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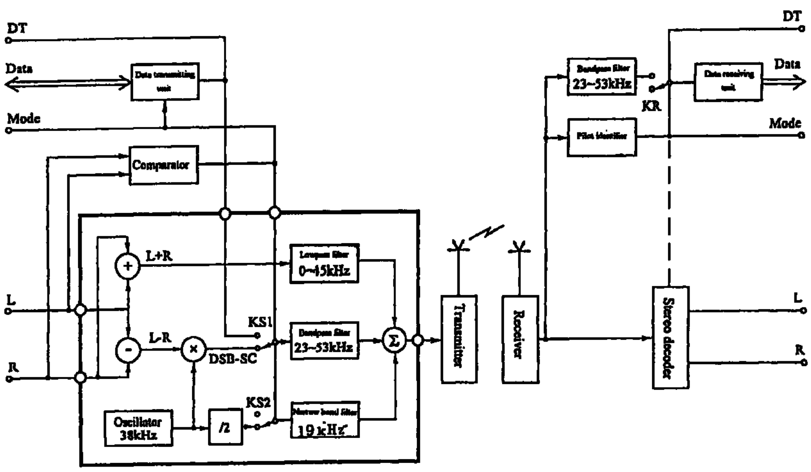
(54) A SYSTEM OF FM DATA BROADCASTING AND A METHOD OF PROCESSING DATA SIGNALS THEREOF

(57) An FM data broadcasting system comprising a transmitting section and receiving section, input terminal, a stereo encoder and out terminals, the system is characterized in that the transmitting section further comprises a data transmitting unit, the first converting switch, the second converting switch, a mode controlling terminal; the receiving section further comprises a data-unit, a band-pass filter, a pilot identifier, the third converting switch, the system is compatible with the conventional FM broadcasting system and is effective to transmit data without any effectiveness on the reception of audio signals.



Principle of an FM "stereo" / "monophony +data" broadcast
Fig3

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Description

Field of the Invention

[0001] The present invention relates to a method for data signals processing during the data broadcast in a conventional FM stereo broadcast system, and a solution for its compatibility. The method can be also used for other data transmission system.

Background of the Invention

[0002] Language (voice) and music are two main compositions of voice broadcasting. For a stereo broadcast with double channels, the stereo effect of language programs does not have practical meaning. In practice, there is only one signal source of voice program in a conventional stereo broadcast. Before the voice signal is transmitted, it is divided into a left signal and a right signal at a branch point X in Fig. 1, then transferred by using two channels respectively, until reaching the left and right ears of a listener. Obviously, if the voice signal is divided into a left signal and a right signal at a branch point Y in Fig. 2 after it reaches the receiver, and supply to two ears of a listener; the practical auditory effect is the same as the former. However, the transmission channel and transmission capacity can be saved up for transmitting data signals and conducting a data broadcast. This broadcast mode of "monophony + data" is suitable for using during the broadcast of all monophonic programs of a stereo broadcast station.

Object of the Invention

[0003] An object of the present invention is to make the conventional FM stereo broadcast system possesses both a double channels "stereo" broadcast mode and a "monophony + data" broadcast mode. These two modes are compatible with each other and can be converted flexibly to realize the data broadcast in a FM stereo broadcast.

[0004] Another object of the present invention is to encode the data with a suitable variable-width code, so that a conventional FM stereo broadcast system possesses the capability of the FM "stereo + low speed data" / "monophony + high speed data" , thus dynamically using the transmission capacity of a FM broadcast.

Summary of the Invention

[0005] The FM L - R data broadcasting system of the present invention includes a sending section and a receiving section. Said sending section includes an input terminal with both left and right audio channels L and R, a stereo encoder, and an output terminal. Said receiving section includes a receiver, a stereo decoder, and an output terminal. Said sending section further includes a data transmitting unit for providing the data signals to be transferred; a first switch KS1 provided among a DSB-SC modulator, a data transmitting unit and a bandpass filter and used to selectively receive data signals or audio signals; a second switch KS2 provided between a divider and a narrow-band filter and used to selectively put through or cut off pilot signals; and a mode controlling terminal connected to the first and second switches KS1 and KS2 for controlling the state of switches, and connected to the data transmitting unit for controlling the data signal transmission of the data transmitting unit. Said receiving section further includes a data receiving unit for receiving the transmitted data signals DT; a bandpass filter, the input of which is connected to said receiver, for separating the data signals DT from the broadcast baseband signals; a pilot identifier, the input of which is also connected to said receiver, for identifying pilot signals and outputting the mode controlling signal at its output; and a third switch KR provided between said bandpass filter and said data receiving unit and connected to the output of the pilot identifier; wherein through using the third switch, the data signals DT is selectively supplied into the data decoder by the mode controlling signal outputted by said pilot identifier, and is recovered into the transmitted data stream.

[0006] A method for FM L-R data broadcasting, said method comprises the steps of: a) setting mode controlling signal mode at the mode controlling terminal in the sending section; b) when conducting a stereo broadcast, the mode controlling signal mode of said sending section instructing the data transmitting unit to stop operation, and instructing the stereo encoder to operate normally to receive FM broadcast baseband signals; c) when a FM L-R data broadcast is in progress, the mode controlling signal mode of said sending section initiating said data transmitting unit, disconnecting said first switch KS1 from the output of DSB-SC modulator DSB-SC and connecting the data transmitting unit for data transmission, disconnecting said second switch KS2 from the output of the divider to cut off the pilot signals; d) placing a second bandpass filter at the receiving section to receive the transmitted broadcast baseband signals comprising data signal ; e) placing a pilot identifier at the receiving section for identifying pilot signals from the baseband signals; f) placing a third switch KR between the bandpass filter and the data receiving unit, said third switch KR is connected to the output of said pilot identifier; g) when said pilot identifier identifies pilot signals from the broadcast baseband signals, the mode controlling signal mode of said output terminal disconnecting said third switch KR from the bandpass filter, stopping receiving the data signals, and instructing the stereo decoder to operate normally to receive stereo broadcast signals; h) when no pilot signal is identified from said input signals by said pilot identifier, the mode controlling signal mode of the output terminal connecting said third switch KR to said bandpass filter to receive the data signals.

[0007] a FM L-R data broadcasting system, said broadcasting system comprises a sending section and a receiving section, said sending section comprising: an input terminal with a left and a right channel L, R , a stereo encoder and an output terminal; said receiving section comprising a receiver, a stereo decoder and an output terminal, said system characterized in that: the sending section further comprises: a data transmitting unit, the output of which is connected to the adder of said stereo encoder, for providing the data signals to be transmitted; a first switch KS1 , provided among said DSB-SC modulator, said data transmitting unit and said first bandpass filter, for selectively receiving data signals or audio signals; a second switch KS2 , provided between said divider and said narrow-band filter, for selectively putting through or cutting off pilot signals; a mode controlling terminal, connected to the first and second switches KS1, KS2 , for controlling the state of said switches KS1, KS2 , said mode controlling terminal also connected to said data transmitting unit for controlling data transmission of the data transmitting unit; the receiving section further comprises: a data receiving unit for receiving the transmitted data signals DT ; a second bandpass filter, the input of which is connected to the said input terminal, for separating the data signals DT from the broadcast baseband signals; a pilot identifier, the input of which is also connected to said receiver, for identifying pilot signals and outputting the mode controlling signal mode at the output terminal; a third switch KR provided between said second bandpass filter and said data receiving unit, and also connected to the output of the pilot identifier; wherein said mode controlling signal outputted by the pilot identifier selectively supplies the data signal DT to said data decoder through said third switch KR, so as to recover the transmitted data stream.

[0008] A method for data signal transmission using FM broadcasting, said method comprises the steps of: a) setting the mode controlling signal mode at the mode controlling terminal in the sending section; b) when conducting a stereo broadcast, the mode controlling signal mode of said sending section instructing the first switch KS1 to close, transferring DSB-SC signals into the adder so that the data transmitting unit is in the low speed transmitting state (e.g. pattern 4); c) when broadcasting monophonic program, the mode controlling signal mode of said sending section cutting off DSB-SC signals through the first switch KS1 , cutting off pilot signals through the second switch KS2 so that the data transmitting unit is in the high speed data transmitting state (e.g. pattern 6); d) placing a high speed bandpass filter at the receiving section to receive the transmitted low speed data signals when conducting a stereo broadcast; e) placing a low speed bandpass filter at the receiving section to receive the transmitted high speed data signals when conducting a monophonic program broadcast; f) placing a pilot identifier at the receiving section for identifying pilot signals from the baseband signals; g) placing a third switch KR among the high and low speed bandpass filter and the data receiving unit, said third switch KR is connected to said pilot identifier; h) when said pilot identifier identifies pilot signals from the broadcast baseband signals, the mode controlling signal mode of said output terminal connecting the low speed filter to the data receiving unit through said third switch KR), and making the data receiving unit to be in the low speed data receiving state; i) when no pilot signal is identified from the broadcast baseband signals by said pilot identifier, the mode controlling signal mode of the output terminal connecting the high speed filter to the data receiving unit through said third switch KR , and making the data receiving unit to be in the high speed data receiving state.

Brief Description of the Drawings

[0009] The present invention will be described in detail in conjunction with the following accompanying drawings, in which

Figure 1 illustrates a transmission mode of voice signals in a conventional stereo broadcast;
Figure 2 illustrates a FM "monophony + data" broadcast mode;
Figure 3 illustrates the principle of a FM "stereo"/"monophony + data" broadcast;
Figure 4 illustrates the principle of the stereo decoding;
Figure 5 illustrates the decoding principle of a binary variable-width code when Lmax = 1;
Figure 6 illustrates the operational principle of a waveform synthesizer;
Figure 7 illustrates the principle of a FM "stereo + low speed data"/"monophony + high speed data" broadcast; and
Figure 8 illustrates the signal waveform and frequency spectrum of a variable-width code.

Preferred Embodiments of the Invention

[0010] There are two kinds of possible ways for the "monophony + data" broadcast in a FM stereo broadcast:

1) Data signals are transferred directly on one channel of a stereo system, and voice signals are transferred on another audio channel. This kind of way has the following advantages: 1) the conventional stereo broadcast and recording devices can be used to synchronously transmit and record the voice and data signals; and 2) the data signals can be switched between different stereo broadcast systems (such as FM, amplitude modulation or wire broadcasting) and record - playback devices without need of modification. However, it has the following disadvantages: 1) a compatible circuit must be added into the conventional FM radios and stereo record - playback devices to automatically erase cross-interference caused by the data signals on voice; and 2) the utility rate of the transmission channel is low.

2) The voice signals are transferred by a "left + right" channel (i.e. L + R channel) with 0 ~ 15 KHz, the data signals are transferred by a "left - right" channel (i.e. L - R channel) with the frequency above 19 KHz, such as 23 ~ 53 KHz, and the stereo pilot signals of 19 KHz are cut off and used as marks for data broadcast. This kind of broadcast way is compatible completely with the conventional FM radios, and has the following features: the data transmission rate is high, and the interference-free performance of the data signals is quite good. This kind of broadcast node is called "FM L - R data broadcast".

