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Deutsche Thomson-Brandt GmbH

7730 Villingen-Schwenningen

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Method for transmitting a signal

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

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

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

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

However, this measure usually results in a

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

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

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

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

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

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

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

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

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

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

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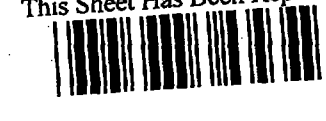

means of the features described in the characterizing part of Claim 1.

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

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

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

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

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

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

Fig. 2 shows asymmetrical window functions and

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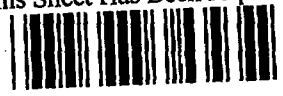

Fig. 3a shows the profile of an input signal with amplitude jump,

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

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

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

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

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

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

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

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

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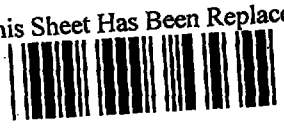

narrow. The window functions of the adjacent windows 2 and 4, that is to say of the preceding window and of the following window have a correspondingly adapted window function in the overlapping area so that in these areas the window functions complement one another to form one. Thus, the window functions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

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

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

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

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

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

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

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

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

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

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

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

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

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Patent claims

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

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

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

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

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

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

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

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

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

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

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

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

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Hanover, 8 June 1990
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Abstract

1. Method for transmitting a signal

2.1 In a method for transmitting a signal which is divided by means of windows into successive overlapping blocks, the signal elements which are contained in the blocks are each converted by transformation into a spectrum, the spectra are subsequently encoded, transmitted, decoded after the transmission and converted back into signal elements by inverse transformation. Finally, the blocks contained in the signal elements are adjoined in an overlapping fashion, the overlapping areas of the block being weighted in such a way that the resultant of the window functions is one.

2.2. In order to avoid faults in adjacent blocks in the case of frequency changes, the window functions and the length of the window functions are selected as a function of frequency changes of the input signal.

2.3. The method is suitable for treating audio signals and video signals which are subjected to a data reduction during transmission.

3. Figure 4a

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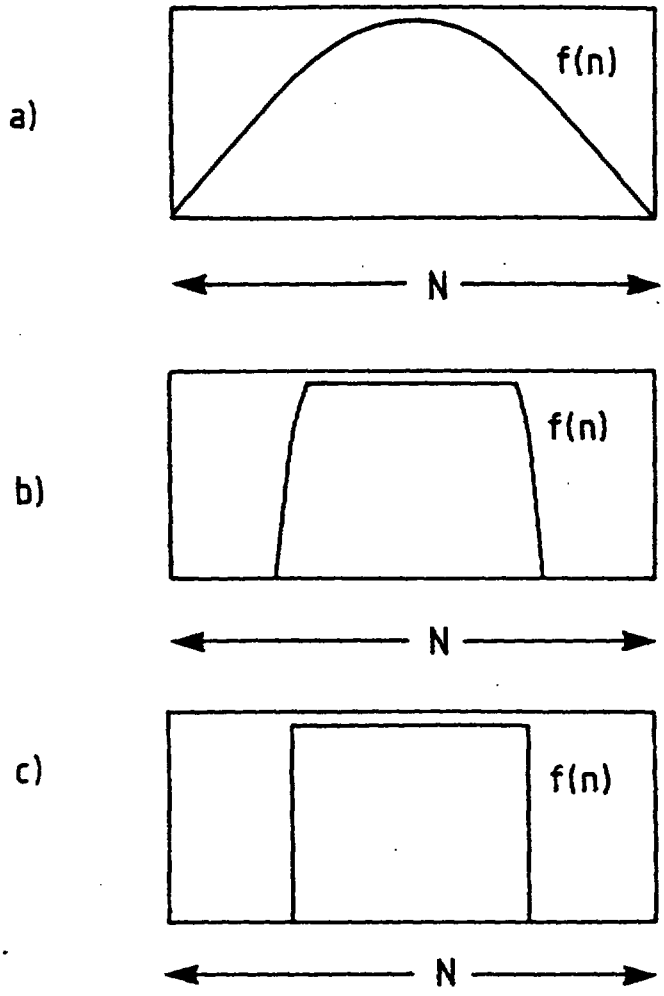

