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(54) **Receiver for receiving digital audio programmes as well as the supplementary data contained in the audio programmes**

(57) The invention relates to a receiver of a transmission wave in which a specific information data signal and its validity identification data signal are transmitted by the same frequency band, in which the receiver desirably reproduces specific information only in a situation where the specific information can be truly reproduced in the reception system. A demodulating unit (31 to 36) receives the transmission wave and demodulates it to a predetermined digital signal. A decoding unit (38 and 40) decodes the digital signal, and a detecting unit 3B obtains an error amount of the digital signal recognized in the decoding process. An evaluating unit 3B evaluates the error amount, and a detecting unit 3A detects an identification data signal from a decoded output. The control part 3C (30) reproduces and outputs the specific information data signal concerning the identification data signal based on the evaluation result and the identification data signal. Even when a code rate of the specific information data signal to be information reproduced differs from that of the validity identification data signal, the specific data signal can be desirably information reproduced at a proper timing. The invention also relates to a digital audio signal receiver having a muting function for a reproduction audio sound and accomplishing a good muting operation even for a digital audio signal having no error detection code. A control part (30) obtains an error amount of a digital signal

to be recognized in the decoding process and a muting block (42) effects the muting of an audio signal to be reproduced based on the digital signal at a muting level according to the error amount.

EP 0 865 176 A2

Description

BACKGROUND OF THE INVENTION

1. Field of the Invention

The invention relates to an information data signal receiver or a digital audio signal receiver in a system in which a specific data signal to be reproduced as information and an identification data signal to identify that the specific data signal is valid are transmitted by transmission waves of the same frequency band and to a system using such a receiver. The invention also relates to a digital audio signal receiver having a muting function.

2. Description of Related Art

As a system for transmitting and receiving a broadcasting wave having a data signal of a predetermined format including a digital audio signal, a digital audio broadcasting (hereinafter, referred to as DAB) system which conforms to the European standard (Eureka 147) is being put into practical use.

The basic construction of a transmission system in the DAB system is shown in Fig. 1.

In Fig. 1, a digital audio signal which is obtained by simply sampling an analog original audio signal at a predetermined rate and quantizing it and has been digitized by, for example, a linear PCM is supplied to a high efficient coder 1. The high efficient coder 1 executes what is called a data compression and, more specifically, processes an input signal by a compression system according to Layer II of ISO/IEC 11172-3 (Layer II of MPEG audio) as an international standard system. This compressing method is also called an MUSICAM (Masking pattern adapted Universal Subband Integrated Coding And Multiplexing) system and reduces necessary audio information by previously omitting audio information by using masking characteristics of the human sense of hearing. A data signal compressed by this method has a predetermined format as shown in Fig. 2.

The MUSICAM format uses predetermined frames (6144 bits) as a unit and is coarsely divided into blocks of a header, side information, main audio data, and data concerning a program. The header is constructed by subblocks of ASSD (audio service sync data), a DAB header, and CRC (Cyclic Redundancy Check). The side information is made up of subblocks of bit allocation information, ScFSI (scale factor selection information), and a scale factor. The main audio data is occupied by a subband sampling signal. The data concerning a program is, further, constructed by subblocks of a stuffing, X-PAD (data concerning a variable length program), ScF-CRC (CRC for a scale factor), and F-PAD (data concerning a fixed length program). The CRC in the header is a CRC which is used for the DAB header, bit

allocation information, and ScFSI.

The compression encoded data signal with the format is further transferred to a channel coder 2, by which a redundancy is added by using a convolutional coding for the purpose of error correction. In the channel coder, a process called a punctured is also executed in addition to a pure convolutional coding. It relates to a process for extracting and transmitting a part of the data signal which has been convolutionally coded. In the process, an UEP (Unequal Error Protection) process for reducing an amount of signal in a portion having a high significance to be extracted and increasing an amount of signal in a portion having a less significance to be extracted is executed. Those processes will now be briefly explained. Data of a value shown by one bit as an input is added with redundancy bits by the convolutional coding and is converted to data of a value indicated, for example, by four bits. Subsequently, for example, one bit among the converted 4-bit data is extracted by the punctured process, so that 3-bit data is generated. In the case of this example, since three bits are finally obtained for one bit of the input, a code rate (encoding ratio) is equal to 1/3. The code rate corresponds to a ratio of the number of input bits and the number of output bits in the channel coder 2. In other words, the code rate corresponds to (the number of information bits)/(the total number of output bits).

When a value of the code rate is high, this means that the number of redundancy bits is small and the number of true encoding bits (information bits) is large, so that this state is called a low protection level. When a value of the code rate is low, this means that the number of redundancy bits is large and the number of true encoding bits is small, so that this state is called a high protection level. The code level can be selected in accordance with the contents of audio services by a method of increasing the protection level in the case of the music broadcasting and decreasing the protection level in the case of a speech broadcasting. The code rate in the channel coder 2 and the number of bits to be extracted in the in the punctured process correspond to each other.

The output data signal of the channel coder 2 obtained as mentioned above is transferred to a time base processor 3, by which it is converted to a data signal which was time interleaved.

Similar processes are also executed to the original audio signal of another channel and the signal is finally converted to a time-interleaved data signal. This process is omitted in the block diagram of Fig. 1. With respect to not only the audio signal but also the general data signal, similar processes are also executed by encoding blocks 4 and 5 and a time base block 6, thereby interleaving the signal. The general data signal, however, differs from the audio signal with regard to a point that an encoding to use a format of a bit stream or a packet multiplex which is not structured is performed as a high efficient encoding. The general data signal

transmits text services of a wide range such as weather forecast or traffic information (TMC: Traffic Message Channel) or program list of one day which are not immediately necessary for selection of a desired program of the receiver.

Each of the time-interleaved data signals obtained as mentioned above is supplied to a multiplexer (MUX) 7. An MUX control unit 8 controls the multiplexer 7 and the control unit forms an MCI (Multiplex Configuration Information) for the input signal of the multiplexer 7 in accordance with information of a service configuration which has previously been given. The multiplexer 7 time-division multiplexes the input signal according to the MCI and supplies a multiplexed output to a frequency base processor 9.

An FIC data signal to be allocated to an FIC (high speed information channel) generated by an FIC generator 10 is also supplied to the frequency base processor 9. The frequency base processor 9 frequency interleaves both of the FIC data signal and the multiplexed output from the multiplexer 7. The MCI from the MUX control unit 8 is supplied to the FIC generator 10. The FIC generator 10 divides the input signal into subblock data on the basis of the MCI and other various predetermined information, adds a CRC for the subblock data, forms a data signal of a main block, and further performs a convolutional coding similar to that by channel coders 2 and 4 to the data signal of the main block, thereby obtaining a final FIC data signal. In the DAB system, as for the FIC data signal, the code rate is fixed to a relatively low value of 1/3 instead of not performing the time interleave, thereby raising an error correcting ability in a reception system.

In addition to the main signal from the frequency base processor 9, a sync signal generated from a sync signal generator 12 is also supplied to an OFDM (Orthogonal Frequency Division Multiplex) modulator 11. On the basis of those signals, the OFDM modulator 11 modulates a number of carriers mutually having an orthogonal relation, thereby obtaining an output. Although a principle of the modulation is not described in detail here because it is already well known, an OFDM signal of a predetermined format as schematically shown in Fig. 3 is derived from an output of the OFDM modulator 11.

That is, the OFDM signal uses a transmission frame (for example, 24 msec in mode II) specified in the DAB as a unit and, therefore, forms a transmission frame series. One transmission frame is largely divided into blocks of a sync channel, FIC, and MSC (main service channel). The sync channel block is constructed by: a null signal portion for coarse synchronization corresponding to the non-existence of a transmission signal (RF signal); and a reference phase symbol serving as a reference phase for a differential QPSK demodulation in an OFDM demodulation. The FIC block is divided into three subblocks and includes high speed information blocks 1 to 3. The MSC block is divided into 72 sub-

blocks and includes data fields 1 to 72.