[0011] Figure 3 illustrates the operational principle of the compatible "L - R data broadcast" in a FM stereo broadcast system according to the present invention. It is necessary to add a data processor 1, a control signal (mode) for broadcast mode selection of the "stereo"/ "monophony + data" broadcast in the sending section, and a pair of mode switches (KS1, KS2) are added in the conventional stereo coding circuit. In the receiving section, data receiving unit consists of a bandpass filter with 23 ~ 53 KHz, a pilot identifier, a mode switch (KR) and a data processor 2.

[0012] When conducting a stereo broadcast, the mode signal of the sending section stops the operation of data processor 1 and instructs the stereo encoder to operate normally: the sum signal (L + R signal) obtained by adding the input signals (L, R) of both left and right channel voice is transferred by the L + R channel; their difference signal (L - R signal) is transferred by the L - R channel after the suppressed carrier double side band (DSB-SC) amplitude modulation of the subcarrier signal of 38 KHz; stereo pilot signals are obtained from the subcarrier signal after its frequency is dichotomized; and normal FM broadcast baseband signals are obtained by superimposing these three signals in the adder (Σ), if necessary, further superimposing the RDS and SCA signals. The broadcast baseband signals are transmitted by a transmitter, and recovered by the discriminator of a receiver. In the receiving section, the pilot identifier is set by the pilot signals in the broadcast baseband signals. The output (mode) of the pilot identifier cuts off the input of the data processor 2 through the switch KR, and stops the operation of the data processor 2. The pilot signals in the broadcast baseband signals makes the stereo decoder operate normally. Figure 4 shows the operational principle of the stereo decoder: the narrow-band filter with 19 KHz separates pilot signals from the baseband signal; subcarrier signals with 38 KHz are recovered by duplicating the frequency of pilot signals; through multiplying (demodulating)the subcarrier to the DSB-SC signals in 23 ~ 53 KHz, and passing the multiplied signals through the lowpass filter with 0 ~ 15 KHz, the L - R signal is recovered; and finally, the output signals (L, R) are obtained by adding and subtracting the L - R signal and the L + R signal in the L + R channel.

[0013] When conducting a FM L - R data broadcast, the voice signals (L + R signal) are still transferred by the L + R channel. Under the control of the mode signal at the sending section, the data processor 1 converts the data stream (data) into the data signals (DT); the KS1 switch cuts off the DSB-SC signal in the stereo decoder, so that the DT signals are supplied into the adder (Σ) through the bandpass filter with 23 ~ 53 KHz; and the KS2 switch cuts off the 19 KHz pilot signals. At the same time, the broadcast baseband signals contain the audio signals of 0 ~ 15 KHz and the data signals of 23 ~ 53 KHz without any stereo pilot signals. At the receiving section, the bandpass filter of 23 ~ 53 KHz separates the data signals DT from the broadcast baseband signals. Because no pilot signal occurs in the broadcast baseband signals, the pilot identifier is set to zero. The output (mode) of the pilot identifier supplies the data signals DT into tile data processor 2 through the KR switch, the supplied data stream (data) is obtained by recovering it. At this time, in the stereo decoder, because no pilot signal of 19 KHz occurs in the broadcast baseband signals, the subcarrier signal of 38 KHz can not be obtained by duplicating frequency method. Then, after passing through the multiplier (X), the data signal still keeps its original signal form of 23 ~ 53 KHz, this signal will be filtered out by the next audio filter circuit. At this time, the L - R signal becomes zero; through adding and subtracting it with the L + R signal, the outputs of two channels of the stereo decoder become the voice signals in the L + R channel. That is to say, at this time the L + R signal is divided into a left signal and a right signal in the stereo decoder.

[0014] Therefore, when conducting a FM L - R data broadcast, using an ordinary radio, whether stereo or monophony, two ears of listener can only hear the voice signal in the L + R channel, and can not hear the data signals in the L - R channel. Once the pilot signals appear in the FM broadcasting, the stereo decoder returns back to the ordinary stereo decoding, and the data processor 2 stops the data demodulation. This is the principle that the "FM L - R data broadcasting" is mutually compatible with the conventional "FM stereo broadcast". The "pilot indication" signal outputted from the stereo decoder can be also used as the mode controlling signal (mode) at the receiving section.

[0015] The broadcasting mode can be set by the mode input apparatus at the sending section, and can be also controlled automatically by a signal comparator. The principle is based on the comparison of the voice input signals of both L and R channels. When the input of both L and R channels are the same ($L - R \approx 0$), a necessary decision procedure is started to determine the selection of broadcast mode.

[0016] When conducting a broadcast retransmitting, the audio output (L, R) of the receiver of the retransmitting station is connected directly to the audio input (L, R) of the transmitter, and the mode output of the receiver is connected directly to mode input of the transmitter. When conducting a data broadcast retransmitting, the retransmitting station connects directly the DT output of receiver to the DT input of the transmitter, without need of the data processor 1 and data processor 2. The retransmitting station can also modify the contents of the data broadcast, at this time, a data processor is needed between the data output of the receiver and the data input of the transmitter in order to modify the contents of data stream.

[0017] When conducting a stereo broadcast, a "lost" phenomenon of the pilot signals may occur due to an interference; at this time, the DSB-SC signal in the L - R channel may be misunderstood as a data signal. Therefore, the transferred data should have a certain error-detecting capability. Once the data processor 2 detects the error data, the data will be abandoned.

[0018] The key to realizing the FM L - R data broadcast is: 1) the effective frequency spectrum of the data signals should be all concentrated on a frequency range of 23 ~ 53 KHz; 2) the cross-interference of the data signals to the adjacent channel must meet the requirements of the broadcast standard, particularly, the cross-interference to the voice signal must be less than -60 dB, and should not excite the frequency-duplicated circuit in the stereo decoder. The effective frequency spectrum means a necessary frequency component to recover a data signal with a certain interference-free performance, and the frequency band of the effective frequency spectrum is called an effective frequency band.

[0019] The present invention uses a code, which transfers data information by discrete values of symbol width, this code is called a variable-width code. The waveform of the variable-width code is bipolar non-return to zero pulse signal, each pulse of which is a symbol, different pulse width represents different information, the other geometric parameters of pulses such as polarity, amplitude, pulse edge, etc. do not carry any information. The symbol width of the variable-width code may have two or more kinds of discrete values to form two or multiple element variable-width codes. The present invention classifies the symbol of a variable-width code into two classes, wherein a symbol with the shortest code width is called S symbol, a symbol string with consecutive S symbols is called a consecutive S symbol string, the TS value represents a symbol period of S symbol; the other variable-width symbols are all called L symbol, a code string with consecutive L symbols is called a consecutive L symbol string, TL value represents a symbol period of L symbol with the longest pulse width. The L symbol in the multiple element variable-width code has more than one symbol periods. Because the variable-width code has different symbol periods, the reciprocal of the symbol period of S symbol is called a symbol rate of a variable-width code, its value $B = 1/TS$; the symbol period ratio of the broadest L symbol to the S symbol is called a variable-width code pulse width ratio K ($K = TL : TS$). The symbol rate (B) and pulse width ratio (K) are hard parameters effecting the effective frequency band of a variable-width code.

[0020] The variable-width code frequency spectrum is characterized in that when the variable-width code pulse width ratio $K \leq 3$, its effective frequency band is distributed at two sides of 0.5 B point; when the consecutive S symbol string in data stream becomes long, its effective frequency spectrum is closed to 0.5 B point; and when the consecutive L symbol string in data stream becomes long, its effective frequency spectrum is diverged from the 0.5 B point to its two sides. Changing the pulse width ratio K value of the variable-width code can also change its effective frequency band.

[0021] It can be seen from the above that: 1) if the length of consecutive L symbols of a variable-width code is controlled in a certain range to make it not greater than Lmax (Lmax is called the maximum consecutive code number of L symbol), then its effective frequency spectrum can be controlled in a certain range of frequency band, and reducing the Lmax value can narrow down the effective frequency band; 2) if the length of consecutive S symbol string of a variable-width code is controlled in a certain range to make it not less than Smin (Smin is called the minimum consecutive code number of S symbol), then increasing the Smin value can also narrow down the effective frequency band of variable-width code. Therefore, Lmax and Lmin are the soft parameters effecting the effective frequency band of a variable-width code.

[0022] The lower limit (Fdn) and upper limit (Fup) of effective frequency band of two-element variable-width code can be expressed respectively as:

$$\mathrm{Fdn} = \frac{\mathrm{Smin} + \mathrm{Lmax} + 1}{2 \times [(\mathrm{Lmax} + 1) \times K + \mathrm{Smin}]} \times B \qquad (1.1)$$

$$\mathrm{Fup} = \frac{\mathrm{Smin} + 3 \times (\mathrm{Lmax} + 1)}{2 \times (\mathrm{Lmax} + 1) \times K + \mathrm{Smin}]} \times B \qquad (1.2)$$

When the channel band width is in or above the range between Fdn and Fup, the pulse width between the transferred S symbol and L symbol and the geometric shape have adequate difference in features to identify and distinguish the waveform. When the channel frequency band is below the range between Fdn and Fup, this difference is reduced rapidly, and the interference-free performance of a variable-width code signal is also reduced rapidly. The following formulae can be obtained from formulae (1.1) and (1.2):

$$B = \frac{2 \times [(Lmax + 1) \times K + Smin]}{Smin + Lmax + 1} \times Fdn \tag{2.1}$$

$$B = \frac{2 \times [(Lmax + 1) \times K + Smin]}{Smin + 3 \times (Lmax + 1)} \times Fup$$

$$\frac{Fdn}{Fup} = \frac{Smin + Lmax + 1}{Smin + 3 \times (Lmax + 1)} \tag{3.0}$$

[0023] It can be seen from these formulae that parameters Lmax, Smin and K are main factors to decide the variable-width code frequency spectrum, interference-free performance and data transfer rate. The code type of a variable-width code consists of the following three part: 1) beginning with an alphabet L, the next number indicates the Lmax value, if $Lmax = \infty$, then it is indicated as LX; 2) beginning with a alphabet K, the next number (including a decimal part) indicates the K value. For example, "LXS1K2.5" code indicates a variable-width code with $Lmax = \infty$, Smin = 1, and K = 2.5; "L1S2" code indicates a variable-width code with Lmax = 1, Smin = 2, and K value is not defined.

[0024] It can be seen from formula (3.0) that for "LXS1" (i.e. $Lmax = \infty$, Smin = 1), the Fup : Fdn = 3, and the signal effective frequency band width $(Fup - Fdn) = 2 \times Fdn$. Therefore, when the channel band width is greater than or equal to 2 x Fdn, and the binary coded information is transferred, each "1" character in the binary coded information can be converted directly into a L symbol, and each "0" character can be converted into a S symbol; or each "0" character converted into a L symbol, and each "1" character converted into a S symbol. This converting is called the "direct coding".

[0025] When the channel band width is less than 2 x Fdn, the effective frequency band of a variable-width code is compressed by a method limiting the Lmax and making Smin = 1. For example, when using a "L1S1" variable-width code, Fdn : Fup = 3/7, and the required channel band width is 4/3 x Fdn. When the channel band width is less than 4/3 x Fdn, Smin value can be increased, thus further compressing the effective frequency band of a variable-width code signal.