FIG.1

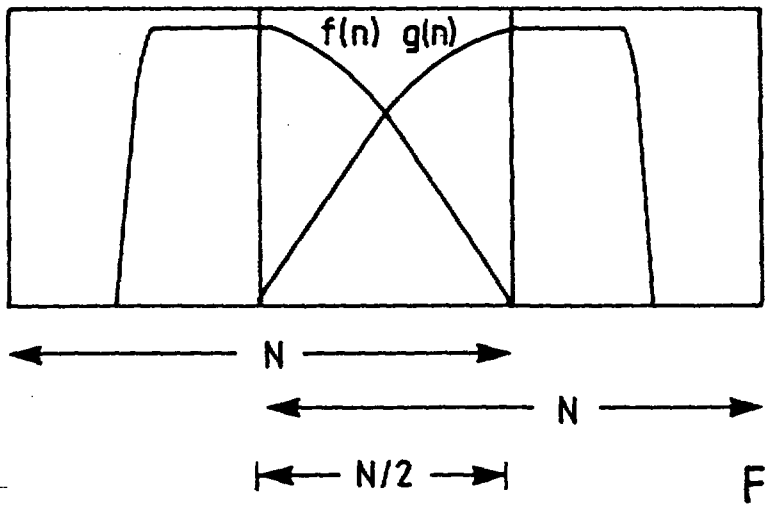

FIG.2

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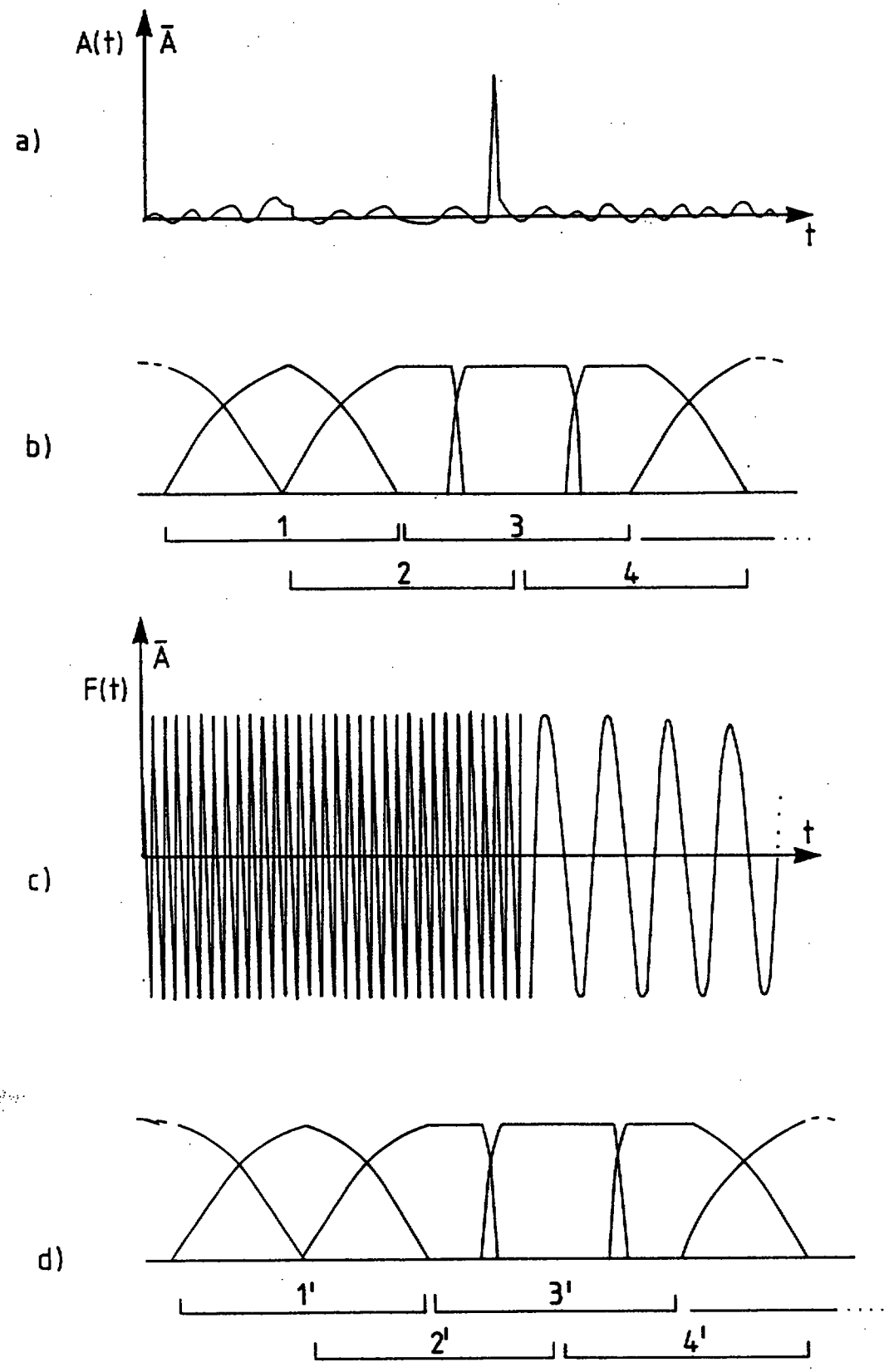

FIG.3

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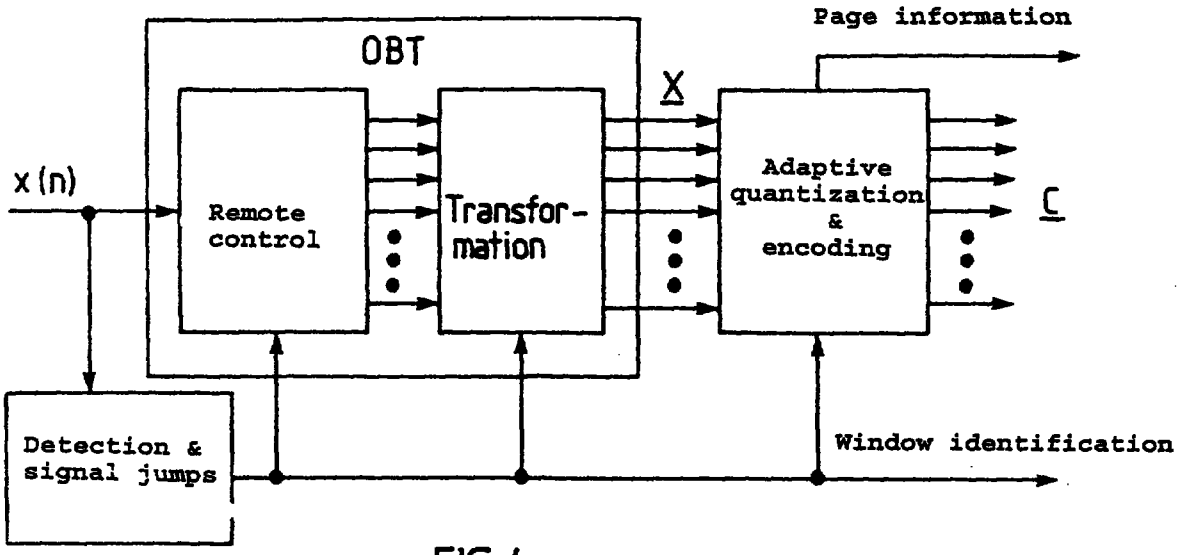

FIG.4a

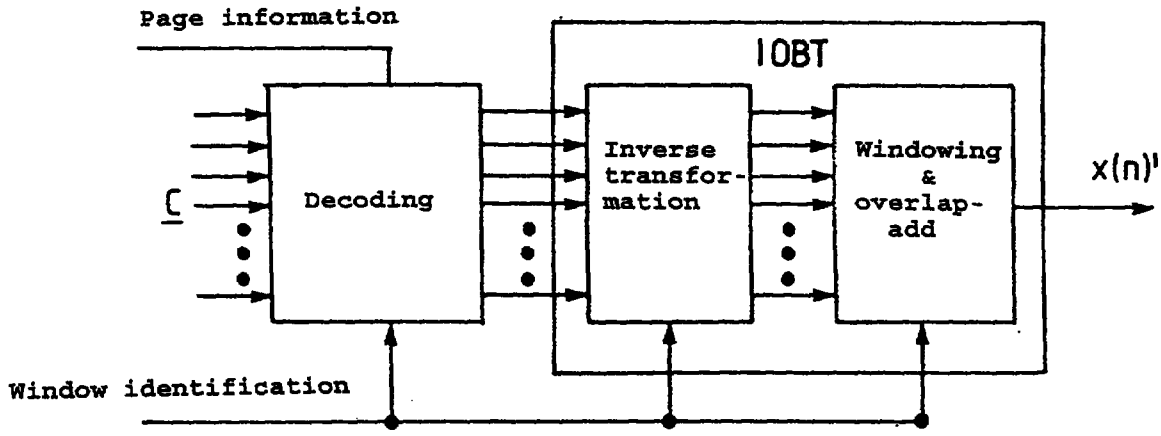

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FIG.4b

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☒ Claim(s) : paged 10 to paged 11
☒ Drawing(s) : paged 1/3 to paged 3/3
☒ Abstract : paged 12 to paged 12
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Fig. 3c shows the profile of an input signal with frequency jump,