The high speed information block is further divided. In addition to a guard interval, the high speed information block has various information data including the foregoing MCI showing a multiplex arrangement or configuration form of the data signals in the MSC, a service name (label) of a program, program contents information, and information such as paging code, traffic message control, various identification codes, and the like. Its CRC code is also provided in the high speed information block together with those various information data. It should be noted that information (ASW: Announcement Switching) indicating that traffic information in the MSC is being broadcasted like TA data which has already been used in an RDS (Radio Data System) which has already been put into practical use in Europe has also been allocated to the various kinds of information data in the FIC.

As a receiver using the ASW information, in the receiver disclosed in Japanese Patent Kokai No. 8-88572, the start of the broadcasting of traffic information is recognized in the FIC, thereby realizing that the relevant traffic information audio sound is preferentially reproduced. It should be further noted that data regarding a code rate which is set to the channel coder 2 is included in the data that is allocated to the FIC. The code rate data includes information indicative of punctured bits, namely, the bits extracted by the punctured process.

It will be understood that each of the divided blocks of the MSC data field is made up of a guard interval arranged at the head of the block and the data symbols subsequent thereto in a manner similar to the FIC block. The guard interval is provided to avoid an influence by an intercode interference due to an influence by multipath. In an actual format, a partial signal waveform in a valid symbol period is used. Referring now to Fig. 1, it will be understood that the MSC block is formed by a system of blocks 1, 2, and 3 and a system of blocks 4, 5, and 6 (namely, main audio information and additional information such as actual traffic information audio data concerning the ASW or the like are stored here). There is, however, hardly a regular relation with audio frames as shown in Fig. 2 and it can be regarded that at least a part of the audio frame includes it.

Although a CRC code has been added to the data signal corresponding to the FIC every high speed information block, no CRC is added to the data signal corresponding to the MSC. A time interleave, however, is not performed to the data signal corresponding to the FIC and a time interleave has been performed to the data signal corresponding to the MSC. This is because the FIC needs to be decoded sufficiently prior to the MSC and in order to certainly demodulate the data signal of the MSC, the data signal of the FIC is promptly demodulated without a delay that is required to deinterleave. It should further be noted that as already mentioned above, although the FIC data signal has a predeter-

mined code rate, a code rate of the MSC data signal is made variable. Since the MSC data signal has been time interleaved, its code rate is higher as a whole than that of the FIC data signal and is set to about 1/2 as an average.

The OFDM signal with the format is converted into an analog signal by a D/A (digital-analog) converter 13, is further subjected to an orthogonal modulation by an orthogonal modulator 14, and is radiated as an RF (Radio Frequency) signal from a transmitting antenna 17 through an up-converter 15 of a frequency and a power amplifier 16.

As shown in Fig. 2, the MUSICAM format data which is used in the DAB is partially added with a CRC and only two CRCs of a CRC (hereinafter, called a header CRC) for the DAB header, bit allocation information, and ScFSI which is arranged at a position near the head and a CRC (ScF-CRC) for a scale factor (weight coefficient of every audio subband) are used.

Since the CRC as an error detection bit has only an error detecting ability (ordinarily, the detecting ability changes in dependence on the number of bits of the CRC) of a predetermined data block, no CRC is added to a subband sample (signal) as will be also understood from Fig. 2, so that the error detection of the sampling signal by the CRC is not performed at all. In place of it, it has been proposed that an error correction of the sampling signal by a Viterbi decoder is executed in the reception system in the DAB. It is, however, not guaranteed that the subband sampling signal having errors is perfectly corrected by the Viterbi decoding.

Hitherto, however, a possibility that the sampling signal is erroneously demodulated in the reception system and an audio sound is reproduced and outputted as it is still remains. As a technique to avoid the inconvenience, a method of muting the reproduction audio sound in response to the error detection by the CRC by using the ScF-CRC for scale factor is considered. In the case of this technique, however inherently, the generation of the error in the scale factor seriously damages the reproduction audio sound of the sampling signal and executes a muting process on the assumption that the sampling signal obviously causes a defective reproduction. In other words, if an error in the scale factor is not generated, the audio reproduction of the subband sampling signal is inevitably performed. Even if the error has occurred in the subband sampling signal itself, no muting process is executed and the signal is generated as a noise sound as it is.

The noise sound sometimes has a possibility that it exerts a serious influence on a quality of the reproduction audio sound in a manner similar to the case of generation of the error in the scale factor. In the case of muting on the basis of only the CRC as in the conventional system, in a state such that the occurrence and non-occurrence of errors are alternately repeated for a short time, since an acoustic output is finely turned on/off in response to the occurrence/non-occurrence,

the reproduction audio sound is fairly hard to listen for the listener.

An important point in the invention which will be described in detail hereinbelow, further, is that in the case of the above technique, in spite of a fact that the information of the FIC has desirably been reproduced, the information reproduction of the MSC is shut off by the muting. For example, when reproducing the traffic information audio sound annexed to the main audio sound, although the ASW information of the FIC is used, according to the conventional reception system, the contents of the ASW information are monitored and when the ASW information is ON, namely, while the traffic information audio sound is being provided or after that the start of the audio sound was shown could be detected, an acoustic output is reproduced with respect to the corresponding traffic information audio data of the MSC. In this instance, even if it is recognized by the ASW information of the FIC that the traffic information is being provided in the reception system, the error is detected by the CRC in the audio frame of the MUSICAM format and the muting process is executed as mentioned above, so that a possibility of contradiction such that the subband sampling signal indicative of the corresponding traffic information audio sound is not reproduced cannot be denied.

An inconvenience due to the contradiction is typical in the case where the system is constructed in a manner such that if traffic information is provided during the reproduction of music by a disc player in, for example, a vehicle-mounted audio system or the like, the music reproduction of the disc player is once stopped and the source is switched to the audio output of the traffic information and, when the providing of the traffic information is finished, the music reproduction by the disc player is restarted or the like. In this case, namely, since the audio sound is muted as mentioned above in spite of a fact that the source has been switched to the audio output of the traffic information, a soundless state continues or an intermittent audio sound output is performed. The source switching to conveniently provide the traffic information contrarily causes a result such that a feeling of physical disorder is given to the listener.

It has also been found out that there is a nature such that even if the ASW information indicative of the providing of the traffic information could be recognized in the reception system, the traffic information audio sound is not always desirably reproduced due to the signal format based on the code rate specified in the DAB. As mentioned above, namely, since the code rate of the FIC data signal is lower (stronger against errors) than that of the MSC data signal, even in the same transmission wave, on the reception side, there is a possibility such that although the FIC lies within the error correction limit, the MSC is out of the error correction limit. A situation that even if the ASW information data belonging to the FIC data signal could be decoded, the corresponding traffic information audio data belonging to the

MSC data signal cannot correctly decoded can occur. This situation also causes the above inconvenience.

OBJECTS AND SUMMARY OF THE INVENTION

The invention is made in consideration of the above points and it is an object of the invention to provide a digital audio signal receiver which can accomplish a preferable muting operation even for a digital audio signal having no error detection code.

Another object of the invention is to provide an information data signal receiver which can desirably reproduce specific information only in a situation where the specific information can be truly reproduced in a reception system.

A further object of the invention is to provide an information data signal receiver in a system in which a specific data signal (also including an additional information signal such as a traffic information audio data signal or the like) to be information reproduced and an identification data signal to identify that the specific data signal is valid are transmitted by a same transmission wave, wherein even when a code rate of the specific data signal and a code rate of the identification data signal are different, the specific data signal can be desirably information reproduced at a proper timing.