[0026] When $Lmax \neq \infty$, the coding principle to transfer data information with the variable-width code is that: grouping each section by string consisting of the same type of symbols, a symbol type identifier is transmitted before each information "group" to identify the symbol type of the information "group", then the information to indicate the length of this group is transferred. The symbol type identifier is a specific code string which begins with a L symbol and consists of several L symbols, and if necessary, some appropriate S symbols. After the type identifier, each S symbol represents a symbol defined by this type identifier, until the next identifier appears.

[0027] When $Lmax \neq \infty$, the decoding principle of the variable-width code is that: the type of the symbol to be arrived is determined by identifying the symbol type identifier in the data stream, then the each successive S symbol is converted into a determined symbol, until the next type identifier appears.

[0028] When an information stream consisting of N kinds of symbols is transferred by the two-element variable-width code, if the symbol type is indicated simply by the length of consecutive L symbol string, then the Lmax of the two-element variable-width code = N; if the symbol type is indicated by the different arrangement forms of L symbol and S symbol, or in conjunction with the appropriate transmission protocol, $Lmax < N$ is possible. Therefore, after N element data information is converted into the two-element variable-width code, $Lmax \leq N$ thus reaching the object to control the effective frequency band of data signals.

[0029] Different expression forms of the type identifier constitute the different code format of a variable-width code. The object to select the different code format is to change the code type soft parameters Lmax and Smin, thus changing the effective frequency band of a variable-width code.

[0030] When the two-element data information consisting of character "0" and "1" is transferred by the two-element variable-width code, there are both A and B basic coding formats (i.e. Smin = 1):

[0031] The A basic coding format : let Lmax = 2, then the variable-width code has two kinds of consecutive L symbol strings with different length, they can be used as the type identifiers for characters "0" and "1" respectively.

[0032] For example, the character type "0" is indicated by the single L symbol, and the character type "1" is indicated by the consecutive two L symbols. At this time, the coding procedure of the A format is that: When a consecutive "0" character string (including the single character "0") appears in the data stream, the encoder outputs firstly a L symbol, then converts every character "0" in this consecutive "0" character string into a S symbol; when a consecutive "1" character string (including the single character "1") appears in the data stream, the encoder outputs firstly two L symbols, then converts every character "1" in this consecutive "1" character string into a S symbol. The decoding procedure of the A format is that: when the single L symbol appears in the variable-width code stream, every S symbol following this L symbol is converted into a character "0", until the next L symbol appears; when the consecutive two L symbols appear in the variable-width code stream, every S symbol following the two L symbols is converted into a character "1", until the next L symbol appears;

[0033] Of course, the character type "1" can be indicated also by the single L symbol, and the character type "0" can be indicated by the consecutive two L symbols.

[0034] The B basic coding format: in order to compress the effective frequency band, let Lmax = 1. At this time, the variable-width code has only one form of consecutive L symbol string (i.e. a single L symbol). Using this consecutive L symbol string can not indicate the concrete type of symbol, but can indicate the phenomenon of "the symbol type has been changed". For example, when a L symbol appears, if a character "0" is transferred before this, then a character "1" must be transferred after this; and if a character "1" is transferred before this, then a character "0" must be transferred after this. Thus, if only the initial state of the decoder in the receiving section can be defined correctly or the operating state of the decoder can be adjusted (set) timely to make it consist with the state of the decoder in the sending section, then the two-element data information can be recovered correctly by the decoder, otherwise the phenomenon of the reversal of "0" and "1" will appear in the decoded two-element data information. This reversal phenomenon of "0" and "1" is called the polarity reversal of two-element data information. Setting the state of decoder is called the polarity synchronization of two-element data information.

[0035] The two-element variable-width code with Lmax = 1 can not define the state of decoder. Therefore, the present invention will further make the two-element data information carry the polarity information of itself, so that the polarity reversal phenomenon can be detected and corrected in the decoding process of the receiving section.

[0036] The principle on the polarity synchronization of the two-element data information is to set a polarity synchronization symbol. If the consecutive "1" character string with a length K is used as a polarity synchronization symbol, i.e. character string "011.......110" (wherein the number of "1" is equal to K), then the character string format to reverse the polarity synchronization symbol, i.e. character string "100......001" (wherein the number of "0" is equal to K) is called the "synchronous inverted character" of the polarity. The following steps are conducted for the two-element data information:

1) "add 1" processing: when the consecutive "1" character string with the length greater than or equal to K appears in the data stream, a character "1" is added in this consecutive "1" character string. The polarity synchronization symbol does not exist in the so processed data stream.
2) "add 0" processing: when the consecutive "0" character string with the length greater than or equal to K appears in the data stream, a character "0" is added in this consecutive "0" character string. The synchronous inverted character does not exist in the so processed data stream.
3) "add synchronization" processing: in the data stream after conducting the "add 1" and "add 0" processing, a suitable number of polarity synchronization symbols are interposed every suitable distance or between the separating data blocks respectively.

[0037] In the two-element data information stream after conducting these three processing steps, once the synchronous inverted character appears, it means that the polarity is reverse. In this case, the correct two-element data information can be obtained by reversing the "0" and "1" in the data.

[0038] It is preferred that the data broadcasting uses the transmission mode of "data packet". At this time, the character of data packet can be used as the polarity synchronization symbol, the character format of reversed character is the synchronous inverted character. Therefore, if the "add 0" processing step is added in the original data packing process, and the character polarity synchronization and "delete 0" processing steps are added in the original data depacketing process, the two-element data information can be transferred by using the two-element variable-width code with Lmax = 1.

[0039] A "packet head symbol" and a "packet tail symbol" should be added respectively in the head part and tail part of data block, in order to avoid that the pseudo character string similar to the character and synchronous inverted character is generated by combining the character with the transferred data. The packet head symbol and packet tail symbol can have a plurality of constitution forms; for example, the first character of packet head symbol and the last character of packet tail symbol can be 1, and the other characters can be used to transfer the additional information such as the data packet length, property, error-detecting and error-correcting, thus providing the user level with multiplex "parallel virtual channels". The packet tail symbol also functions as clearing the register in encoder and decoder. The "add 1" and "add 0" processing should be conducted for the data block together with its packet head symbol and packet tail symbol, then it is connected to the polarity synchronization symbol.

[0040] The encoding process of B format is that the data to be transferred are separated into the data blocks. A packet head symbol and a packet tail symbol are added respectively at the head and tail ends of every data block, then "add 1" and "add 0" processing are conducted to form the data packets. A suitable amount of separator is interposed between the data packet and data packet, and connecting them again. Then, every character (whether it is "0" or "1") in this data stream is converted into a S symbol, and a L symbol is interposed at a place where the character is changed front "0" to "1", or changed from "1" to "0".

[0041] Figure 5 illustrates the principle and process of decoding, polarity synchronizing, and depacketing of B format. The initial state ("0" or "1") of character register in decoder can be set arbitrarily. When a variable-width code stream is supplied to the decoder, if a S symbol is an input, then a character in the character register is outputted from the decoder; if a L symbol is an input, then the character polarity in the character register is reversed for one time, at this time, the decoder does not output any character. The data stream outputted from the decoder passes through a polarity adjuster. This polarity adjuster further has a state control input terminal, its state control signal comes from the output of the state register. The initial state of the state register output can be set arbitrarily. When the state control signal is "0", the data stream still keep its original polarity after passing through the polarity adjuster; when the state control signal is "1", the character polarity of the data stream is reversed after passing through the polarity adjuster. The data outputted from the polarity adjuster are temporarily stored in a data temporary storage device, and a synchronous discriminator consisting of "K + 2" bit shift register. When a synchronous inverted character appears in the synchronous discriminator, its output makes the state in the state register reverse for one time and gives up the data temporarily stored in the data temporary storage device, thus the stack pointer of the data temporary storage device is moved back to the originating point. When a character (i.e. polarity synchronization symbol) appears in the synchronous discriminator, the data in the data temporary storage device must be processed for one time. At this time, if the data in the data temporary storage device is greater than a certain value, then the data in the data temporary storage device is a effective data packet, and it can be supplied to a depacketing device for depacketing processing; otherwise, the data in the data temporary storage device is not effective, thus giving it up. No matter whether the data of the temporary storage device is effective, the stack pointer of the data temporary storage device should be moved back to its originating point after each processing. The "delete 1" and "delete 0" processing are conducted for the data in the depacketing device. The "delete 1" processing is that: when the consecutive "1" character string in which the number is greater than K appears in the data stream, a character "1" is deleted from this character string. The "subtract 0" processing is that: when the consecutive "0" character string in which the number is greater than K appears in the data stream, a character "0" is deleted from this character string. Then, the transferred data block is obtained by cutting off the packet head symbol and packet tail symbol of the data packet. The transferred two-element data information is obtained by connecting the recovered data blocks. In order to complete the character polarity error-correcting synchronization process discussed above, a suitable amount of characters should be interposed between the data packets.

[0042] A Miller code consists of three symbol with different width, the ratio of the code width between them is 2 : 3 : 4, they are called respectively M2 code, M3 code and M4 code. When the Miller code information is transferred by the two-element variable-width code, there are two kinds of basic coding format (i.e. Smin = 1) : C and D.

[0043] The C basic coding format: let Lmax = 3, then the variable-width code has three kinds of consecutive L symbol sirings with different length, they can be used respectively as three type identifiers of symbols of Miller code.

[0044] For example, because the utility factor of M2 symbols is the highest, a single L symbol is used as the type identifier of M2 code; a consecutive L symbol string in which the length is equal to 2 is used as the type identifier of M3 code; and a consecutive L symbol string in which the length is equal to 3 is used as the type identifier of M4 code. In this case, the coding procedure of C format is that: when the consecutive "M2" code string (including a single M2 code) appears in the Miller code stream, the encoder firstly outputs a L symbol, then converts every M2 symbol in this consecutive "M2" code string into a S symbol; when the consecutive "M3" code string (including a single M3 code) appears in the Miller code stream, the encoder first consecutively outputs two L symbols, then converts every M3 symbol in this "M3" code string into a S symbol; and when the consecutive "M4" code string (including a single M4 code) appears in the Miller code stream, the encoder first consecutively outputs three L symbols, then converts every M4 symbol in this consecutive "M4" code string into a S symbol. In this case, the decoding procedure of C format is that: when a single L symbol appear in the variable-width code stream, every S symbol following the L symbol is converted into a M2 code, until the next L symbol appears; when two consecutive L symbols appear in the variable-width code stream, every S symbol following them is converted into a M3 code, until the next L symbol appears; and when three consecutive L symbol appears; and when three consecutive L symbols appear in the variable-width code stream, every S symbol following them is converted into a M4 code, until the next L symbol appears.

[0045] The D basic coding format: let Lmax = 2, so as to compress the effective frequency band of the variable-width code. In this case, three kinds of code strings with different forms can be constituted by two or less L symbols in conjunction with suitable S symbols, these strings can be used respectively as three type identifiers of symbols of Miller code.