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

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

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

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

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

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

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

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

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

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

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

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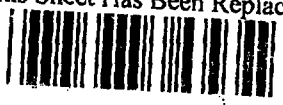

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advance of one block of N sampled values in the time domain and a transformation of two N sampled values, in turn only N sampled values are also obtained in the image range so that no additional coefficients have to be transmitted. The OBT also fulfils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

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

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

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

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

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Patent claims

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

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

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

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

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

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

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

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

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

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

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

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

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Abstract

Method for transmitting a signal

In a method for transmitting a signal which is divided by means of windows into successive overlapping blocks, the signal elements which are contained in the blocks are each converted by transformation into a spectrum, the spectra are subsequently encoded, transmitted, decoded after the transmission and converted back into signal elements by inverse transformation. Finally, the blocks contained in the signal elements are adjoined in an overlapping fashion, the overlapping areas of the block being weighted in such a way that the resultant of the window functions is one.

In order to avoid faults in adjacent blocks in the case of frequency changes, the window functions and the length of the window functions are selected as a function of frequency changes of the input signal.

The method is suitable for treating audio signals and video signals which are subjected to a data reduction during transmission.

Figure 4a

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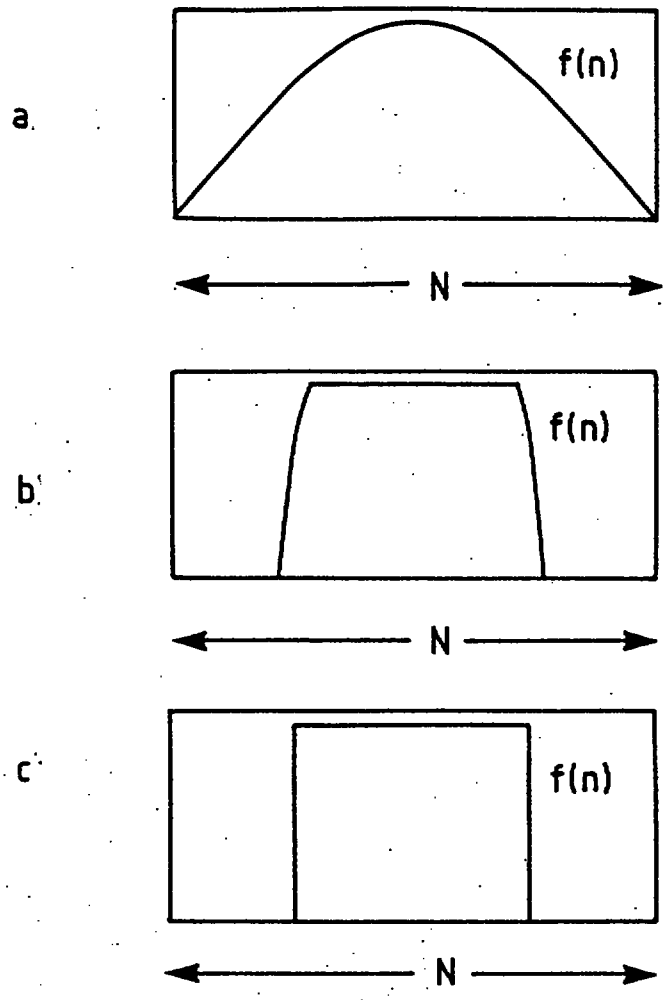