A receiver according to the present invention is an information data signal receiver for receiving a transmission wave in which a specific information data signal and an identification data signal to identify that the specific information data signal is valid are transmitted by a same frequency band, the receiver comprising: receiving demodulating means for receiving the transmission wave and demodulating it to a predetermined digital signal; decoding means for decoding the digital signal; error detecting means for obtaining an error amount of the digital signal which is recognized in a decoding process by the decoding means; evaluating means for evaluating the error amount; identification data detecting means for detecting the identification data signal from a decoded output of the decoding means; and control means for reproducing and outputting the specific information data signal concerning the identification data signal on the basis of an evaluation result of the error amount by the evaluating means and the identification data signal.

A system according to the present invention is system using an information data signal receiver, which system comprises: an information data signal receiver for receiving a transmission wave in which a specific information data signal and an identification data signal to identify that the specific information data signal is valid are transmitted by a same frequency band, the receiver including receiving and demodulating means for receiving the transmission wave and demodulating it to a predetermined digital signal, decoding means for decoding the digital signal, error detecting means for obtaining an error amount of the digital signal which is

recognized in a decoding process by the decoding means, evaluating means for evaluating the error amount, identification data detecting means for detecting the identification data signal from a decoded output of the decoding means, and control means for reproducing and outputting the specific information data signal concerning the identification data signal on the basis of an evaluation result of the error amount by the evaluating means and the identification data signal; an audio signal output source different from the receiver; selecting means for selectively outputting either one of an output audio signal of the audio output source and an output audio signal of the receiver in accordance with the reproduction output control signal; and means for generating an acoustic output in accordance with a selection output of the selecting means.

According to the present invention, there also is provided a digital audio signal receiver having a muting function for a reproduced sound, and comprising a receiving and demodulating means for receiving a transmission wave including a digital audio signal and demodulating the received signal to a predetermined digital signal, decoding means for decoding the digital signal, a control means for obtaining an error amount of the digital signal recognized in a decoding process by said decoding means, and a muting means for effecting a muting of an audio signal to be reproduced based on the digital signal at a muting level corresponding to the said error amount.

BRIEF DESCRIPTION OF THE DRAWINGS

Fig. 1 is a block diagram showing a schematic construction of a transmission system in a DAB system;

Fig. 2 is a diagram showing a format of an audio frame according to MPEG audio layer II in the DAB system;

Fig. 3 is a diagram showing a format of a transmission frame in the DAB system;

Fig. 4 is a block diagram showing a schematic construction of a DAB receiver in an embodiment according to the invention;

Fig. 5 is a flowchart showing a processing procedure for a reproduction output control which is executed by a control unit in a receiver in Fig. 4;

Fig. 6 is a block diagram showing a schematic construction of a DAB receiver in the second embodiment according to the invention;

Fig. 7 is a flowchart showing a processing procedure for a reproduction output control which is executed by a control unit in the receiver of Fig. 6;

Fig. 8 is a block diagram showing a schematic construction of a DAB receiver in the third embodiment according to the invention;

Fig. 9 is a block diagram showing a schematic construction of a system using a DAB receiver in the fourth embodiment according to the invention;

Fig. 10 is a block diagram showing a schematic construction of a DAB receiver according to the invention;

Fig. 11 is a flowchart showing a processing procedure for a muting control which is executed by a control unit in the receiver of Fig. 10;

Figs. 12A to 12C are conceptual time charts showing muting control formats which are realized by the construction of Fig. 10 and processes in Fig. 11;

Fig. 13 is a block diagram showing a schematic construction of a DAB receiver in the second embodiment according to the invention;

Fig. 14 is a flowchart showing a part of a processing procedure for a muting control which is executed by a control unit in the receiver of Fig. 13;

Fig. 15 is a flowchart showing another portion of the processing procedure for the muting control which is executed by the control unit in the receiver of Fig. 13;

Figs. 16A - 16C are conceptual time charts showing a muting control format which is realized by the construction of Fig. 13 and processes in Figs. 14 and 15;

Fig. 17 is a block diagram showing a schematic construction of a DAB receiver in the third embodiment according to the invention;

Figs. 18A to 18C are conceptual time charts showing a muting control format which is realized by the construction of Fig. 17;

Fig. 19 is a block diagram showing a schematic construction of a DAB receiver in the fourth embodiment according to the invention;

Fig. 20 is a flowchart showing a processing procedure for a muting switching operation which is executed by a control unit in the receiver of Fig. 19;

Figs. 21A to 21C are conceptual time charts showing muting control formats which are realized by the construction of Fig. 19 and processes in Fig. 20;

Fig. 22 is a flowchart showing a processing procedure for a muting addition which is executed by a control unit in a modification example based on the construction of the receiver of Fig. 17; and

Fig. 23 is a block diagram showing a schematic construction of a DAB receiver in the fifth embodiment according to the invention.

DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENTS

Embodiments of the invention will now be described hereinbelow in detail with reference to the drawings.

Fig. 4 shows a schematic construction of a DAB receiver of an embodiment as a digital audio signal receiver according to the invention.

In Fig. 4, the RF signal captured by the receiving antenna 31 is supplied to a front-end 32 as tuning means. The front-end 32 converts a signal of a desired

frequency in the RF signal into an intermediate frequency signal in cooperation with a phase locked loop (PLL) 33 and supplies the intermediate frequency signal to an orthogonal demodulator 34. The PLL 33 generates a local oscillating frequency signal for a frequency conversion in a high frequency amplifying unit in the front-end 32 in accordance with a station selection signal from a control part 30 and sets a substantial station selecting operation, namely, a tuning frequency.

The orthogonal demodulator 34 demodulates a QPSK wave and comprises, for example, two mixers, a local oscillator, and a phase shifter. One of the mixers mixes an oscillation signal which is generated from the local oscillator and an intermediate frequency signal and generates an in-phase component signal I of a baseband signal. The other mixer mixes the signal obtained by shifting the phase of the oscillation signal generated from the local oscillator by only 90° and the intermediate frequency signal and generates an orthogonal component signal Q of the baseband signal.

The in-phase component signal I and orthogonal component signal Q are supplied to an A/D (analog-digital) converter 35. The A/D converter 35 converts the in-phase and orthogonal component signals I and Q to digital signals and supplies them to an OFDM demodulator 36. The OFDM demodulator 36 executes a process [including high speed Fourier transformation (FFT) and a differential demodulation of every carrier for a signal of every carrier derived by the FFT] opposite to the modulating process by the OFDM modulator 11 shown in Fig. 1. Its demodulating principle is not described in detail here because it is well known by various literatures.

A demodulation output of the OFDM demodulator 36 has a format shown in Fig. 3 and is subjected to a process opposite to a time interleave and a frequency interleave, namely, a de-interleave with respect to the time/frequency by a time/frequency base processor 37 and is supplied to an error corrector 38. A Viterbi decoder is actually used as an error corrector 38. The Viterbi decoder specifies a data train signal of a most probable value from the de-interleaved demodulation signal while performing an inverse punctured process (which will be clarified hereinafter). The Viterbi decoding is a decoding to decode a convolutional coding which is executed in channel coders 2 and 5 in Fig. 1 and can be regarded as a process for converting the convolutional coding signal having been input to a signal indicative of the same value as that of the output signals of the high efficient coders 1 and 4 in Fig. 1.

The Viterbi decoded signal is supplied to a signal distributing circuit 39. The signal distributing circuit 39 distributes the FIC data signal in the input signal to the control part 30 and distributes the MSC data signal to an audio decoder 40 and a data decoder 41. More specifically speaking, the data signal regarding the audio signal in the MSC data signal is supplied to an audio decoder 40. The data signal regarding the general data

signal is supplied to the data decoder 41. The audio decoder 40 is what is called an MUSICAM decoder for performing a decoding (namely, expansion of data) corresponding to the high efficient coder 1 in Fig. 1. The data decoder 41 performs a decoding corresponding to the high efficient coder 4 in Fig. 1.