[0046] For example, a L symbol added by a S symbol (indicated as "L + S" code string) can be used as the type identifier of M2 code; two consecutive L symbols (indicated as "L + L" code string) can be used as the type identifier of M3 code; and a L symbol added by a S symbol and added by a L symbol (indicated as "L + S + L " code string) can be used as the type identifier of M4 code. In this case, the coding procedure of D format is that: when the consecutive "M2" code string (including a single M2 code) appears in the Miller code stream, the encoder firstly outputs a "L + S" code string, then converts every M2 symbol in this consecutive "M2" code string into a S symbol; when the consecutive "M3" code string (including a single M3 code) appears in the Miller code stream, the encoder firstly outputs a "L + L" code string, then converts every M3 symbol in this consecutive "M3" code string into a S symbol; and when the consecutive "M4" code string (including a single M4 code) appears in the Miller code stream, the encoder firstly outputs a "L + S + L " code string, then converts every M4 symbol in this consecutive "M4" code string into a S symbol. In this case, the decoding procedure of D format is that : when the "L + S" code string appears in the variable-width code stream, the decoder firstly outputs a M2 symbol, then converts every S symbol following it into a M2 code, until the next L symbol appears; when the "L + L" code string appears in the variable-width code stream, the decoder converts every S symbol following it into a M3 code, until the next L symbol appears; and when the "L + S + L " code string appears in the variable-width code stream, the decoder converts every S symbol following it into a M4 code, until the next L symbol appears.

[0047] Because the type identifier is added in the coding procedure, the transferred data volume in enlarged. On the other hand, in the process of transmission of Miller code information, when the symbol period of M2 code is equal to TS and TS = 2Δ , only the 2Δ is spent on the specific transmission time for the M3 code with the 3Δ period and the M4 code with the 4Δ period. That is to say, after encoding by the variable-width code, the data volume of Miller code is compressed, the instant maximum value of this compressibility factor can reach 2 : 1. After combining the enlargement effect with, the compression effect, the code efficiency η of the variable-width code (η is equal to the ratio between the data volumes before and after coding) is a dynamic value, depending on the variable-width code parameters, code format, and data stream structure. If the encoding/decoding procedure of the variable-width code is used as one of the composition of "transmission", then the transmission rate of "user" data is also a dynamic value, and equal to B x η. When the variable-width code pulse width ratio K is equal to 2, the code efficiency of B basic coding format is $0.333 < \eta < 1$, and the statistical average value is about 0.711; and the code efficiency of D basic coding format is $0.283 < \eta < 2$, and the statistical average value is about 0.730.

[0048] Based on the basic coding format of variable-width code, let Smin > 1, the effective frequency band of variable-width code can be further compressed, or the interference-free performance can be improved. In this case, the code efficiency will be reduced when Smin value is increased. The coding principle of the variable-width code when Smin > 1 is that: the S symbols with the number = (Smin - 1) are firstly added next to every type identifier, then the symbol is transferred. In this case, the decoding principle of variable-width code is that : the S symbols following every type identifier with the number = (Smin - 1) are jumped, then the successive S symbols are converted into the symbols defined by this type identifier, until the next L symbol appears.

[0049] The very rich low and high frequency harmonic wave components are still contained in the encoded variable-width code signals, they can not be supplied directly to the transmission channel, must be very strictly filtered to meet the requirements of the technical standard for broadcast. For example, the cross-interference to the audio channel must be less than -60 dB, and can not excite the subcarrier reset circuit in the stereo decoder. It is difficult to obtain such a filter effect using a hardware circuit or a data filter.

[0050] The waveform synthesizer used in the present invention is a "code/pulse" converter, it determines the shape of the output pulse on the basis of the code form of the input variable-width code stream. These pulse shapes are a group of previously optimized modular waveforms. The object of the optimized processing is that the frequency spectrum distribution of the signal made up by these modular waveform can meet the special requirements.

[0051] Figure 6 is a block diagram illustrating the principle of waveform synthesis. The symbol window is a shift register of the symbol of the variable-width code with a suitable length. All the code string formats of the variable-width codes which may appear in the symbol window, so-called "code string", are found out previously, and these code strings are stored in the "code" area of the "module library". Then these code strings are previously replaced one by one by some original waveform such as an amplitude-modulated rectangular wave in which each pulse has the same area; or the rising edge of the pulse is further changed from -90° to +90°, and the back edge of the pulse is further changed from +90° to -90° to be a variable-amplitude plane-top sine curve, then an ideal waveform after the ideal filtering is found out. Thus, each code string corresponds to an ideal waveform, the pulse provided at the center of an ideal waveform is a waveform module, it is an identification number of the symbol provided at the center of the code string. The quantized value of each waveform module can be used as a data group and stored in the "module" area of "module library". The "code" is associated one to one with its correspondent "module" through this "module library". This is the previous optimized processing. When supplying the data, the data stream of variable-width code is shifted to this symbol window. Once every shifting, the code string which appears in the symbol window is used as a retrieval base, the correspondent code string is found out in the code area of module library, a correspondent waveform module is found out through this code sting, then this waveform module (a group of data value) is supplied to a D/A converter. Under the driving of a sample clock, and based on a principle on which the positive and negative polarities appear alternatively, this group of data value is converted into a pulse waveform through the D/A converter. After an identification number waveform is generated, one bit of the data stream is shifted in the symbol window, then synthesis of next identification number waveform is started. This waveform synthesis procedure is completed by a computer in conjunction with the D/A converter.

[0052] After the data signal is amplitude-limited amplified, its pulse width is identified and the variable-width code can be recovered. When a passing-zero-point identification method is used directly, the interference-free performance of the system is not high, because after the harmonic component of the variable-width code signal is filtered off, the waveform produces a distortion. The symbol identification equipment used in the present invention integrates the variable-width code signal after conducting an amplitude-limit amplification, then the time period tx of passing-zero-point of this integration value is found out. When tx is greater than the period threshold tm, it is identified as a L symbol; otherwise, it is identified as a S symbol. The symbol identification equipment can be kept at the optimized operational state through adjusting the period threshold tm and the integral time constant tr, it can identify a seriously distorted variable-width code, thus greatly improving the interference-free performance of data receiver.

[0053] A plurality of data broadcast modes can be flexibly constituted by using the variable-width code in the FM broadcasting, for example:

Mode 1: if a "L1S1" code type is used in conjunction with the B coding format, then the variable-width code signal with an effective frequency band of 22.714 ~ 53 KHz is basically consistent with the conventional L - R channel. In this case, if let K = 2 or 2.5, then the data rate is 53.8 Kbps or 64.6 Kbps respectively.

Mode 2: if a "L1S25" code type is used in conjunction with the B coding format, then the variable-width code signal with a effective frequency band of 53.069 ~ 60.931 KHz is consistent with the conventional RDS channel (57 ± 4 KHz). In this case, if let K = 2 or 2.5, then the data rate is about 18.2 Kbps or 18.5 Kbps respectively. When suitably increasing the Smin value and decreasing the data rate, the interference-free performance of data signals can be improved and the isolated degree of the RDS channel from its adjacent channels can be increased. Compared with using the ASK or PSK manner, the data broadcast conducted by using tile variable-width code technology in the RDS channel has the following advantages: high data rate, simple receiving circuit, and reliable system.

Mode 3: at present, there is not any standard for the frequency band upper-limit of 61 KHz or above frequency band (called SCA auxiliary communication channel) in the FM broadcast baseband signals and for the SCA channel usage. If a "L1S16" code type is used in conjunction with the B coding format, then the data broadcast can be conducted by using the variable-width code signal with a effective frequency band of 61.2 ~ 74.8 KHz in the SCA channel of 61 ~ 75 KHz. In this case, if let K = 2 or 2.5, then the data rate is 30.5 Kbps or 31.3 Kbps respectively.

Mode 4: The RDS channel and SCA channel are merged as a data channel with a band width of 53 ~ 75 KHz. The data broadcast can be conducted by using the "L1S8" code type and B coding format. In this case, if let K = 2 or 2.5, then the data rate is 42.2 Kbps or 44.0 Kbps respectively.

Although the code efficiency of the B basic coding format is higher than that of the direct coding format (about 36.3%), it is a related coding. When some symbol is interfered to produce an error code, this error code may effect the successive symbols. This error code cross-interference will not be over the data packet to effect the next data packet. Compared with it, the error code in the direct coding format will not effect the other symbols.

Mode 5: if the data channel is extended through 20 ~ 60 KHz (i.e. the L - R channel plus the RDS channel, the SCA channel is not effected), or is extended through 23 ~ 69 KHz or 23 ~ 75 KHz, or uses the higher frequency upper limit, then the data broadcast can be conducted by using the "LXS1" type variable-width code and direct coding format. In accordance with the probabilities of the "0" and "1" characters in the information stream being equal and K = 2, the broadcasted data rate and the signal interference-free performance can be computed: the rate can reach respectively 53.3 Kbps, 61.7 Kbps, 66.7Kbps or more, and the performance can reach 23 to 16 dB.

Mode 6: The four-element variable-width code with the "L1S1" pulse width ratio of 1 : 2.0 : 2.5 : 3.0 is used, for example, the S symbol pulse width TS = 2Δ , the three L symbol pulse widths are respectively 4Δ, 5Δ and 6Δ, they are called respectively T4, T5 and T6 code; the effective frequency band of this variable-width code is just 22.928 ~ 75 KHz. After the binary information is converted into the Miller code, the four-element direct coding format can be used, for example, a single T4 symbol can be used as the symbol type identifier of M2 code, a single T5 symbol used as the symbol type identifier of M3 code, a single T6 symbol used as the symbol type identifier of M4 code, and every successive S symbol can indicate a defined Miller symbol. In this case, the symbol rate B is 107.142 KHz, and the data rate is much greater than that of the application modes discussed above.

[0054] The data broadcast mode 1, mode 2, mode 3 and mode 4 discussed above all meet the conventional FM broadcast standard. The data broadcast mode 5 and broadcast mode 6 have the features of high data rate, good interference-free performance and stable system (small error code cross-interference). The data broadcast of mode 1, mode 5 and mode 6 will be interrupted by broadcasting the stereo program. A plurality of data broadcast modes can be sampled simultaneous by a FM broadcast station, the correspondent data receiving unit is provided in the data receiver, the received data stream is grouped and then supplied to the computer. When the effective frequency band of one of the data broadcast modes is not compatible with the stereo signal, the data transmitting and receiving of the mode is controlled by the mode controlling signal (mode).

[0055] As shown in Figure 7, when a FM station broadcasts a stereo program, the mode controlling signal (mode) at the sending section supplies the DSB-SC signal to the adder through the KS1 switch, and supplies the pilot signals to the adder through the KS2 switch; at the same time, it makes the data transmitting unit be at the low speed data transmitting state, just like the mode 4. In this case, the mode controlling signal (mode) outputted from the pilot identifier at the receiving section instructs the a low speed bandpass filter to connect the data receiving unit through the KR switch, and makes the data receiving unit be at the low speed decoding operational state. When a FM station broadcasts a program, the mode controlling signal (mode) at the sending section cuts off the DSB-SC signal through the KS1 switch, and cuts off the pilot signals through the KS2 switch; at the same time, it makes the data transmitting unit be at the high speed data transmitting state, just like the mode 6. In this case, the mode controlling signal (mode) at the receiving section instructs the high speed bandpass filter to connect the data receiving unit, and makes the data receiving unit be at the high speed decoding operational state.

[0056] Tile variable-width code and waveform synthesis technology can be used also for the other frequency bands of broadcasting, and other data communication channels.