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FIG.1

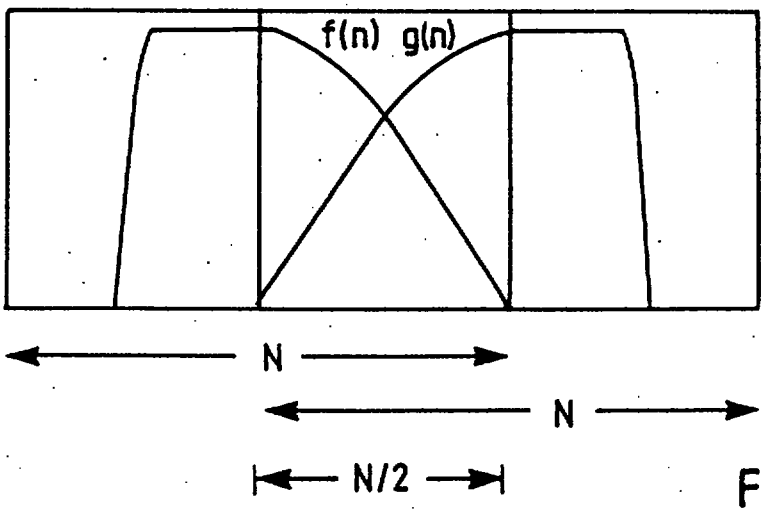

FIG.2

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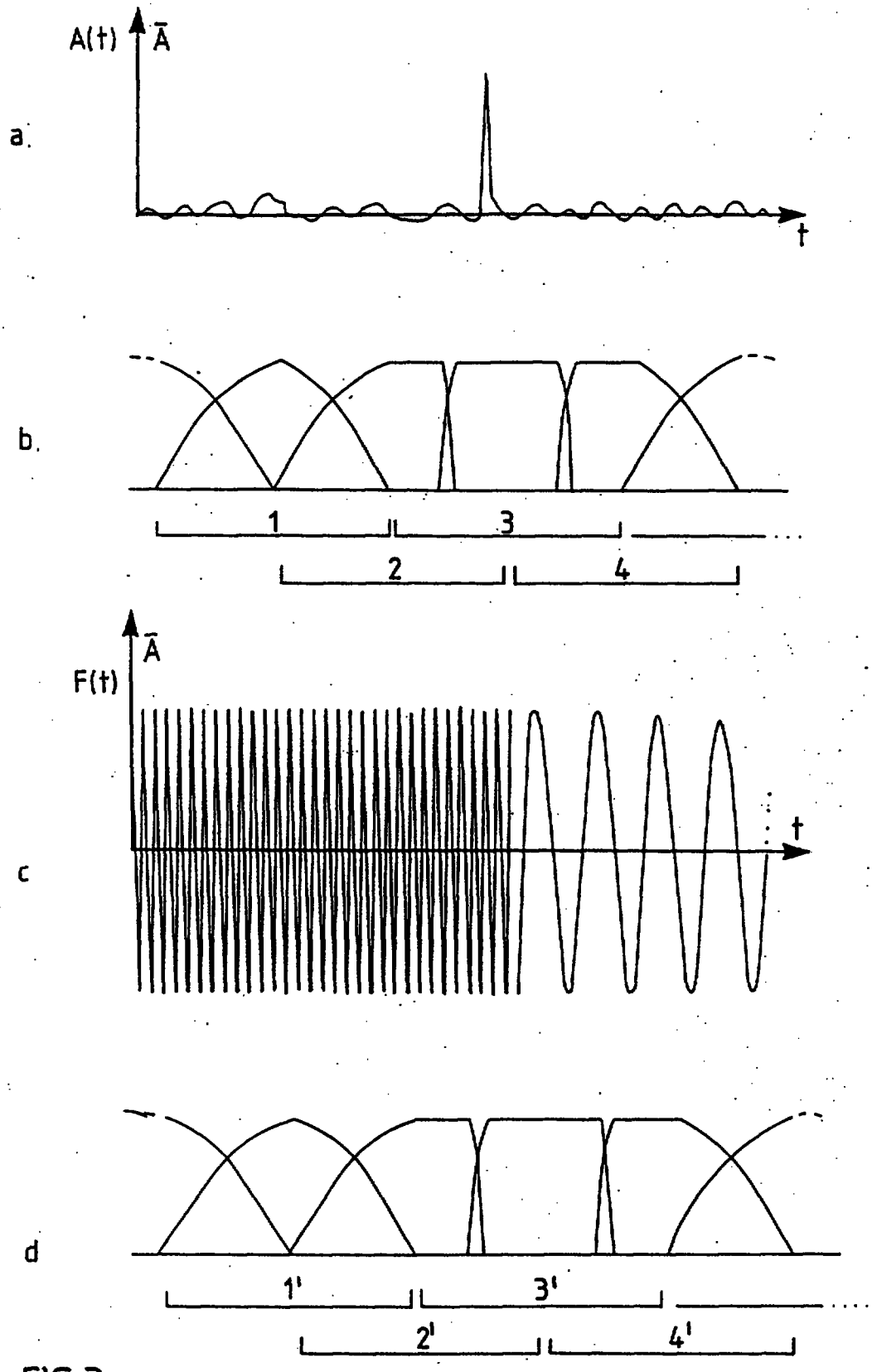

FIG.3

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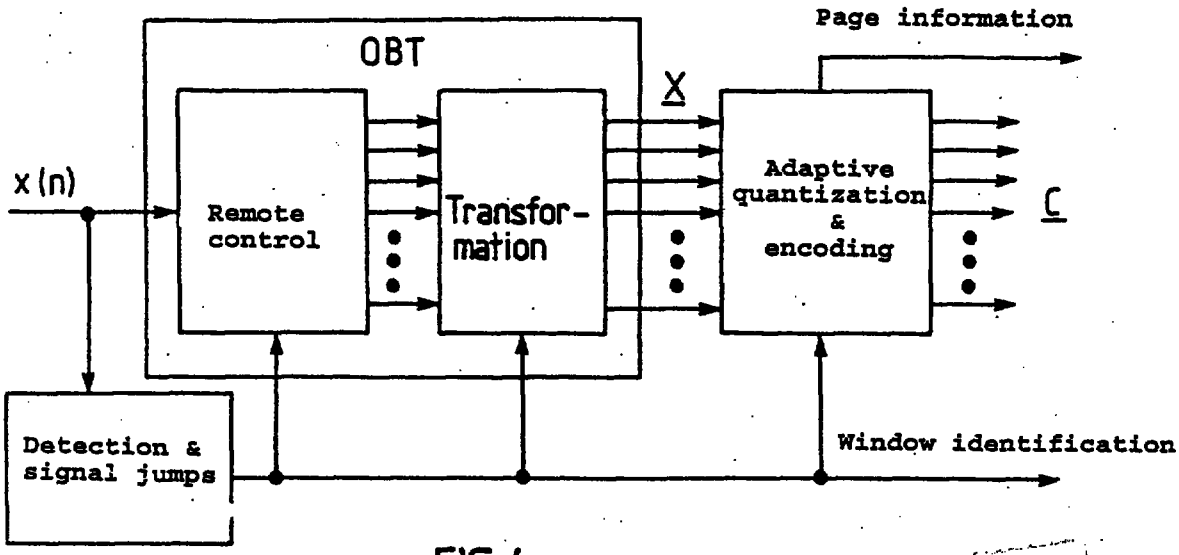

FIG.4a

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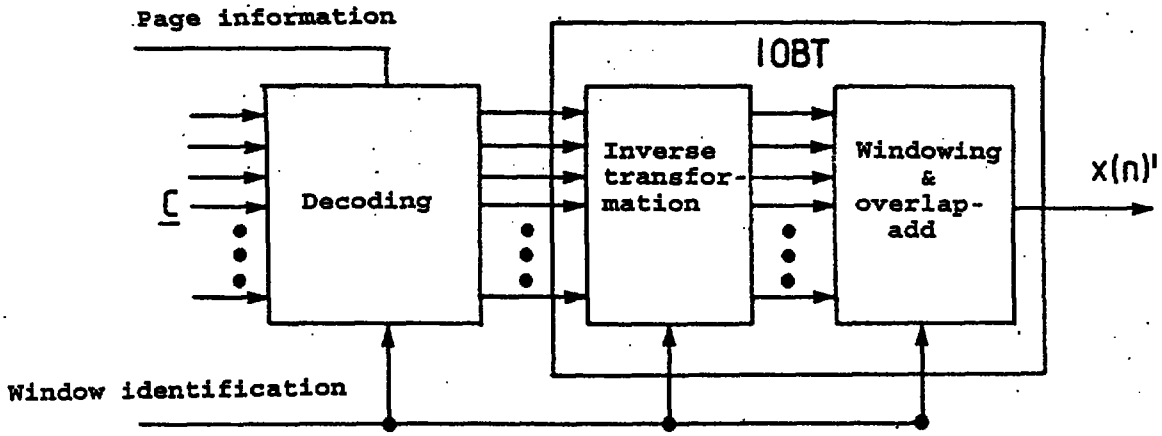

FIG.4b

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Transform coder with adaptive window function

The invention concerns a method for transferring a signal according to the preamble of claim 1.