The data signal which is supplied to the audio decoder 40 has a format in which the foregoing audio frame shown in Fig. 2 is treated as a unit. The audio decoder 40 decodes the data signal so as to be reconstructed to an original digital audio signal, supplies a decoded output as a decoding result to a D/A (digital-analog) converter 43, and transfers error information which is obtained during the decoding, namely, error detection information based on at least one of the header CRC and the ScF-CRC to the control part 30. A header detecting process to recognize the start of the audio frame is included in the decoding process of the audio decoder 40. The header detection information which is obtained by the header detecting process is also transferred to the control part 30.

The control part 30 receives the error detection information and the header detection information, executes a predetermined arithmetic operating process based on those information and the FIC data signal from the signal distributing circuit 39, and supplies a reproduction output control signal according to the arithmetic operation result to the audio decoder 40. As for the audio decoder, whether the audio signal of which channel in the MSC should be decoded or generated is designated by the reproduction output control signal.

More specifically speaking, the control part 30 has: an FIC decoder 3A to detect the ASW information signal in the FIC data signal; error rate calculating evaluating means 3B for calculating an error rate under predetermined conditions of the input data signal based on the error detection information and the header detection information from the audio decoder 40, discriminating and evaluating whether the calculated error rate is larger than a predetermined value or not, and generating an error evaluation signal indicative of the discrimination result; and an AND gate 3C to which the ASW information signal and the error evaluation signal are supplied. The control part 30, consequently, generates a reproduction output control signal which is made significant only when the ASW information signal indicates that the traffic information has been provided and the error evaluation signal indicates that the error rate of the input data signal of the audio decoder 40 is smaller than the predetermined value.

When the reproduction output control signal is significant, namely, is set to the high level in the embodiment, the audio decoder 40 selects the audio channel of the traffic information in the MSC in response to the reproduction output control signal at the high level. A decoding switching operation from the audio channel so far, for example, the audio channel of the main broadcasting program to the audio channel of the traffic infor-

mation is consequently executed. After that, the decoding process and output of the decoded signal, of the traffic information audio signal are executed until the reproduction output control signal is set to the low level.

When the reproduction output control signal is at the low level, the audio decoder 40 restarts the decoding process with respect to the audio channel which has been decoding processed just before the decoding process of the audio channel of the traffic information is executed.

In addition to the reproduction output control signal described here, the audio decoder 40 inherently has a function for decoding and switching according to an input from an operating unit provided for the receiver. That is, the input data signal of the audio decoder 40 has a plurality of audio channels. So long as an audio decoder 40 with a single construction, the decoding process is ordinarily executed with regard to the audio channel which is designated in the station selection by the user. Various modes can be selected by the operating unit. For instance, it is also possible to construct in a manner such that when a mode to preferentially reproduce and output the traffic information is designated, the audio decoder 40 automatically switches from the audio channel of the main broadcasting program to the audio channel of the traffic information in accordance with the reproduction output control signal as mentioned above and that if this mode is not designated, only when a station selecting instruction of the audio channel of the traffic information is issued from the user, the channel can be also switched to the relevant traffic information channel irrespective of the reproduction output control signal.

When a data error of a corresponding digital additional information signal (signal excluding the subband sampling signal, refer to Fig. 2) is detected by the header CRC or ScF-CRC, the audio decoder 40 itself mutes the audio signal of the decoding output corresponding to the detected data error.

The D/A converter 43 converts the audio signal which was decoded and generated as mentioned above to an analog signal. The resultant analog audio signal drives a speaker 45 through an amplifier 44 and an acoustic sound is reproduced.

Even on the output side of the data decoder 41, an audio reproduction system similar to the audio decoder 40 can be constructed. It is, however, now assumed that information to be treated by the data decoder 41 includes not only an audio signal but also an image or the like and its construction is not described in detail here because its application is wide.

The control part 30 is made up of, for example, a microcomputer. Although explanation has been made here on the assumption that the control part 30 performs the station selection for the front-end 32 and the control of the audio decoder 40, the control part 30 executes not only the input control from the operating unit and the mode control but also various controls of the

whole receiver. It will be also obviously understood that the control part 30 can execute various controls or modes other than those mentioned above for the other structural blocks in cooperation with the operating unit or the input means. Their detailed description is omitted here.

Although it is assumed above that the control part 30 has the functional blocks 3A to 3C as a construction of hardware, this construction can be actually easily realized as software of the microcomputer. Fig. 5 shows a processing procedure for the reproduction output control which is executed by the control part 30 as will be explained in detail hereinbelow.

In Fig. 5, the control part 30 calls the reproduction output control processing routine at the start of the reception or during the operation. As an initial step of this processing routine, the control part 30 clears a counter N indicative of the number of audio frames (refer to Fig. 2) and a counter n showing the number of times of occurrence of errors for the purpose of initialization (step S1). The control part 30 discriminates whether the header detection information to notify of the arrival of the head of the audio frames has been received from the audio decoder 40 or not (step S2). When it is not received, this discrimination is continued. When it is received, the count value of the frame counter N is increased by 1 (step S3).

After step S3, the control part 30 discriminates whether the error detection information based on the header CRC or ScF-CRC has been sent from the audio decoder 40 or not (step S4). If there is the error detection information, the count value of the error counter n is increased by 1 (step S5). Step S6 follows. When there is not the error detection information, the processing routine immediately advances from step S4 to step S6. The audio decoder 40 generates the error detection information every audio frame and resets the error detection information at the end of one audio frame.

In step S6, a check is made to see if the value of the frame counter N has reached a predetermined value Nm. The predetermined value Nm denotes the number of frames necessary for an arithmetic operating process of the error rate in this routine and can be properly set to an integer of 2 or more. When $N \neq Nm$ in step S6, the processing routine advances to step S2 mentioned above. When $N = Nm$, n is divided by N and the division result is stored into a register X (step S7). Since X denotes (the number of audio frames whose errors were detected: n)/(a predetermined number of audio frames: N), it corresponds to the error rate.

A flow in steps S2 to S6 will now be described in detail. The error detection discrimination in step S4 is executed by only the number of times corresponding to the Nm audio frame, namely, only Nm times. In step S5, how many times the error detection has been discriminated in the error detecting discriminations of Nm times is counted in step S5. Therefore, the flow in steps S2 to S6 can be said in other words that a process for count-

ing the number of times of occurrence of errors in the Nm continuous audio frames. An error rate X is, therefore, inevitably obtained every Nm continuous audio frames.

When the error rate X is calculated in this manner, the control part 30 discriminates whether the value of X is smaller than a predetermined value Xth or not (step S8). When the value of X is smaller than the predetermined value Xth in step S8, a check is made to see if the ASW information in the FIC data signal from the signal distributing circuit 39 indicates that the traffic information is being provided (namely, whether the ASW is ON or not) (step S9). When the ASW is ON, the control unit generates a signal to control the audio decoder 40 so as to perform the decoding process and decoding output of the traffic information audio signal (step S10). When the value of X is equal to or larger than the predetermined value Xth in step S8, therefore, when it is possible to decide that the error rate of the input data signal of the audio decoder 40 is fairly large, the control unit generates a signal to control the audio decoder 40 so as to perform the decoding process and decoding output of the main broadcasting program audio signal irrespective of the ASW information (step S1A). In step S1A, the main broadcasting program as a target of reproduction can be made correspond to the channel selected by the user as mentioned above or the channel of a predetermined default.

Even when the ASW is determined to be OFF in step S9, the processing routine advances to step S1A and the main broadcasting program audio signal is reproduced.

As mentioned above, in the embodiment, the channel is switched to the audio channel of the traffic information under the conditions such that not only the ASW is ON but also the error amount due to the CRC of the input data signal of the audio decoder 40 is sufficiently small. That is, the reproduction output of the traffic information audio signal is permitted only when a situation that is presumed such that large errors by the CRC will not be detected is discriminated. In the embodiment, therefore, different from the prior art in which the channel is soon switched to the audio channel of the traffic information when the ASW is turned on, in a situation that many errors by the CRC are detected, even if the ASW is turned on, the channel is not switched to the audio channel of the traffic information. It is, therefore, prevented that the system enters a soundless state due to the audio muting of the CRC detection error response or an intermittent audio output state after the switching to the channel, so that an audio sound which gives an unpleasant feeling to the listener can be suppressed.