[0057] The upper figure of Fig 8 illustrates a signal wave of the two-element variable-width code, with the symbol window width = 15, Lmax = 1, Smin = 1, pulse width ratio = 2; and the lower figure illustrates the frequency spectrum of the variable-width code.

Claims

1. An apparatus for sending data on L-R channels in a FM broadcasting system, said FM L-R channels comprising a subtracter for receiving input signals, a DSB-SC modulator, a bandpass filter and a adder, characterized in that said apparatus further comprises:

   a data transmitting unit, the input of which is connected to the data interface (data), for providing data signals to be transmitted;
   a first switch (KS1), provided among said DSB-SC modulator, said data transmitting unit and said bandpass filter, for selectively receiving data signals or voice signals;
   a second switch (KS2), provided between said divider and said narrow-band filter, for selectively putting through or cutting off pilot signals;
   a mode controlling terminal, connected to the first and second switches (KS1, KS2), for controlling the state of the switches, said mode controlling terminal also connected to the data transmitting unit for controlling data signal transmission of the data transmitting unit.

2. An apparatus according to claim 1, further comprises a comparator, the input of which is connected to the left and right channels respectively, for comparing the left and right channels, the output of which is connected to the mode controlling terminal, for automatically controlling the operations of the switches (KS1, KS2) and the data transmitting unit.

3. An apparatus according to claim 1, wherein said mode controlling terminal may also manually control the operation of said first and second switches (KS1, KS2) and the data transmitting unit.

4. An apparatus according to claim 1, wherein said data transmitting unit further comprises a data encoder for encoding the input data, and providing the encoded data to the data transmitting unit.

5. An apparatus according to claim 1, further comprises a data signal terminal (DT) connected to said first switch (KS1) for directly retransmitting data signals of the data signal terminal (DT) of the receiving section when retransmitting.

6. An apparatus according to claim 1, wherein a data processor is provided between the data interface (data) of the receiving section and the data interface (data) of the sending section for modifying of the contents of the data stream when retransmitting.

7. An apparatus according to claim 1, wherein said mode controlling terminal (mode) may be directly connected to the mode controlling terminal (mode) of the receiving section for retransmitting the mode controlling signal when retransmitting.

8. An apparatus according to claim 4, wherein said data encoder uses a variable-width code to concentrate all effective spectrum of data signals in the desired frequency band, said variable-width code characterized in that data information is transmitted in the form of discrete values of the width of the symbol, and said variable-width code is a bipolar non-return to zero pulse signal with each pulse of which being a symbol, different width of pulse representing different information.

9. An apparatus according to claim 4, wherein said data transmitting unit further comprises a waveform synthesizer for performing wave synthesis on the encoded data outputted by the data encoder, so that the cross-interference to the adjacent channels caused by the data signals meets the requirement of broadcast standard.

10. An apparatus according to claim 9, wherein said waveform synthesizer is a "code/pulse" transformer for determining the shape of output pulses based on the form of code of the input data stream.

11. An apparatus according to claim 10, wherein the shape of said pulse has been previously optimized.

12. An apparatus for receiving data broadcasted on the L-R channels in a FM broadcasting system, said apparatus comprising an input terminal, a stereo decoder and an output terminal, characterized in that said apparatus further comprising:

a data receiving unit for receiving the transmitted data signal (DT);
a bandpass filter, the input of which is connected to the said receiving section, for separating data signals (DT) from the broadcast baseband signals;
a pilot identifier, the input of which is also connected to said receiving section, for identifying pilot signals and outputting mode controlling signal (mode) at the output terminal;
a third switch (KR), provided between said bandpass filter and said data receiving unit, and also connected to the output of said pilot identifier;
wherein said mode controlling signal outputted by the pilot identifier selectively supplies the data signals (DT) to the data receiving unit through the third switch (KR), so as to recover the transmitted data stream.

13. An apparatus according to claim 12, wherein said data receiving unit comprises an amplitude-limited amplifier, a symbol identifier and a data decoder, for decoding the received data signals, which has been encoded by the variable-width code and been amplitude-limited amplified and symbol identified, to generate the data stream.

14. An apparatus according to claim 13, wherein a integrator is provided between said amplitude-limited amplifier and symbol identifier for decreasing tile error rate of the symbol identifier.

15. An apparatus according to claim 14, wherein said integrator may be a data integrator.

16. An apparatus according to claim 12, wherein the data signals (DT) received are directly connected to the data signal terminal of the next stage data receiving unit when retransmitting.

17. An apparatus according to claim 16, wherein a data processor is provided between the data interface of the receiving section and the sending section for modifying of the contents of the data broadcast.

18. A method for sending data on L-R channels in a FM broadcasting transmitting system, said method comprises the steps of:

a) setting mode controlling signal (mode) at the mode controlling terminal in the sending section;
b) when conducting a stereo broadcast, said mode controlling signal (mode) of the sending section instructing the data transmitting unit to stop operation, and instructing the stereo encoder to operate normally to receive FM broadcast baseband signals;
c) when a FM L-R data broadcast is in progress, said mode controlling signal (mode) of the sending section initiating said data transmitting unit, disconnecting said first switch (KS1) from the output of DSB-SC modulator and connecting said data transmitting unit for data transmission, disconnecting said second switch (KS2) from the output of said divider to cut off pilot signals.

19. A method according to claim 18, wherein at said step a) the mode controlling signal (mode) is automatically controlled through the comparison of the left and right (L, R) voice input signal by the comparator.

20. A method according to claim 18, wherein said step a) manually controls the mode controlling signal (mode).

21. A method according to claim 18, wherein said step c) further comprises;

d) encoding the input data stream.

22. A method according to claim 18, further comprises the step of directly supplying the data signals (DT) to the data signal terminal of the another stage data receiving unit when retransmitting, and the step of selectively modifying of the contents of the data broadcast by the data processor provided between the data interface of the receiving section and the sending section.

23. A method according to claim 21, wherein said encoding step d) further comprises the step of encoding the data stream by a variable-width code, said variable-width code characterized in that data information is transmitted in the form of discrete values of the width of symbol, and said variable-width code is a bipolar non-return to zero pulse signal with each pulse of which being a symbol, different width of pulse representing different information.

24. A method according to claim 23, wherein the pulse width of said variable-width code may have two or more discrete values to form a two-element or multiple element variable-width code, wherein

the symbol with the smallest pulse width is named as a S symbol and consecutive S symbols are referred to consecutive S symbol string;
all other types of variable-width codes are L symbols and consecutive L symbols are referred to consecutive L symbol string;
grouping each field by information string consisting of the same type of symbols, a symbol type identifier is transmitted before each group, then the length information of the group is transmitted, said symbol identifier being a specific code string that starts with a L symbol, and consisting of several L symbols and, when necessary, some appropriate S symbols, each S symbol representing a symbol defined by this (preceding) type identifier, until next type identifier occurs.

25. A method according to claim 24, wherein the spectrum characteristic of two-element variable-width code satisfies:

$$\mathrm{Fdn} = \frac{\mathrm{Smin} + \mathrm{Lmax} + 1}{2 \times [(\mathrm{Lmax} + 1) \times \mathrm{K} + \mathrm{Smin}]} \times \mathrm{B}$$

$$\mathrm{Fup} = \frac{\mathrm{Smin} + 3 \times (\mathrm{Lmax} + 1)}{2 \times [(\mathrm{Lmax} + 1) \times \mathrm{K}\ \mathrm{Smin}]} \times \mathrm{B}$$

symbol rate B satisfies:

$$\mathrm{B} = \frac{2 \times [(\mathrm{Lmax} + 1) \times \mathrm{K} + \mathrm{Smin}]}{\mathrm{Smin} + \mathrm{Lmax} + 1} \times \mathrm{Fdn}$$

or

$$\mathrm{Fup} = \frac{2 \times [(\mathrm{Lmax} + 1) \times \mathrm{K} + \mathrm{Smin}]}{\mathrm{Smin} + 3 \times (\mathrm{Lmax} + 1)} \times \mathrm{Fup}$$

by adjusting the values of Lmax, Smin, K and B, the spectrum of variable-width code, the interference-free performance and the data transmission speed can be determined.

26. A method according to claim 25, wherein when Lmax is infinity (LX), directly encoding method can be employed, i.e. a L symbol corresponds to character "1" and a S symbol to character "0", or a L symbol corresponds to character "0" and a S symbol to character "1", so that the symbol type identifier can be omitted.

27. A method according to claim 25, wherein when Lmax is not infinity, the encoding steps comprises:

1) setting the variable-width code to have two consecutive L symbol strings with different length, used as type identifiers of character "0" and "1" respectively;

2) when one of character "0" and "1" occurs in the data stream, the encoder first outputs consecutive L symbols with single length;

4) when identical characters consecutively occur after the character, the encoder converts each character in the string to a S symbol;

5) when different characters occur after the character or consecutive identical characters, the encoder outputs consecutive L symbol with said another length;

6) repeating steps of 4) - 6) to form encoded data.

28. A method according to claim 25, wherein said polarity synchronizing step comprises:

1) setting consecutive L symbol with single length for indicating changes of symbol type.

2) performing the polarity synchronization on the two-element data information to detect and correct the converse polarity in the decoding process, said polarity synchronizing step comprises:

29. A method according to claim 28, wherein said polarity synchronizing step comprises:

1) setting a "polarity synchronous symbol" and a "synchronous inverted character";

2) performing the "add 1" operation on the two-element data information, so that there is no such polarity synchronous symbol in the processed data stream;

3) performing the "add 0" operation on the two-element data information, so that there is no such synchronous inverted character in the processed data stream;

4) performing the "add synchronization" operation on the "add 1" and "add 0"' processed data stream, dividing said data stream into "block", inserting appropriate numbers of polarity synchronous symbols between blocks.

30. A method according to claim 28, wherein when data blocks are transmitted in the form of "data packet", the character of said data packet can be used as polarity synchronous symbol and the character conversed by said character can be used as synchronous inverted character.

31. A method according to claim 25, wherein the encoding step of transmitting Miller code information (M2, M3, M4) by two-element variable-width code comprises:

1) setting the variable-width code to have three types of consecutive L symbol strings with different length, used as type identifiers of three types of Miller symbol (M2, M3, M4) respectively;

2) when the first type of code string (M2) occurs in Miller code stream, the encoder outputs the L symbol with the first length, and converts each identical symbol in the code string to a S symbol;

3) when the second type of code string (M3) occurs in Miller code stream, the encoder outputs the L symbol with the second length, and converts each identical symbol in the code string to a S symbol;

4) when the third type of code string (M4) occurs in Miller code stream, the encoder outputs the L symbol with the third length, and converts each identical symbol in the code string to a S symbol;

5) repeating steps of 2) -4) to form said code stream.

32. A method according to claim 25, wherein the encoding step of transmitting Miller code information (M2, M3, M4) by two-element variable-width code comprises:

setting the variable-width code to consist of two or less L symbols, combined with appropriate S symbols to form three types of different code strings, used as three types of Miller symbol type identifier respectively, so that the Miller code encoded data can be formed.