When transferring an audio signal, e.g. with radio broadcast, cable transmission, satellite transmission and with recording devices, it is known to convert the analog audio signal into a digital audio signal with a defined resolution, to transmit it in this form, and to reconvert it into an analog signal upon reproduction. An increased signal-to-noise ratio is achieved by the digital transmission, particularly during reproduction.

The band width necessary for the transfer of such a signal is essentially determined by the number of sampling values per time unit which have to be transmitted as well as by the resolution.

In practice, there exists the requirement that the band width necessary for the transfer is to be kept as small as possible in order to make do with a narrow band channel or to be able to transfer as many audio signals as possible over an existing channel. The band width required can be itself lowered by reducing the sampling values or the number of bits per sampling value.

However, as a rule this measure results in a deterioration in the reproduction. In order to improve the reproduction quality, in a known method (DE OS 35 06 912.0) the digital audio signal in temporal successive segments is transformed into a short term spectrum which represents, for the respective time segments, e.g. 20 ms, the spectral components of the signal. Because of psycho-acoustic laws, components which are not perceived by the listener, and are therefore irrelevant from the point of view of providing information, can usually be more easily found in the short term spectrum than in the time domain. These components are less heavily weighted or completely left out upon transfer. By means of this measure, a considerable part of the otherwise necessary data can be omitted during transfer so that the average bit rate can be significantly reduced.

The method described by J.P. Princen and A.B. Bradley in "Analysis/Synthesis Filter-bank Design used on Time Domain Aliasing Cancellation", IEEE Transactions

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Acoustics, Speech, Signal Processing, volume ASSP-34, pages 1153 - 1161, October 1986, is suitable for the partitioning of the signal into segments. This describes a transformation in which overlapping blocks with rounded-off window functions are generated in the windows without additional coefficients in the frequency domain. In this method, N values are firstly cut out from the input signal by means of a window function f(n) of length N, and subsequently transformed into N/2 significant coefficients in the frequency domain. The inverse transformation calculates from the N/2 coefficients, N sampling values, which are again weighted using the window function f(n).

However, the output signal of the inverse transformation differs from the input signal of the forward transformation. The precise reconstruction of the input signal is only made possible when the output values of successive inverse transformations are added in the overlapping region of N/2 sampling values in each case. In order that the input signal can be recovered by means of this so-called "overlap-add", the window function f(n) must comply with the following conditions:

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

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

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

The choice of window length when using this method in transformation coders leads to the following results. A large window length with a shape which is as rounded-off as possible enables a good frequency selectivity. Here, the error extends over the entire effective window length due to quantization of the coefficients after the inverse transformation. This can have a negative effect on the subjective quality of the coded signal, especially with large jumps in the amplitude of the signal which is to be coded.

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The choice of a shorter window causes a deterioration in the frequency selectivity which has a negative effect on the transformation gain to be achieved, particularly with strongly correlated input signals. On the other hand, errors can be limited to the window concerned by quantizing the coefficients in case of large signal jumps so that their effects on neighbouring windows are avoided.

In "Frequenz", Vol.43,No.9, September 1989, pages 252-256 there was proposed for the signal to be coded an adaptive window changeover controlled by the signal energy, in which in particular a high-pass filter is used for the recognition of amplitude jumps within signal blocks. In IEEE Transactions on Acoustics, Speech, and Signal Processing, Vol. ASSP-27, No.4, August 1979, pages 328-335, there is described an adaptive window controlled by a "Pitch Detection" for use in a cepstrum.

It is the object of the invention to effectively improve a method for transferring a signal so that there is achieved an optimum frequency selectivity and a high subjective quality for the coded and decoded signal. This object is achieved by the features of claim 1. It is the further object of the invention to provide a coder device for implementation of the method according to the invention. This object is achieved by the device of claim 10.

The invention is based on the idea that a window change-over is also useful when the useful signal has frequency jump with constant energy. In certain cases, this could be even necessary because of the psycho-acoustic circumstances. For this, the useful signal frequency range (for example, 0 ... 20 kHz) is partitioned into at least two, according to psycho-acoustic viewpoints, sensible, separate frequency ranges. The input signal is separately evaluated in the respective frequency ranges and, following that, a logical combination of the separate results is carried out.

In a further development of the invention, the change in frequency is also used for the filter band change-over of a sub-band coder (adaptive windowing), whereby, as a final result, the transformation gain is increased.

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Moreover, by using the frequency jump recognition, an adaptive quantization and coding can also be controlled like audio coding methods such as, for example, NICAM, MUSICAM, MSC, etc. which permit a variable block length. Furthermore, the use of frequency jump recognition allows the time and frequency of occurrence of scale factors, bit allocation factors, etc. to be calculated so that there is possible an improved controllability and observability of the transformation and the above-mentioned audio coding method.

It is advantageous if an amplitude change recognition is undertaken in addition to the frequency change recognition, and if the adaptive windowing, quantization and coding are carried out by logical combination of the two results.

In the following, the invention is more closely illustrated in a drawing with reference to one of the many possible embodiments. In the drawing:

Fig. 1a-c show window functions with differing widths,
Fig.2 shows asymmetric window functions, and
Fig.3a shows the course of an input signal with an amplitude jump,
Fig.3b shows a window function which is matched to the input signal shown in Fig.3a,
Fig.3c shows the course of an input signal with a frequency jump,
Fig.3d shows a window function which is matched to the input signal according to Fig.3c,
Fig.4a shows the block circuit diagram of a coder section of a transmitter,
Fig.4b shows the block circuit diagram of a decoder section of a receiver.

Several windows of constant width (b) and differing window functions f(n) are shown in Fig. 1. The windows are in each case put together in such a way that the neighbouring windows overlap by one half. This also results in an overlap by half in the sine wave window function f(n) in Fig. 1a, in the representation according to Fig. 1b in a partial overlap which is less than one half, and in Fig. 1c only in a touching of the two sides of the window function f(n).

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Fig.2 shows the overlaying of two windows with asymmetric window functions f(n) and g(n). However, the window functions are designed in their overlap area in such a manner that their resultants together equal one.

An amplitude characteristic A(t) of an input signal is shown in Fig.3a. As can be seen, at first the signal has a steady low amplitude which is followed by a signal jump whereupon the original amplitude continues. The window functions of the windows which enable optimum processing of this signal are shown underneath in Fig.3b. There are sine wave window functions in the first region 1 which render possible a high frequency selectivity. The window 3, in which the signal with the high amplitude occurs, possesses, in comparison, a window function which is very narrow. The window functions of the adjacent windows 2 and 4, i.e. the preceding and the following window, have an appropriately adapted window function in the overlap area so that the window functions in this area sum to one. Thus the window functions in these two adjoining windows 2 and 4 have an asymmetric structure. Quantization disturbances which would appear within window 3, in which the signal jump occurs, are thereby restricted to the area of the window function of window 3, and thus reduced to about one half of the temporal extent when compared to a window function with sine wave course. Despite the reduced frequency selectivity, a subjective improvement in audio quality results because of the masking effect of the signal jump.