The second embodiment according to the invention will now be described.

Fig. 6 shows a schematic construction of a DAB receiver of the second embodiment and portions similar to those in Fig. 4 are designated by the same reference numerals.

In Fig. 6, different from the construction of Fig. 4, the reproduction output control process is executed without using the error detection information and the header detection information from the audio decoder 40. For this purpose, a control part 30a has means for

generating a reproduction output control signal on the basis of the input/output signal of the error corrector 38. Explaining in detail, an output signal of the T/F base processor 37 is a convolutionally coded signal and is supplied to buffering means 3D provided in the control part 30a. The error corrector 38 performs the decoding of the convolutional coding. Reencoding means 3E provided in the control part 30a again convolution encodes a Viterbi decoding output in which the convolution encoding was decoded and to which an error correction was performed by the error corrector, thereby forming a convolution encoding signal showing the same value as that of the input of the error corrector 38. That is, the corrected convolution encoding signal is derived from the reencoding means 3E. The buffering means 3D fetches the convolution encoding signal from the T/F base processor 37, delays it by only a processing time that is required in the error corrector 38 and reencoding means 3E, and supplies the delayed signal to comparing means 3F. The corrected convolution encoding signal from the reencoding means 3E is also supplied to the comparing means 3F. The comparing means 3F compares both of the input signals. The comparing process includes a process for discriminating the coincidence/dissidence every bit. An estimation error rate signal according to the number of dissident bits [or a ratio (n/m) of a number n of times of dissidence to a predetermined discrimination number m of bits] is generated. The estimation error rate signal is supplied to evaluating means 3G. Only when the error rate signal is smaller than a predetermined value, the evaluating means 3G generates an error rate evaluation signal at the high level and supplies it to the AND gate 3C. The ASW information signal from the FIC decoder 3A is supplied to the other input of the AND gate 3C in a manner similar to Fig. 4.

Only when the number of dissident bits which is detected by the comparing means 3F is sufficiently small, therefore, the control part 30a permits the reproducing operation of the traffic information channel of the audio decoder 40 in response to the ASW.

The execution of the comparison of both signals is accomplished by the buffering means 3D with respect to the same sampling period. The control part 30a receives the FIC data signal from the signal distributing circuit 39 by the FIC decoder 3A. Code rate information as a decoding result of the FIC data signal by the FIC decoder 3A is also supplied to the reencoding means 3E. The reencoding means 3E also executes a punctured process according to the code rate information.

The punctured process is performed in the channel coder in the transmission system as mentioned above. The signal which is inputted to the error corrector 38 in

the reception system, therefore, has a form such that the bits as many as only the number corresponding to the code rate were extracted by the punctured process. The error corrector 38 itself detects the data regarding the code rate in the FIC data signal and identifies code rate data including the extraction information showing which bits were extracted from the detection data. The error corrector 38 performs a decoding process to the bit-extracted input signal (punctured signal) so as to be adapted to the extraction information. More specifically speaking, bits indicative of an undefined value are allocated to the extraction bits shown by the extraction information, a pure convolution encoding signal (corresponding to the signal subjected to the convolution encoding which is performed before the punctured process in the channel coder 2 in Fig. 1) is obtained, and a Viterbi decoding is performed to this signal.

As mentioned above, the error corrector 38 executes the decoding process while performing what is called an inverse punctured process adapted to the input data by itself. The reencoding means 3E, therefore, again executes the punctured process adapted to the code rate data in the FIC data signal by using this data so as to match with the punctured signal from the T/F base processor 37 (buffering means 3D), thereby enabling the comparison between the same punctured signals to be performed in the comparing means 3F.

The comparing means 3F uses the reencoding signal from the reencoding means 3E as a comparison reference and generates a difference between the comparison reference and the encoding signal from the buffering means 3D as an error. That is, the comparing means 3F generates an error on the assumption that the reencoding signal is a signal subjected to a proper error correction and is a signal of a correct value. If the error corrector 38 itself, therefore, executes an improper correcting process to the input signal, the comparison reference becomes improper and a resultant error is also unreliable. In the embodiment, therefore, the estimation error rate signal and the reproduction output control signal are generated on the basis of the assumption as a prerequisite.

Although it is assumed that the control part 30a has the functional blocks 3A and 3C to 3G as a construction of hardware in the above embodiment, this construction can be actually easily realized as software of a micro-computer. Fig. 7 shows a processing procedure for the reproduction output control which is executed by the control part 30a in this case and will be explained in detail hereinafter.

In Fig. 7, the control part 30a calls the reproduction output control processing routine at the start of the reception or during the operation. As an initial process, the control part 30a clears the counter M to count the number of bits to be compared and the counter n showing the number of times of occurrence of the bit errors for the purpose of initialization (step S11). The control part 30a fetches convolution encoding signals A (1, 2,

..., m) as many as m bits from the T/F base processor 37 (step S12). m indicates a predetermined number of bits to be compared in the subsequent processing steps and A(x) denotes a signal of the x-th bit from, for example, the head bit among the m bits in the convolution encoding signals.

When the input convolution encoding signals are fetched, the control part 30a subsequently generates convolution encoding signals B to be compared therewith (step S13). More specifically speaking, the control part 30a fetches error corrected decoding signals B' (1, 2, ..., m') which correspond to the convolution encoding signals A (1, 2, ..., m) and were outputted from the error corrector 38 and executes a convolution encoding similar to that is executed by the reencoding means 3E to the signals B'. The control part 30a further executes the punctured process according to the information concerning the code rate in the FIC data signal as mentioned above to the signals B', thereby obtaining the final corrected convolution encoding signals B (1, 2, ..., m).

The control part 30a which generated the corrected convolution encoding signals B as mentioned above counts up the bit counter M by 1 (step S14).

After step S14, the control part 30a discriminates whether the input encoding signal A and the corrected encoding signal B coincide between the bits indicated by the counter M or not (step S15). When they don't coincide, the bit error counter n is counted up by 1 (step S16) and step S17 follows. When they coincide, the processing routine soon advances to step S17 from step S15.

In step S17, a check is made to see if the count value of the bit counter M has reached the predetermined number m of bits (namely, the number of bits of the signals A and B held in steps S12 and S13). The predetermined number m of bits is the number of bits necessary for the arithmetic operating process of the error rate in the present routine and can be properly set to an integer of 2 or more. In step S17, if $M \neq m$, step S14 follows. When $M = m$, n is divided by m and the division result is stored into a register X' (step S18). Since X' denotes [(the number of bits which were decided to be dissident (errors): n)/(the predetermined discrimination number of bits: m)], it corresponds to the error rate.

A processing flow in steps S14 to S17 will now be described in detail. The discrimination about the dissidence in step S15 is executed the number of times corresponding to only m bits, namely, only m times. The number of times of dissidence discrimination among the m discriminating times is counted in step S16. The processing flow in steps S14 to S17, therefore, can be said in other words such that a process to count the number of bit errors occurred between both of the encoding signals A and B of the m continuous bits is executed. The error rate X', consequently, is inevitably obtained every m continuous bits.

When the error rate X' is calculated as mentioned

above, the control part 30a discriminates whether the value of X' is smaller than a predetermined value X'th or not (step S19). When the value of X' is smaller than the predetermined value X'th in step S19, a check is made to see if the ASW information in the FIC data signal from the signal distributing circuit 39 indicates that the traffic information is being provided (namely, whether the ASW is ON or not) (step S20). When the ASW is ON, the control part 30a generates a signal to control the audio decoder 40 so as to perform the decoding process and the decoding output of the traffic information audio signal (step S21). On the other hand, when the value of X' is equal to or larger than the predetermined value X'th in step S19, therefore, when the error rate of the input data signal of the error corrector 38 is determined to be fairly large, the control part generates a signal to control the audio decoder 40 so as to perform the decoding process and the decoding output of the main broadcasting program audio signal irrespective of the ASW information (step S22). In step S22 as well, the reproduction output can be performed with respect to the channel selected by the user or the channel of a predetermined default.