33. A method according to claims 31 or 32, wherein said code string may also be single code in said encoding step.

34. A method according to claim 21, further comprises:

e) the step of performing waveform synthesis on said encoded data signals, wherein said waveform synthesizer determines the shape of output pulses based on the code form of the input data stream.

35. A method according to claim 34, wherein the shape of said pulse is a group of waveform module which have been previously optimized.

36. A method according to claim 25, wherein when the encoding process uses the code pattern with Lmax = 1, Smin = 1, (i.e. L1S1), the effective band of the variable-width code signal is 22.714 ~ 53 KHz.

37. A method according to claim 25, wherein when the encoding process uses the code pattern with Lmax = 1, Smin = 25, (i.e. L1S25), the effective band of the variable-width code signal is 53.069 ~ 60.931 KHz.

38. A method according to claim 25, wherein when the encoding process uses the code pattern with Lmax = 1, Smin = 16, (i.e. L1S16), the effective band of the variable-width code signal is 61.2 ~ 74.8 KHz.

39. A method according to claim 25, wherein when the encoding process uses the code pattern with Lmax = 1, Smin = 8, (i.e. L1S8), the effective band of the variable-width code signal is 53 ~ 75 KHz.

40. A method according to claim 25, wherein when the encoding process uses the code pattern with Lmax = $\infty 1$, Smin = 1, (i.e. LXS1), the effective band of the variable-width code signal is 20 ~ 60 KHz.

41. A method according to claim 24, wherein when there are three types of L symbols, and the pulse width ratio of S symbol and L symbol is 1: 2.0 : 2.5 : 3.0, four-element code pattern with Lmax = 1, Smin = 1 (i.e. L1S1) is used to encode the binary information which has been converted to Miller code, so that the effective spectrum of the variable-width code signal is 22.925 ~ 75KHz.

42. A method for receiving data on L-R channels in a FM broadcasting system, said method comprises the steps of:

a) placing a bandpass filter at the receiving section to receive the transmitted broadcast baseband signal (comprising data signal);
b) placing a pilot identifier at the receiving section for identifying pilot signals from the baseband signals;
c) placing a third switch (KR) between said bandpass filter and said data receiving unit, said third switch (KR) is connected to the output of said pilot identifier;
d) when said pilot identifier identifies pilot signals from the broadcast baseband signals, said mode controlling signal (mode) of said output terminal disconnecting said third switch (KR) from the bandpass filter, stopping receiving the data signals, and instructing the stereo decoder to operate normally to recover stereo broadcast signals;
e) when no pilot signal is identified from the broadcast baseband signals by said pilot identifier, the mode controlling signal (mode) of the output terminal connecting said third switch (KR) to the bandpass filter to receive the data signals.

43. A method according to claim 42, wherein said receiving step e) further comprises:

f) discriminating the pulse width of the received variable-width code data signal, which has been waveform synthesized after the amplitude-limited amplification, to recover the pulse width code;
g) decoding the data signals which has been encoded by the pulse width code.

44. A method according to claim 43, wherein said step f) further comprises:

f1) supplying the amplitude-limited amplified variable-width code data signal to a integrator to eliminate the distortion of L symbol and S symbol, so that the error rate of the symbol identifier can be decreased;
f2) supplying the integrated data signals to the symbol identifier to recover the variable-width code.

45. A method according to claim 42, further comprises the step of directly supplying the received data signals (DT) to the data signal terminal (DT) of the next stage data receiving unit, and selectively modifying of the contents of the data signals by the data processor provided between the data interface of the receiving section and the sending section when conducting data retransmmision.

46. A method according to claim 43, wherein said decoding step (g) comprises:

h) identifying the symbol type identifier in encoded data stream which has been encoded by the variable-width code to determine the type of symbol that will arrive;

i) converting each following S symbol to a determined symbol, until next type identifier occurs.

47. A method according to claim 43, wherein when two-element variable-width code is used to transmit data with Lmax being infinity (LX), said decoding step g) comprises the step of directly converting L symbol and S symbol to the corresponding character "0" or "1".

48. A method according to claim 43, wherein said decoding step g) further comprises when the data is transmitted in the form of "data packet", "subtract 1" and "subtract 0" operation will be performed on said decoded data.

49. A FM L-R data broadcasting system, said broadcasting system comprises a sending section and a receiving section, said sending section comprising: an input terminal with a left and a right channel (L, R), a stereo encoder and an output terminal: said receiving section comprising a receiver, a stereo decoder and an output terminal, said system characterized in that:

the sending section further comprises:

a data transmitting unit, the output of which is connected to the adder of said stereo encoder, for providing the data signals to be transmitted;
a first switch (KS1), provided among said DSB-SC modulator, said data transmitting unit and said first bandpass filter, for selectively receiving data signals or audio signals;
a second switch (KS2), provided between said divider and said narrow-band filter, for selectively putting through or cutting off pilot signals;
a mode controlling terminal, connected to the first and second switches (KS1, KS2), for controlling the state of said switches (KS1, KS2), said mode controlling terminal also connected to said data transmitting unit for controlling data transmission of the data transmitting unit;
the receiving section further comprises:

a data receiving unit for receiving the transmitted data signals (DT);
a second bandpass filter, the input of which is connected to the said input terminal, for separating the data signals (DT) from the broadcast baseband signals;
a pilot identifier, the input of which is also connected to said receiver, for identifying pilot signals and outputting the mode controlling signal (mode) at the output terminal;
a third switch (KR) provided between said second bandpass filter and said data receiving unit, and also connected to the output of the pilot identifier.;
wherein said mode controlling signal outputted by the pilot identifier selectively supplies the data signal (DT) to said data decoder through said third switch (KR), so as to recover the transmitted data stream.

50. A system according to claim 49, wherein each of said sending section and said receiving section comprises a data signal terminal respectively, for directly receiving the data signals transmitted by the previous stage data signal terminal, or directly supplying the data signals to the next stage data signal terminal when retransmitting.

51. A system according to claim 49, wherein a data processor for modifying the contents of the data stream is provided between the data interface of said sending section and said receiving section.

52. A system according to claim 49, wherein the mode controlling terminal of said receiving section may be connected to the mode controlling terminal of the sending section to retransmit the mode controlling signal when retransmitting.

53. A method for FM L-R data broadcasting, said method comprises the steps of:

a) setting mode controlling signal (mode) at the mode controlling terminal in the sending section;
b) when conducting a stereo broadcast, the mode controlling signal (mode) of said sending section instructing the data transmitting unit to stop operation, and instructing the stereo encoder to operate normally to receive FM broadcast baseband signals;
c) when a FM L-R data broadcast is in progress, the mode controlling signal (mode) of said sending section initiating said data transmitting unit, disconnecting said first switch (KS1) from the output of DSB-SC modulator (DSB-SC) and connecting the data transmitting unit for data transmission, disconnecting said second switch

(KS2) from the output of the divider to cut off the pilot signals;

d) placing a second bandpass filter at the receiving section to receive the transmitted broadcast baseband signals (comprising data signal);
e) placing a pilot identifier at the receiving section for identifying pilot signals from the baseband signals;
f) placing a third switch (KR) between the bandpass filter and the data receiving unit, said third switch (KR) is connected to the output of said pilot identifier;
g) when said pilot identifier identifies pilot signals from the broadcast baseband signals, the mode controlling signal (mode) of said output terminal disconnecting said third switch (KR) from the bandpass filter, stopping receiving the data signals, and instructing the stereo decoder to operate normally to receive stereo broadcast signals;
h) when no pilot signal is identified from said input signals by said pilot identifier, the mode controlling signal (mode) of the output terminal connecting said third switch (KR) to said bandpass filter to receive the data signals.

54. A method according to claim 53, wherein each of said sending section and said receiving section comprises a data signal terminal respectively, for directly receiving the data signals transmitted by the previous stage data signal terminal, or directly supplying the data signals to the next stage data signal terminal when retransmitting.

55. A method according to claim 53, wherein a data processor for modifying the contents of the data stream is provided between the data interface of said sending section and said receiving section.

56. A method according to claim 53, wherein the mode controlling terminal of said receiving section may be connected to the mode controlling terminal of the sending section to retransmit the mode controlling signal when retransmitting.

57. A data transmitting system using FM broadcasting, said system comprises a sending section and a receiving section, said sending section comprising: an input terminal with a left and a right channel (L, R), a stereo encoder and an output terminal; said system characterized in that:

the sending section further comprises:

a data transmitting unit, the output of which is connected to the adder of said stereo encoder;
a first switch (KS1) provided between said bandpass filter and said adder, for selectively receiving low speed data signals or high speed data signals;
a second switch (KS2) provided between said narrow-band filter and said adder, for selectively putting through or cutting off pilot signals;
a mode controlling terminal, connected to said first and second switches (KS1, KS2), for controlling the data signal transmission pattern of the data transmitting unit;
the receiving section further comprises:

a data receiving unit for receiving the transmitted data signals (DT);
a low speed bandpass filter, the input of which is connected to said receiver, for separating the low speed data signals (DT) from the broadcast baseband signals;
a high speed bandpass filter, the input of which is connected to said receiver, for separating the high speed data signals (DT) from the broadcast baseband signals;
a pilot identifier, the input of which is also connected to the receiver, for identifying pilot signals and outputting the mode controlling signal (mode) at the output terminal;
a third switch (KR) provided between said high speed filter and said data receiving unit, and also connected to the output of the pilot identifier,.

58. A system according to claim 57, wherein each of said sending section and said receiving section comprises a data signal terminal respectively, for directly receiving the data signals transmitted by the previous stage data signal terminal, or directly supplying the data signals to the next stage data signal terminal when retransmitting.

59. A system according to claim 57, wherein a data processor for modifying the contents of the data stream is provided between the data interface of said sending section and said receiving section.

60. A system according to claim 57, wherein the mode controlling terminal of said receiving section may be connected to the mode controlling terminal of the sending section to retransmit the mode controlling signal when retransmitting.

61. A method for data signal transmission using FM broadcasting, said method comprises the steps of:

a) setting the mode controlling signal (mode) at the mode controlling terminal in the sending section;
b) when conducting a stereo broadcast, the mode controlling signal (mode) of said sending section instructing the first switch (KS1) to close, transferring DSB-SC signals into the adder so that the data transmitting unit is in the low speed transmitting state (e.g. pattern 4);
c) when broadcasting monophonic program, the mode controlling signal (mode) of said sending section cutting off DSB-SC signals through the first switch (KS1), cutting off pilot signals through the second switch (KS2) so that the data transmitting unit is in the high speed data transmitting state (e.g. pattern 6);
d) placing a high speed bandpass filter at the receiving section to receive the transmitted low speed data signals when conducting a stereo broadcast;
e) placing a low speed bandpass filter at the receiving section to receive the transmitted high speed data signals when conducting a monophonic program broadcast;
f) placing a pilot identifier at the receiving section for identifying pilot signals from the baseband signals;
g) placing a third switch (KR) among the high and low speed bandpass filter and the data receiving unit, said third switch (KR) is connected to said pilot identifier;
h) when said pilot identifier identifies pilot signals from the broadcast baseband signals, the mode controlling signal (mode) of said output terminal connecting the low speed filter to the data receiving unit through said third switch (KR), and making the data receiving unit to be in the low speed data receiving state;
i) when no pilot signal is identified from the broadcast baseband signals by said pilot identifier, the mode controlling signal (mode) of the output terminal connecting the high speed filter to the data receiving unit through said third switch (KR), and making the data receiving unit to be in the high speed data receiving state.