The frequency path F(t) of an input signal is illustrated in Fig.3c. As can be seen, at first the signal has a constant frequency (for example, 10 kHz) whereupon there occurs after that a frequency jump to a significantly lower frequency (for example, 200 Hz). During this the amplitude of the signal remains constant. Window functions of the windows, with which this signal can be optimally processed, are shown in Fig. 3d.

In the first region 1 there are sine wave window functions which render possible a high frequency selectivity. The window functions of windows 2 and 3, in which the frequency jump occurs, posses an appropriately adapted window function in the overlap area in such a way that in this region the window functions sum to 1. Here the overlap

area is of very narrow form. Thus the window functions in the two adjoining windows 2 and 3 have an asymmetric structure. Quantization disturbances which would appear within window 3, in which the signal jump occurs, are thereby restricted to the area of the window functions of window 3, and therewith reduced to about one half of the temporal extent when compared to a window function with sine wave course. Thus, despite the reduced frequency selectivity, a subjective improvement in audio quality results because of the masking effect of the signal jump.

For the most simple frequency jump recognition, the useful signal with the useful signal frequency range (0 ... 20 kHz) is in this case partitioned into two separate frequency ranges (0 ... 1 kHz and 1 kHz ... 20 kHz). Each frequency range is now separately evaluated and following that, the separate results are logically combined with each other.

Under psycho-acoustical considerations, this leads to the following window change-over: Upon reduction of the signal energy in the upper frequency range and simultaneous increase in the lower frequency range, a window change-over must be executed because this is necessary from a psycho-acoustic point of view. Then, the case in the region of the frequency jump according to Fig.3c is present. A window change-over is not necessary with energy reduction in the upper range and no increase in the lower range because this case is not critical. Equally, a window change-over can be omitted if there is an energy increase only in the lower range. If there is an energy increase only in the upper range then a change-over must take place in the usual way. If there is an energy increase in both ranges then a change-over to small window lengths must occur because the signal must contain impulses.

Analysing filters, like those used already in sub-band coders, are suitable for the detection of frequency changes. However, there are also simpler filters (for example, minimum phase filters or filter banks without reconstruction possibilities) which can be considered.

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The block circuit diagram of a coder section in a transmitter is shown in Fig.4a. The coder shown in this case uses the OBT (overlapping block transform) type of transformation which belongs to the class of half overlapping transformations. Upon an advance of one block of N sampling values in the time domain and a transformation of two N sampling values, one again only obtains N sampling values in the picture area so that additional coefficients need not be transmitted. Moreover, the OBT fulfils the requirement that the transformation coefficients should correspond to the spectrum of the input signal. By using non-square-shaped windows, which is possible with the OBT, the block effects are also diminished and the frequency selectivity improved. By using adaptive windowing, which is possible with OBT, i.e. changing over the transformation length and the window shape as well as window length, the upper limit for N in this case is not determined by the masking time. Thus the adaptive windowing produces an additional improvement in that a good frequency resolution and a higher transformation gain are achieved through the size of the transformation length, and with appropriate window adaption, the pre-echoes from changing over to shorter transformation lengths can be suppressed. A signal jump recognition (a combination of amplitude jump and frequency jump recognition) is provided for the window adaption in order to detect changes of pulse, amplitude, phase and frequency in a block in advance. The signal jump recognition generates a window signal - hereinafter called window characteristic- by which the windowing, transformation and adaptive quantization and coding in the coder section of the transmitter are controlled. The pre-analysis of the input signal avoids the possibility of a data block being transformed and coded several times with differing transformation lengths, and it is then decided by comparison which of the coded values are to be transferred.

Together with the adaptive bit allocation and adaptive quantization, which exploit the masking characteristics of human hearing in the frequency range, the coder realized presents a possibility for taking into consideration the psycho-acoustic circumstances to a large extent.

By using window recognition, a corresponding adaptive decoding and inverse OBT can

be carried out in the decoder, see Fig. 4b, so that only the signals with psycho-acoustic relevance are sent out as output signal. The signal X(n)' has in fact, objectively seen, a lower information content than the signal X(n) but the difference cannot be perceived by human hearing so that a completely correct signal matching the psycho-acoustic conditions can be perceived.

The invention described in the foregoing is in no way restricted to the OBT, but moreover, can enable control of all known audio coding methods which permit variable block lengths like, for example, NICAM, MUSICAM, MSC. The use of frequency jump detection allows the time and frequency of occurrence of scale factors, allocation factors, etc. to be calculated.

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CLAIMS

1. A method for the coding, transmission or decoding of an audio signal (x), which is partitioned in the time domain by window functions into successive, overlapping windows, the part signals contained in the windows being respectively converted by transformation into a spectrum (X) and the spectra are coded and after a transfer are decoded and are changed back into part signals by inverse transformation and the blocks containing the part signals are placed overlapping each other, the overlapping regions of the blocks being weighted by the window functions so that the resultant of the window functions (f(n), g(n)) amounts to one, characterised in that the window functions are selected in dependence upon frequency changes of the signal evaluated in at least two different frequency ranges, wherein a window change-over is undertaken:
- in the case of the reduction in the signal energy in an upper frequency range and a simultaneous increase of the signal energy in a lower frequency range;
- in the case of the increase of the signal energy only in the upper frequency range;
- in the case of the increase of the signal energy in the upper and in the lower frequency range.

2. A method according to claim 1, wherein the window change-over occurs even with constant amplitude of the signals in the upper and lower frequency range.

3. A method according to claim 1 or 2, wherein the transformation length and the window length are changed adaptively.

4. A method according to one or more of claims 1 to 3, wherein the blocks have constant spacing and the lengths of the window functions in comparison to the length (N) of the blocks is selected in dependence on the frequency changes.

5. A method according to claim 1, wherein the spacings of the windows are selected in dependence on the frequency changes.

6. A method according to one or more of claims 1 to 5, wherein the length of the

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window functions is selected to be small with large frequency changes and to be large with small frequency changes.

7. A method according to any one of claims 1 to 6, wherein the window functions or their length or the transformation length are additionally selected in dependence on amplitude changes.