Even when the ASW is decided to be OFF in step S20, the processing routine advances to step S22 and the main broadcasting program audio signal is reproduced.

In the embodiment, the channel is switched to the audio channel of the traffic information under the conditions such that not only the ASW is ON but also the error amount (X') of the input data signal of the error corrector 38 is sufficiently small. A fact that the estimated error rate X' is large denotes that when the audio data signal received at that time is reproduced, an audio sound containing much noise components is derived. In this situation, the operation such that even if the ASW is turned on, the audio output of the traffic information is not performed is accomplished. Although no CRC is used here, the reproduction output of the traffic information audio signal is permitted only when a situation where many errors of the input data signal of the error corrector 38 will not be detected is discriminated. In the embodiment as well, therefore, in a situation where the received data signal cannot be sufficiently desirably reproduced, even if the ASW is turned on, the channel is not switched to the audio channel of the traffic information. It is, therefore, prevented that the system enters a noise-like audio output state after the switching to the channel, thereby making it possible to suppress an audio sound which gives an unpleasant feeling to the listener.

An embodiment for further certainly preventing a noise audio sound or a soundless state after the switching to the traffic information channel by combining the first and second embodiments can be also further realized.

Fig. 8 shows a schematic construction of a DAB receiver according to the third embodiment and portions

similar to those in Figs. 4 and 6 are designated by the same reference numerals.

In Fig. 8, a control part 30b has the first error rate calculating evaluating means 3B. The first error rate calculating evaluating means 3B generates a first error evaluation signal showing a quality evaluation result of the error rate X every Nm unit frames on the basis of the error detection information and the header detection information by the header CRC or ScF-CRC from the audio decoder 40 in a manner similar to the foregoing processes shown in Fig. 4.

On the other hand, the means 3D to 3G used in the control part 30a in Fig. 6 also execute the processes shown in Fig. 7 in cooperation with each other and generates a second error evaluation signal based on the input/output of the error corrector 38. The second error evaluation signal shows a quality evaluation result of the error rate X' every m bits.

The control part 30b also uses a 3-input AND gate 3C' in place of the 2-input AND gate. In addition to the ASW data signal, the AND gate 3C' receives the first and second error evaluation signals and sets the reproduction output control signal to the high level when all of those signals are significant, namely, at the high level, thereby designating the reproducing mode of the traffic information channel to the audio decoder 40.

Even in the reproduction output control which is accomplished by the control part 30b, in a situation where the received data signal cannot be sufficiently desirably reproduced, even if the ASW is turned on, the channel is not switched to the audio channel of the traffic information, so that it is prevented that the system enters the noise-like audio output state after the switching to the channel. Since the two error evaluation signals are used, the above preventing effect is further improved.

The control part 30b can be realized by software by properly combining the necessary processing steps by a person with ordinary skill in the art by referring to the flowcharts of Figs. 5 and 7 described already in detail above.

Although the embodiment has mainly been described above with respect to the switching control between the station selection state of the main broadcasting program and the station selection state of the traffic information program, the invention is not limited to the main broadcasting program and the traffic information program but can be applied to other controls. That is, the invention can be applied to a control when switching from a certain station selection state to another specific station selection state. The switching source is also not limited to the station selection state.

Fig. 9 will now be explained as a simple example in the case of a reproducing state from another source in which the switching source is different from the receiver.

Fig. 9 shows a partial schematic construction of an audio or audio/visual system in which both of a DAB receiver 51 and a disc player 52 are compatible and

execute an acoustic reproduction as mentioned above.

In Fig. 9, the reproduction output control signal which is obtained as mentioned above is generated to the outside and is used as a selection control signal of a selector 53 for selectively generating an analog audio output signal of the receiver 51 and an analog audio output signal of the disc player 52. The selector 53 relays either selected one of the audio output signals to a driving amplifier 54. The driving amplifier 54 drives a speaker 55 in accordance with the relayed audio output signal.

According to the construction, when the receiver 51 detects that the ASW is turned on and a state where a good reproduction output can be derived during the acoustic output by the disc player 52, the reproduction output control signal is generated. The selector 53, therefore, supplies the audio output signal from the receiver 51 to the driving amplifier 54 instead of the audio output signal from the disc player 52 so far in response to the generated reproduction output signal. When a state where the reproduction output control signal is not generated is further detected, the selector 53 again selects and generates the audio output signal from the disc player 52. The user of the system, accordingly, can desirably listen to the audio sound of the traffic information through an interruption or with preference, during the reproduction of music by the disc player 52.

The invention is not limited to the disc player but what is called an ordinary FM radio broadcasting tuner can be also used as a switching target and various forms are considered.

According to the form of realizing the invention by the software in each of the embodiments, the on/off state of the ASW is discriminated after the error rate or estimation error rate was evaluated. On the contrary, however, the error rate or estimation error rate can be also evaluated after the on/off state of the ASW was discriminated.

Although each of the embodiments has been described on the assumption that the switching destination is the traffic information channel, the upper concept of the invention doesn't limit it. That is, for example, a channel of a weather forecast or emergency news information can be also used as a switching destination instead of the traffic information channel. The switching destination is also not limited to the audio channel. So long as a general data system using the data decoder 41 is used, switching destinations of characters, images, or other various information channels can be established.

Although each of the embodiments has been described with respect to the specific numerical values of Nm, m, and the other parameters, it will be obviously understood that the invention is not limited to those values.

Although each of the embodiments has been described with respect to the receiver adapted to the

specific DAB system and the system using the receiver, the invention is fundamentally not restricted to only the DAB system. In brief, the invention can be applied to every system having the system for receiving the digital audio signal and the identification signal and is effective to a data format having a possibility such that a demodulating sensitivity of the identification signal is higher than a demodulating sensitivity of the digital audio signal. Particularly, in the embodiments, it can be said that the form of presuming the errors of the data signal on the basis of the input/output of the error corrector 38 is extremely effective for a digital signal of a format having no error detection signal.

Furthermore, although the embodiments have been described while limiting various means, they can be also properly modified within the purview where they can be designed by those with ordinary skill in the art.

According to the invention as described in detail above, the information data signal receiver which can desirably reproduce the specific information in only a situation where the specific information can be truly reproduced in the reception system can be provided. In the system in which the specific data signal (also including the additional information signal such as a traffic information audio data signal or the like) to be information reproduced and the identification data signal to identify that the specific data signal is validated are transmitted by the transmission wave of the same frequency band, even if the code rate of the specific data signal and the code rate of the identification data signal are different, the specific data signal can be desirably information reproduced at a proper timing.

Embodiments of a digital audio signal receiver according to the present invention will now be described with reference to Figs. 10 to 23.

Fig. 10 schematically shows a construction of a DAB receiver as an embodiment of the digital audio signal receiver according to the invention.

The explanation of portions the same as or similar to those portions of the DAB receiver shown in Fig. 4 will not be repeated.

The data signal which is supplied to the audio decoder 40 has the format in which the audio frames are used as a unit as shown in Fig. 2 mentioned above. The audio decoder 40 decodes the data signal so as to be reconstructed to the original digital audio signal, supplies a decoding output as a decoding result to a digital attenuator 42, and transfers the error detection information based on the error information which is derived during the decoding, namely, at least one of the header CRC and the ScF-CRC to the control part 30. A header detecting process to recognize the start of the audio frame is included in the decoding process of the audio decoder 40. The header detection information which is obtained by the header detecting process is also transferred to the control part 30.