62. A system according to claim 61, wherein each of said sending section and said receiving section comprises a data signal terminal respectively, for directly receiving the data signals transmitted by the previous stage data signal terminal, or directly supplying the data signals to the next stage data signal terminal when retransmitting.

63. A system according to claim 61, wherein a data processor for modifying the contents of the data stream is provided between the data interface of said sending section and said receiving section.

64. A system according to claim 61, wherein the mode controlling terminal of said receiving section may be connected to the mode controlling terminal of the sending section to retransmit the mode controlling signal when retransmitting.

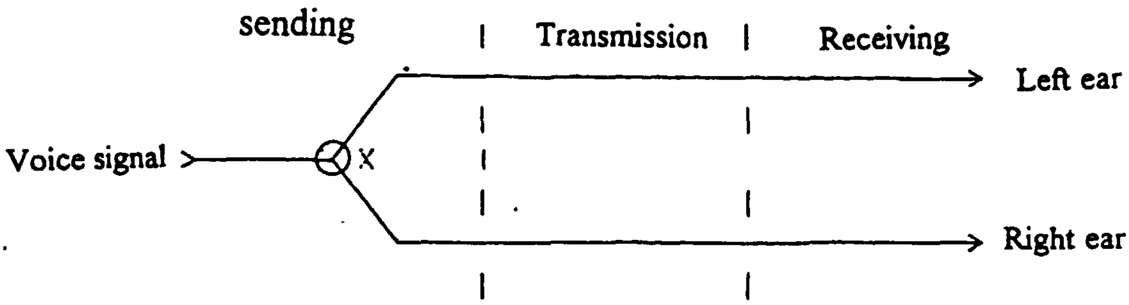


Fig.1 A transmission mode of a voice signal in a conventional stereo broadcast

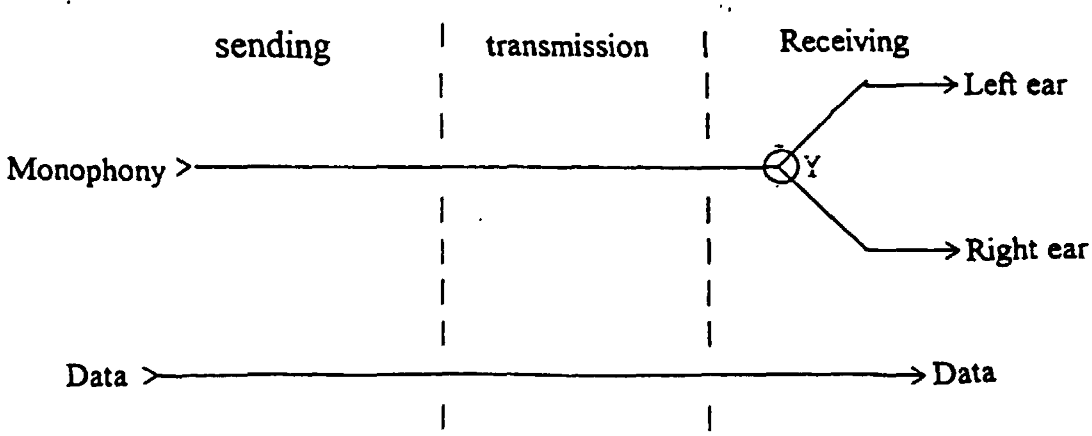
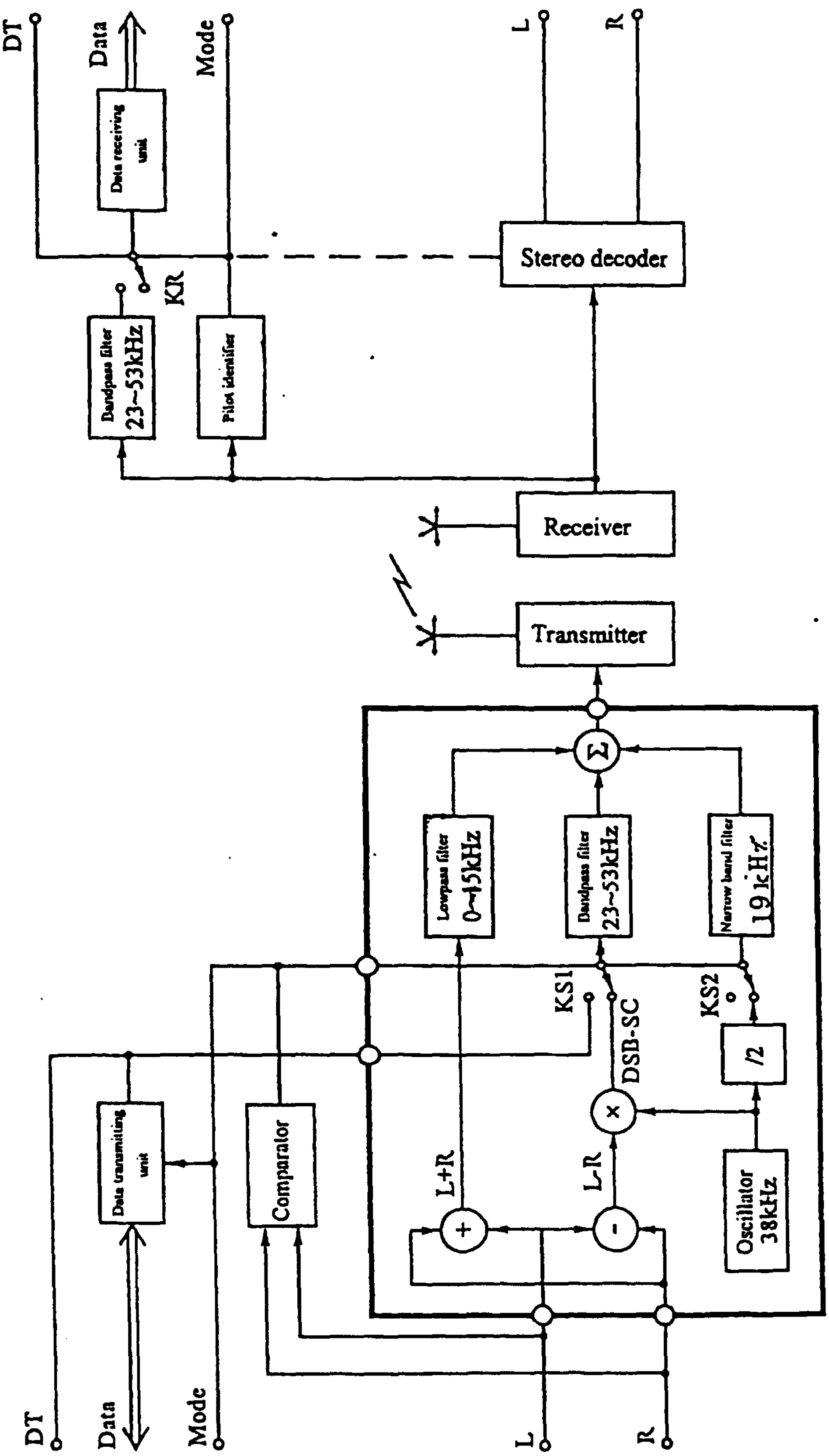


Fig.2 A broadcast mode of an FM "monophony +data"