8. A method according to claim 7, wherein the length of the window functions is selected to be small with large amplitude changes and to be large with small amplitude changes.

9. A method according to one or more of claims 1 to 8, wherein the window function of a succeeding window in the overlapping region with that of the preceding windows is formed by reflection of the window function in this region.

10. A coder device for an audio signal (x), which is partitioned in the time domain by a windowing unit into successive overlapping windows, the part signals contained in the windows being respectively converted by a transformation unit into a spectrum (X) and these spectra are coded in a unit for adaptive quantisation and coding, the overlapping regions of the windows being so weighted by the window functions that the resultant of the window functions (f(n), g(n)) amounts to one, characterised in that the window functions can be selected by means of a window-recognition signal generated in a signal jump recognition unit in dependence upon frequency changes of the signal evaluation in at least two different frequency ranges, wherein a window change-over is undertaken:

- in the case of the reduction in the signal energy in an upper frequency range and a simultaneous increase of the signal energy in a lower frequency range;
- in the case of the increase of the signal energy only in the upper frequency range;
- in the case of the increase of the signal energy in the upper and in the lower frequency range.

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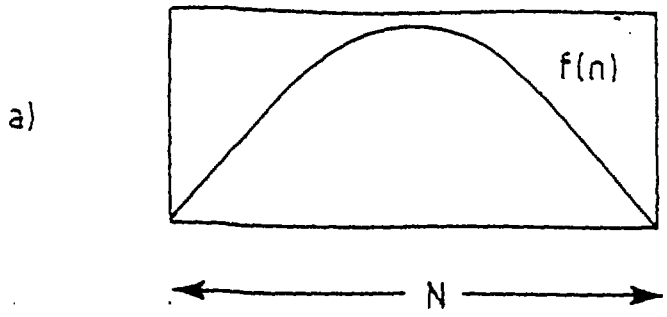

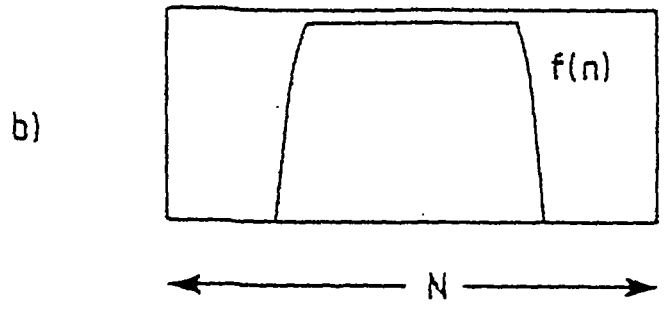

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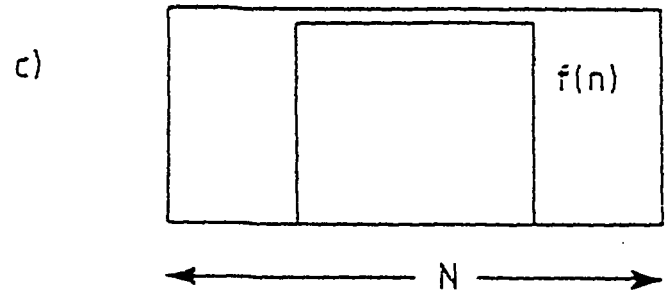