The control part 30 receives the error detection information and the header detection information, exe-

cutes a predetermined arithmetic operating process based on those information, and supplies a muting control signal according to the arithmetic operation result to the digital attenuator 42. The digital attenuator 42 attenuates the signal from the audio decoder 40 by an attenuation amount according to the muting control signal by a digital process and supplies the attenuated digital audio signal to the D/A (digital-analog) converter 43. The D/A converter 43 converts the input digital audio signal to an analog signal. The resultant analog audio signal drives the speaker 45 through the amplifier 44, thereby reproducing an acoustic sound.

An audio reproduction system similar to that of the audio decoder 40 can be also constructed on the output side of the data decoder 41. Information to be treated by the data decoder 41, however, includes not only the audio signal but also images or the like and it is assumed that its construction is not described in detail here because its application is wide.

The control part 30 is constructed by, for example, a microcomputer. Although explanation has been made here on the assumption that the control part 30 performs the station selection for the front-end 32 and the control of the digital attenuator 42, the control part 30 executes other various controls of the whole receiver. The control part 30 can also obviously execute various controls or modes for the other constructing blocks in cooperation with an operation or input means (not shown). The details are omitted here.

The muting control process which is executed by the control part 30 will now be described in detail with reference to a flowchart of Fig. 11.

In Fig. 11, the control part 30 calls the muting output control processing routine at the start of the reception or during the operation. As an initial step of this processing routine, the control part 30 clears the counter N indicative of the number of audio frames (refer to Fig. 2) and the counter n showing the number of times of occurrence of errors for the purpose of initialization (step S1). The control part 30 discriminates whether the header detection information to notify of the arrival of the head of the audio frames has been received from the audio decoder 40 or not (step S2). When it is not received, this discrimination is continued. When it is received, the count value of the frame counter N is increased by 1 (step S3).

After step S3, the control part 30 discriminates whether the error detection information based on the header CRC or ScF-CRC has been sent from the audio decoder 40 or not (step S4). If there is the error detection information, the count value of the error counter n is increased by 1 (step S5). Step S6 follows. When there is not the error detection information, the processing routine immediately advances from step S4 to step S6. The audio decoder 40 generates the error detection information every audio frame and resets the error detection information at the end of one audio frame.

In step S6, a check is made to see if the value of the

frame counter N has reached the predetermined value Nm. The predetermined value Nm denotes the number of frames necessary for an arithmetic operating process of the error rate in this routine and can be properly set to an integer of 2 or more. When $N \neq Nm$ (or $N < Nm$) in step S6, the processing routine advances to step S2 mentioned above. When $N = Nm$, n is divided by N and the division result is stored into a register X (step S7). Since X denotes (the number of audio frames whose errors were detected: n)/(a predetermined number of audio frames: N), it corresponds to the error rate.

A processing flow in steps S2 to S6 will now be described in detail. The discrimination about the error detection in step S4 is executed the number of times corresponding to only Nm audio frames, namely, only Nm times. The number of times of error detection discrimination among the Nm discriminating times is counted in step S5. The processing flow in steps S2 to S6, therefore, can be said in other words such that a process to count the number of times of errors occurred among the Nm continuous audio frames is executed. The error rate X, consequently, is inevitably obtained every Nm continuous audio frames.

When the error rate X is calculated as mentioned above, the control part 30 generates a muting control signal according to the value of X and supplies it to the digital attenuator 42 (step S8). An attenuation amount is set in the digital attenuator 42 in accordance with the muting control signal. By passing through the digital attenuator 42, the decoded digital audio signal from the audio decoder 40 becomes a signal having the maximum permission level or dynamic range according to the error rate X.

Although "soft mute" such that the output audio sound is controlled by the muting level (or muting amount) according to the error rate X is accomplished, a detailed format of the soft mute in the embodiment is as follows.

That is, separately from the muting control process of Fig. 11, the audio decoder 40 itself executes a "full mute" (it is possible to consider to fix the level of the output audio signal to 0) to the output audio signal in response to the error detection based on the header CRC or ScF-CRC. Since the "full mute" control is executed every audio frame, the output audio signal is turned on/off in an extremely short divided time. If the output audio signal of the audio decoder 40 is acoustically reproduced as it is, particularly, in the case where the error detection and the non-detection are almost alternately repeated every frame or the like, a noisy sound such that it is intermittent at a high frequency is generated.

Fig. 12B conceptually shows the above situation. In Fig. 12B, in correspondence to each frame of the audio frame series, the audio decoder 40 full-mutes the audio signal which is generated when the error is detected on the basis of the header CRC or ScF-CRC and generates the audio signal at a fixed dynamic range when no

error is detected. The frame numbers 1, 2, ... allocated to the audio frame series correspond to the numbers (1), (2), ... allocated to the audio decoder output and attenuator output series, which will be explained hereinafter.

Fig. 12C shows a format of the audio signal by the muting control process in the embodiment, namely, the audio signal from the attenuator 42. An example in which a period of the muting control process is set to five frames is mentioned here. At the first period (#1) shown in Fig. 12A, since the error based on the header CRC or ScF-CRC is not detected at a preceding period, the attenuator 42 transmits the audio signal from the audio decoder 40 without substantially changing the dynamic range.

At a period (#2), since the error based on the header CRC or ScF-CRC has been detected twice at the preceding period (#1), the audio signal from the audio decoder 40 is soft-muted at the muting level according to the error rate of $X = 2/5$. As will be also understood from Figs. 12A to 12C, the signal is soft-muted substantially in only an off-muting period of time in the audio decoder output and is held as it is for a full-muting period of time.

Even at a period (#3), since the error based on the header CRC or ScF-CRC has similarly been detected three times at the preceding period (#2), the audio signal from the audio decoder 40 is soft-muted at the muting level according to the error rate of $X = 3/5$ and is relatively higher than the previous one. The signal is also soft-muted substantially in an off-muting period of time in the audio decoder output.

As will be understood from the above description, according to the embodiment, as the number of times of error detection increases, the muting level is increased and the level of the audio signal from the audio decoder 40 is suppressed to a low level. An influence by an audio sound having a feeling of physical disorder such that it is intermittent at a high frequency can be reduced in accordance with an extent of the low level.

The second embodiment according to the invention will now be described.

Fig. 13 shows a schematic construction of a DAB receiver of the second embodiment and portions similar to those in Fig. 10 are designated by the same reference numerals.

In Fig. 13, different from the construction of Fig. 10, the muting control process is executed without using the error detection information from the audio decoder 40. For this purpose, means for generating the muting control signal on the basis of the input/output signal of the error corrector 38 and output start information from the audio decoder 40 is provided for the control part 30a. The output start information shows the timing at which the decoder 40 finishes the decoding process of one frame and starts the decoding output of the frame.

An output signal of the T/F base processor 37 is the convolution encoded signal and is supplied to the buff-

ering means 3D provided in the control part 30a. The error corrector 38 decodes the convolution encoding. The reencoding means 3E provided in the control part 30a again convolution encodes the Viterbi decoding output in which the convolution encoding was decoded and the error correction was performed by the corrector, thereby generating the convolution encoding signal showing the same value as that of the input of the error corrector 38. That is, the corrected convolution encoded signal is derived from the reencoding means 3E. The buffering means 3D fetches the convolution encoding signal from the T/F base processor 37, delays it by only the processing time which is required for the error corrector 38 and reencoding means 3E, and supplies the delayed signal to the comparing means 3F. The corrected convolution encoding signal from the reencoding means 3E is supplied to the comparing means 3F. The comparing means compares both of the input signals. The comparing process includes a process for discriminating the coincidence/dissidence every bit and the muting control signal according to the number of times of dissidence [or the ratio (n/m) of the number n of times of dissidence to the predetermined discrimination number m of bits] is generated. Output start information from the audio decoder 40 is also supplied to the comparing means 3F. On the basis of this information, the comparing means performs an output control of the muting control signal to the digital attenuator 42.

The digital attenuator 42 increases its attenuation amount as the number of times of occurrence of bit dissidence is large, reduces the dynamic range of the audio signal from the audio decoder 40, and supplies the resultant signal to the D/A converter 43.

