an2	sn1	0
0	0	0	0	an1	an2	an3	an4	sn2	0
0	0	an1	-an2	an3	-an4	an5	-an6	sn3	0
an1	an2	an3	an4	an5	an6	an7	an8	sn4	1
an8	an7	-an6	-an5	an4	an3	-an2	-an1	sn5	0
an3	-an4	an5	-an6	an7	-an8	0	0	sn6	0
an5	an6	an7	an8	0	0	0	0	sn7	0
an7	-an8	0	0	0	0	0	0	sn8	0

*

in which

sn1 are the components of the synthesis window function,



ani are the components of the analysis window function, x represents a normalised time, having the value 0 at the beginning of a block and the value 1 at the end.



* C L M S *