FIG.1

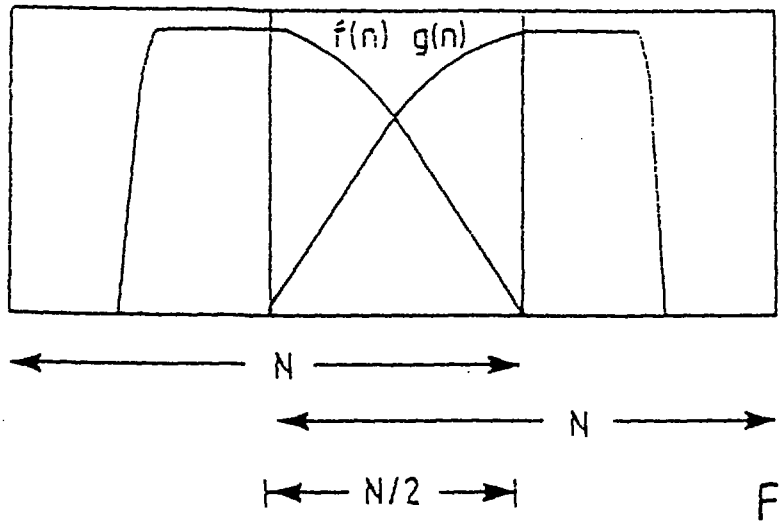

FIG.2

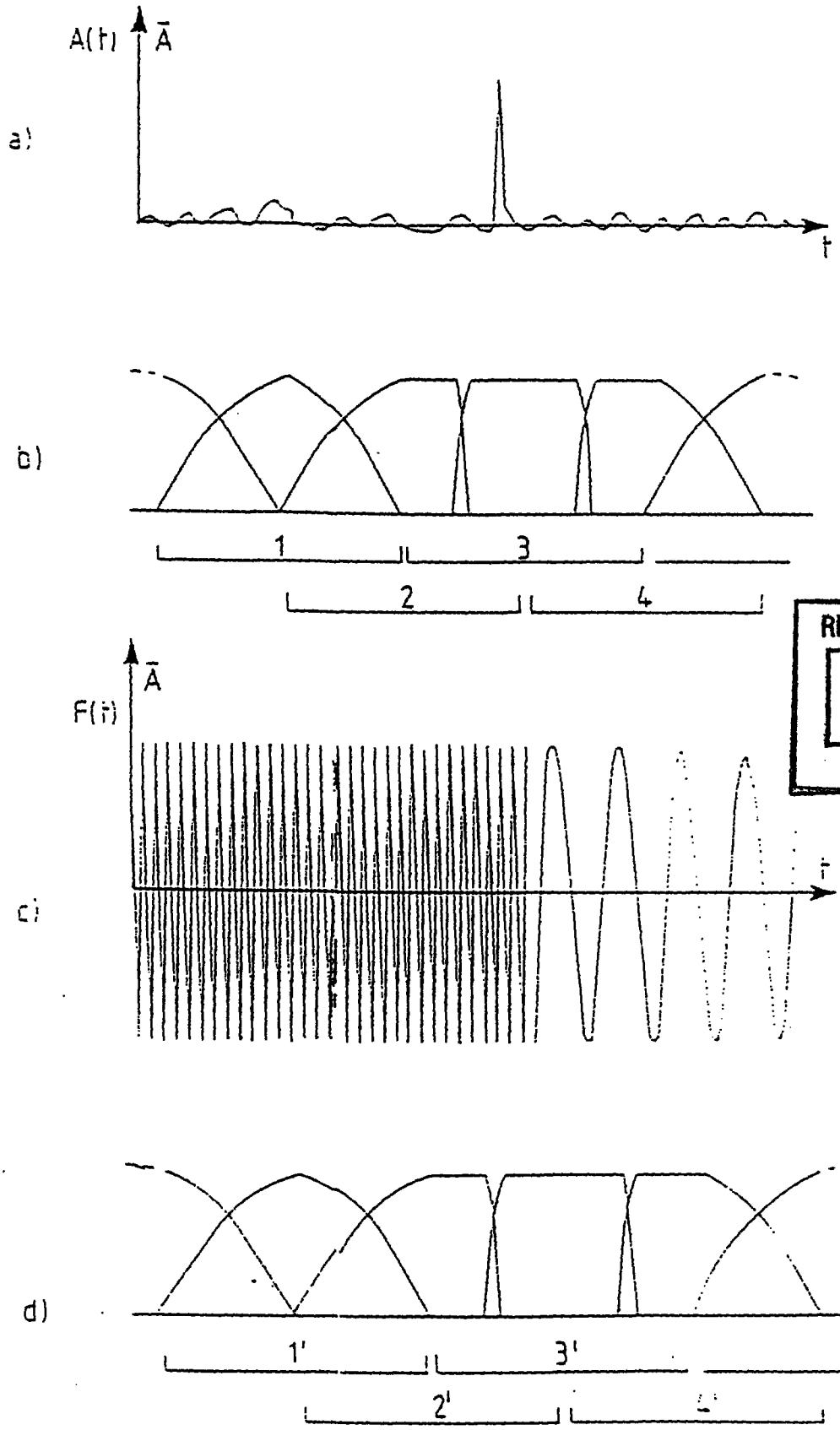

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FIG.3

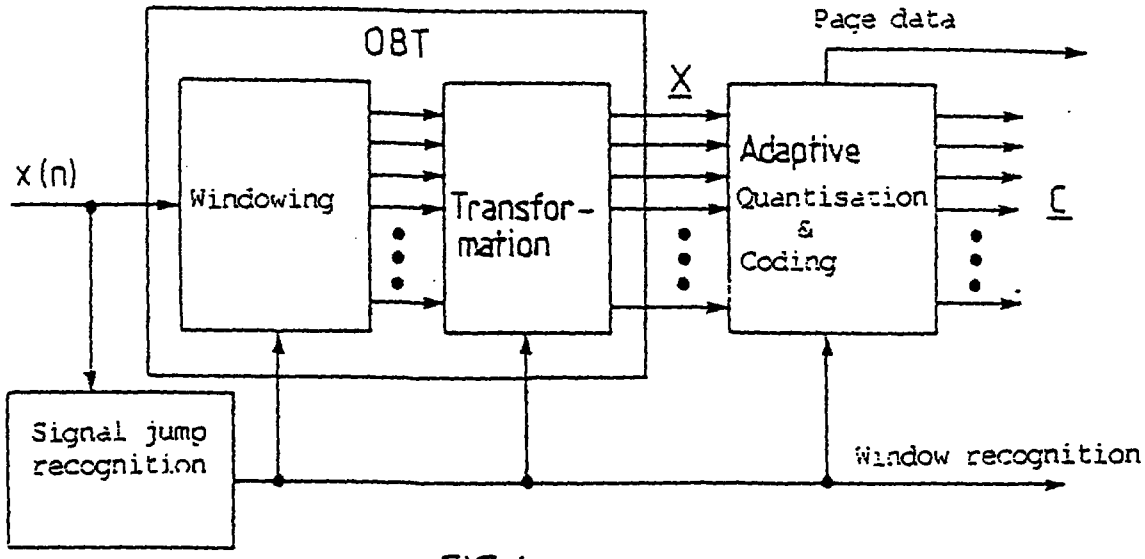

FIG.4a

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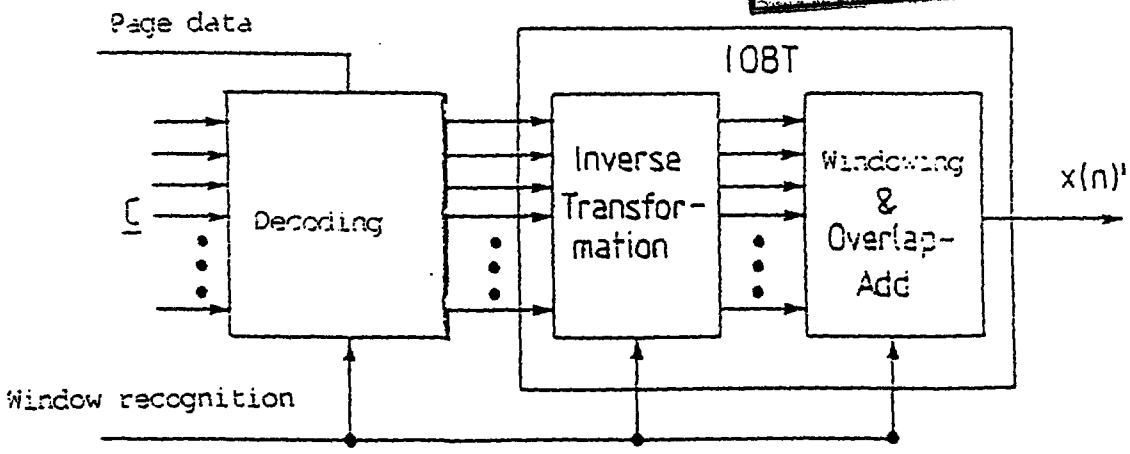

FIG.4b

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TRANSFORM CODER WITH ADAPTIVE WINDOW FUNCTION

ABSTRACT

In a method for transmitting a signal which is partitioned into successive, overlapping blocks, the partial signals contained in the blocks are always converted into a spectrum by means of transformation, and following that, the spectrums are coded, transmitted, and after transmission decoded and reconverted back into partial signals using re-transformation. The blocks containing the partial signals are finally put together with overlaps, whereby the overlapping areas of the blocks are weighted in such a way that the resultant of the window functions equals one.

In order to avoid interference with neighboring blocks during frequency changes, the window functions and the length of the window functions are selected depending on the frequency changes of the input signal.

The method is suitable for the treatment of audio and video signals which are subjected to a data reduction during transmission.

Figure 4a