Principle of an FM "stereo " / "monophony +data" broadcast

Fig.3

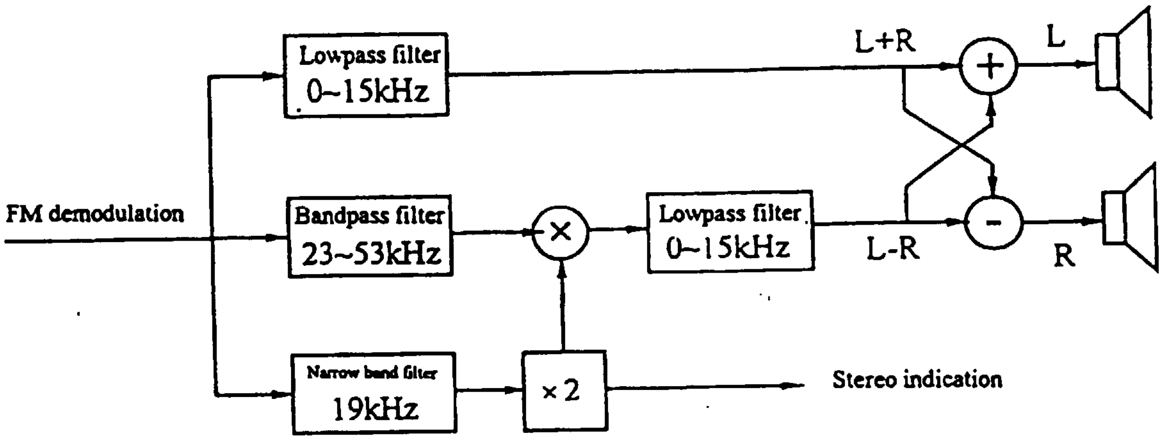


Fig.4 FM stereo decoder

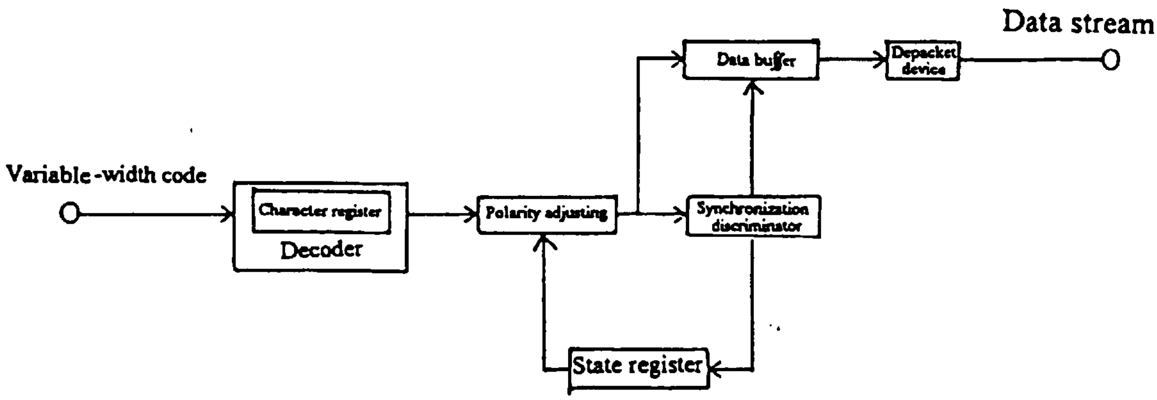


Fig5 Decoding principle of variable-width code when L_max=1

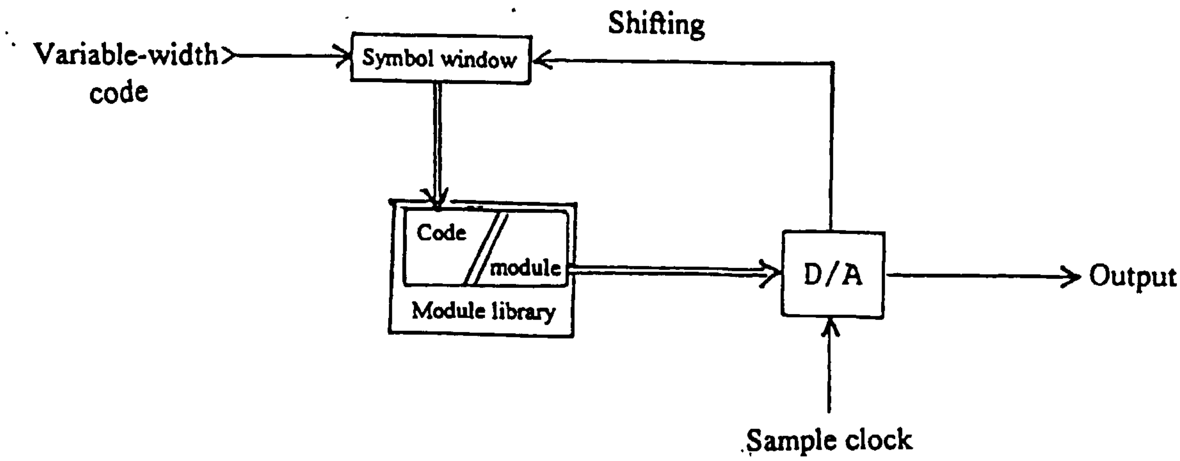


Fig6 Operating principle of a waveform synthesizer

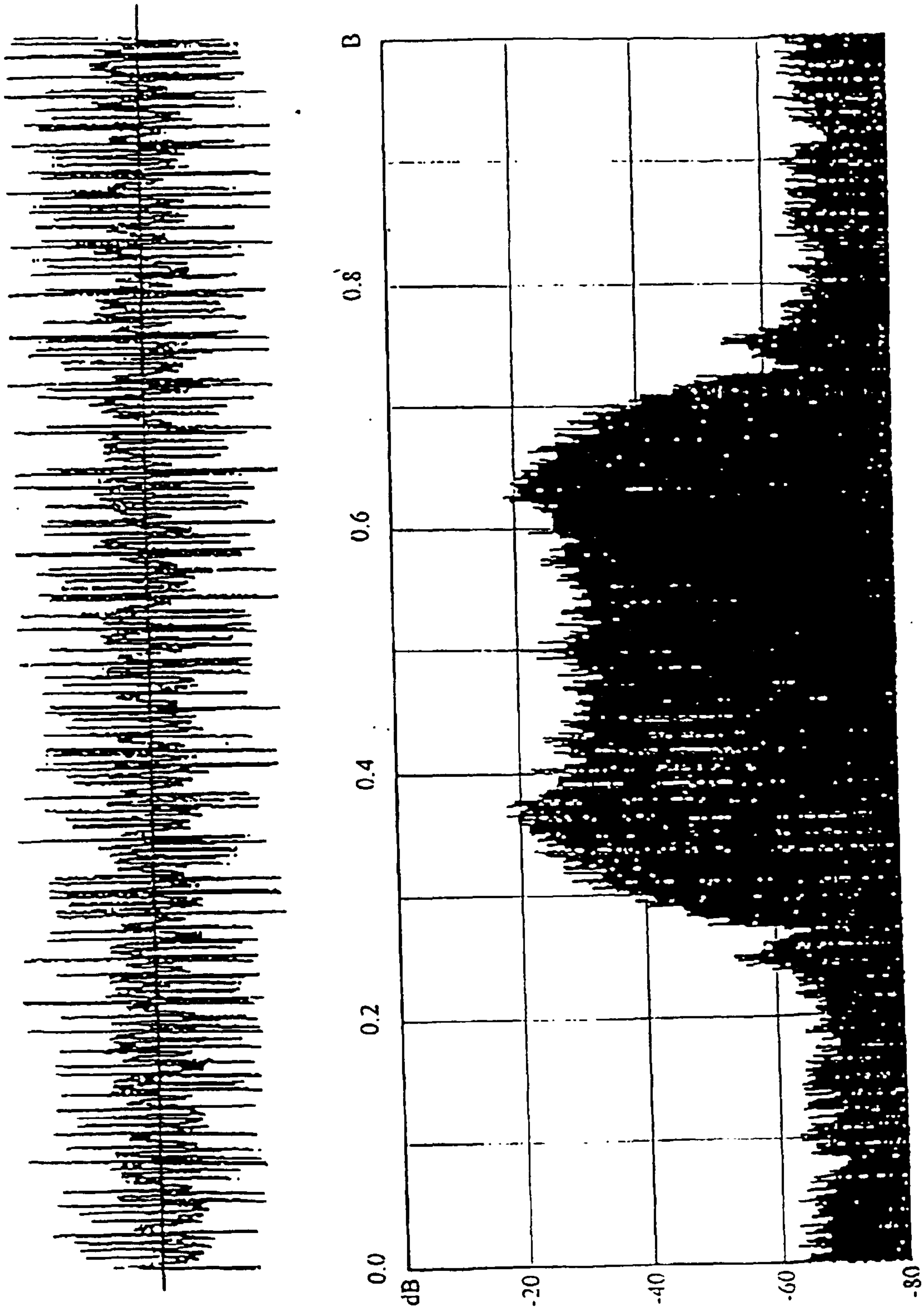
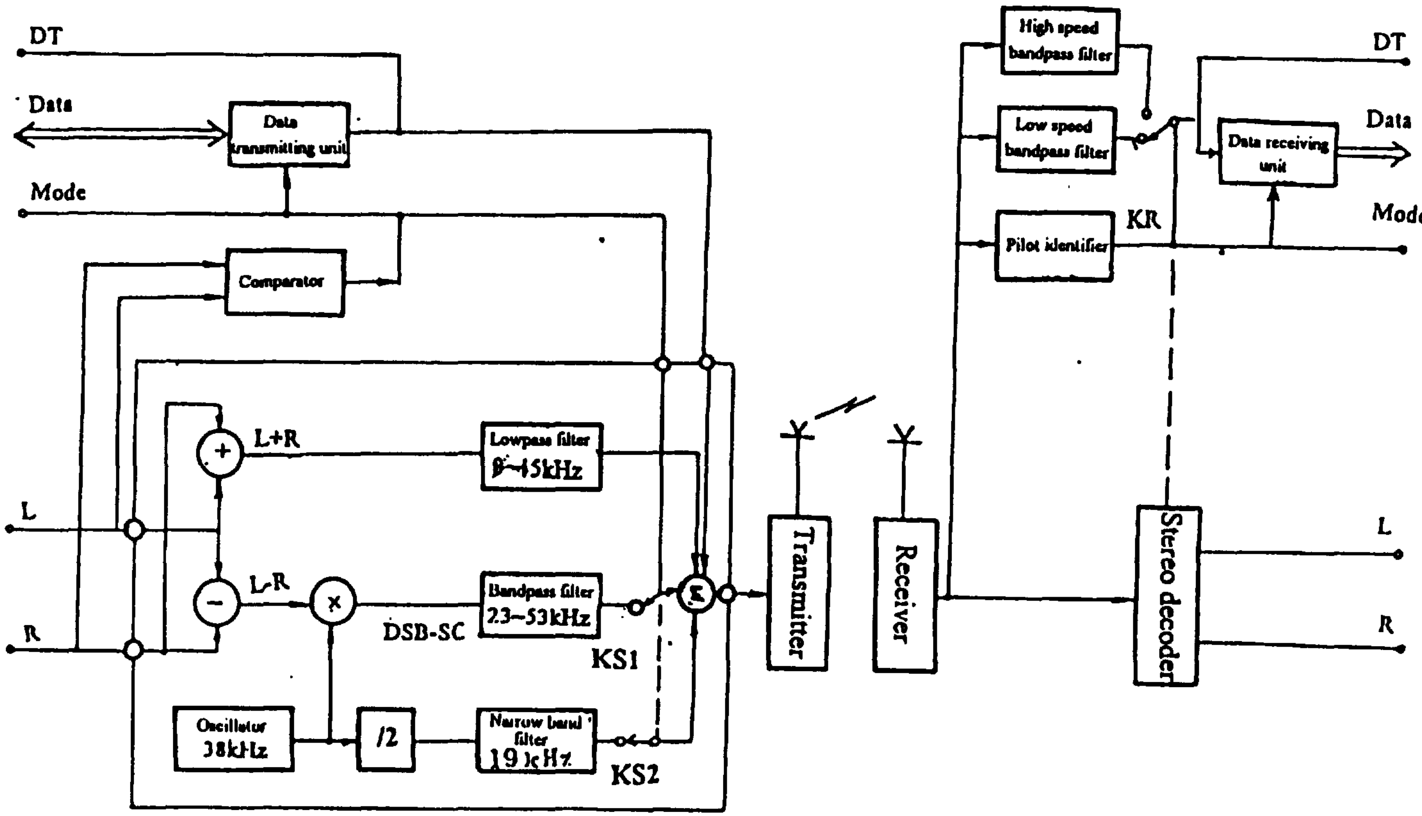


Fig7 Waveform and frequency spectrum of a variable-width code signal



Principle of an FM "stereo+low speed data "/"monophony+hign speed data"broadcast

Fig8

INTERNATIONAL SEARCH REPORT	International application No. PCT/CN 96/00089

A. CLASSIFICATION OF SUBJECT MATTER

IPC^6 H04H 5/00

According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)

IPC^6 H04H 5/00

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practicable, search terms used)

C. DOCUMENTENTS CONSIDERED TO BE RELEVANT

Category*	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
A	US—A—5119503(Roy J. Mankovitz), 02. June 1992(02. 06. 92) Abstract	1,12,18
A	US—A—5222143 (Sam Sung Electronics Co.Ltd.), 22. June 1993(22. 06. 93) Abstract	1,12,18
A	US—A—5038402(General Instrument Corporation), 06. August 1991(06. 08. 91) Abstract	1,12,18

☐ Further documents are listed in the continuation of Box C. ☒ See patent family annex.

* Special categories of cited documents:
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"T" later document published after the international filing date or priority date and not in conflict with the application but cited to understand the principle or theory underlying the invention
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"Y" document of particular relevance; the claimed invention cannot be considered to involve an inventive step when the document is combined with one or more other such documents, such combination being obvious to a person skilled in the art
"&" document member of the same patent family

Date of the actual completion of the international search	Date of mailing of the international search report
16 January 1997(16. 01. 97)	23 JAN 1997 (23.01.97)
Name and mailing address of the ISA/	**Authorized officer**
Chinese Patent Office, 6 Xitucheng Rd. Jimen Bridge, Haidian District, 100088 Beijing, China	Song, Yanqin
Facsimile No. (86—10)62019451	Telephone No. (86—10)[redacted]

Form PCT/ISA/210(second sheet)(July 1992)

INTERNATIONAL SEARCH REPORT
Information patent family members

International application No.
PCT/CN 96/00089

Patent document cited in search report	Publication date	Patent family member(s)	Publication date
US—A—5,119,503	02.06.92	无	
US—A—5,222,143	22.06.93	无	
US—A—5,038,402	06.08.91	EP—A2—372499	13.06.90

Form PCT/ISA/210(patent family annex)(July 1992)