The execution of the comparison of both of the signals is accomplished by the buffering means 3D with respect to the same sampling timing. The control part 30a receives the FIC data signal from the signal distributing circuit 39 by the FIC decoder 3A. Code rate information as a decoding result of the FIC data signal by the FIC decoder 3A is also supplied to the reencoding means 3E. The reencoding means 3E also executes a punctured process according to the code rate information.

The punctured process is performed in the channel coder in the transmission system as mentioned above. The signal which is inputted to the error corrector 38 in the reception system, therefore, has a form such that the bits as many as only the number corresponding to the code rate were extracted by the punctured process. The error corrector 38 itself detects the data regarding the code rate in the FIC data signal and identifies code rate data including the extraction information showing which bits were extracted from the detection data. The error corrector 38 performs a decoding process to the bit-extracted input signal (punctured signal) so as to be adapted to the extraction information. More specifically speaking, bits indicative of an undefined value are allocated to the extraction bits shown by the extraction infor-

mation, a pure convolution encoding signal (corresponding to the signal subjected to the convolution encoding which is performed before the punctured process in the channel coder 2 in Fig. 1) is obtained, and a Viterbi decoding is performed to this signal.

As mentioned above, the error corrector 38 executes the decoding process while performing what is called an inverse punctured process adapted to the input data by itself. The reencoding means 3E, therefore, again executes the punctured process adapted to the code rate data in the FIC data signal by using this data so as to match with the punctured signal from the T/F base processor 37 (buffering means 3D), thereby enabling the comparison between the same punctured signals to be performed in the comparing means 3F.

The comparing means 3F uses the reencoding signal from the reencoding means 3E as a comparison reference and generates a difference between the comparison reference and the encoding signal from the buffering means 3D as an error. That is, the comparing means 3F generates an error on the assumption that the reencoding signal is a signal subjected to a proper error correction and is a signal of a correct value. If the error corrector 38 itself, therefore, executes an improper correcting process to the input signal, the comparison reference becomes improper and a resultant error is also unreliable. In the embodiment, therefore, the muting control signal is generated on the basis of the assumption as a prerequisite.

Although it is assumed that the control part 30a has the functional blocks 3A to 3D as a construction of hardware, this construction can be actually easily realized as software of a microcomputer. Figs. 14 and 15 show a processing procedure for the muting control which is executed by the control part 30a in this case and will be explained in detail hereinafter.

In Figs. 14 and 15, the control part 30a calls the muting control processing routine at the start of the reception or during the operation. As an initial process, the control part 30a clears the counter M to count the number of bits to be compared and the counter n showing the number of times of occurrence of the bit errors for the purpose of initialization (step S11). The control part 30a fetches convolution encoding signals A (1, 2, ..., m) as many as m bits from the T/F base processor 37 (step S12). "m" indicates the predetermined number of bits to be compared in the subsequent processing steps and A(x) denotes the signal of the x-th bit from, for example, the head bit among the m bits in the convolution encoding signals.

When the input convolution encoding signals are fetched, the control part 30a subsequently generates convolution encoding signals B to be compared therewith (step S13). More specifically speaking, the control part 30a fetches error corrected decoding signals B' (1, 2, ..., m') which correspond to the convolution encoding signals A (1, 2, ..., m) and were outputted from the error corrector 38 and executes a convolution encoding simi-

lar to that is executed by the reencoding means 3E to the signals B'. The control part 30a further executes the punctured process according to the information concerning the code rate in the FIC data signal as mentioned above to the signals B', thereby obtaining the final corrected convolution encoding signals B (1, 2, ..., m).

The control part 30a which generated the corrected convolution encoding signals B as mentioned above counts up the bit counter M by 1 (step S14).

After step S14, the control part 30a discriminates whether the input encoding signal A and the corrected encoding signal B coincides between the bits indicated by the counter M or not (step S15). When they don't coincide, the bit error counter n is counted up by 1 (step S16) and step S17 follows. When they coincide, the processing routine soon advances to step S17 from step S15.

In step S17, a check is made to see if the count value of the bit counter M has reached the predetermined number m of bits (namely, the number of bits of the signals A and B held in steps S12 and S13). The predetermined number m of bits is the number of bits necessary for the arithmetic operating process of the error rate in the present routine and is properly set to an integer of 2 or more. In step S17, if $M \neq m$ (or $M < m$), the processing routine is returned to step S14. When $M = m$, n is divided by m and the division result is stored into a register X_j (step S18). Since X_j denotes [(the number of bits which were decided to be dissident (errors): n)/(the predetermined discrimination number of bits: m)], it corresponds to the error rate.

A processing flow in steps S14 to S17 will now be described in detail. The discrimination about the dissidence in step S15 is executed the number of times corresponding to only m bits, namely, only m times. The number of times of dissidence discrimination among the m discriminating times is counted in step S16. The processing flow in steps S14 to S17, therefore, can be said in other words such that a process to count the number of bit errors occurred between both of the encoding signals A and B of the m continuous bits is executed. The error rate X_j, consequently, is inevitably obtained every m continuous bits.

When the value of j is equal to 1 in this instance, it is obtained as a first error rate X₁. Similarly, as the value of j increases, it is obtained as error rates X₂, X₃,

After step S18, a check is made to see if the value of j is equal to a predetermined value J (step S19). When they are not equal, namely, when $j < J$, the counter j is counted up by 1 (step S20). A number of the error rate X_j to be obtained has been set to J. After j was set to a new value in step S20, the processes in steps S12 to S18 are again executed and the error rate X_j is obtained with respect to the new value of j.

When $j = J$ is discriminated in step S19, the average of the error rates X₁, X₂, ..., X_j obtained so far is calculated and the resultant average value is stored into the

register X' (step S21). The error rate of all of the continuous (m x J) bits is obtained.

The reason why J error rates are obtained every m bits and one error rate is finally derived in this manner is to suppress a memory capacity to store each sampling data of the convolution encoding signals A and B. In the case of soon deriving the error rates of (m x J) bits, a register to previously fetch the signals A (1, 2, ..., m x J) and B (1, 2, ..., m x J) is necessary.

When the error rate X' is calculated in this manner, the control part 30a confirms the start of the output of the decoder 40 (step S22), generates a muting control signal according to the value of X', and transmits it to the digital attenuator 42 (step S23). An attenuation amount according to the muting control signal is set into the digital attenuator 42. By passing through the digital attenuator 42, the decoded digital audio signal from the audio decoder 40 becomes a signal having the maximum permission level or dynamic range according to the error rate X'.

In this manner the "soft mute" to control the output audio signal at the muting level according to the error rate X' is accomplished. A soft-muting operation in the embodiment has a fairly high response speed as compared with that in the first embodiment.

That is, as will be also understood from Figs. 12A to 12C, in the first embodiment, since the CRC error which is obtained every frame is used, the error rate is derived for the first time by a plurality of frames, namely, Nm frames. Therefore, the "soft mute" is also performed on a plural frame unit basis and the generated CRC error is reflected to the actual "soft mute" after the elapse of at worst Nm-1 frames from the frame where the error occurred. In the second embodiment, on the other hand, since the bit error which is obtained every bit is used, the error rate is obtained by only the predetermined number m of bits or (m x J) bits in the frame. According to the second embodiment, therefore, the "soft mute" can be performed every frame and the generated bit error can be reflected to the actual "soft mute" in the present frame.

A muting format in the second embodiment can be conceptually shown in Figs. 16A to 16C.

Even in Figs. 16A to 16C, the output format of the attenuator 42 is made correspond to the audio frame series and the output of the audio decoder 40 and is shown in a manner similar to Figs. 12A to 12C. A point that the audio decoder output is subjected to the "full mute" of the output audio signal by itself in response to the error detection based on the header CRC or ScF-CRC different from the muting control process is the same as the first embodiment. A point that the "soft mute" is valid for the off muting period of time of the audio decoder output is also the same.

The "soft mute" according to the second embodiment, however, can be more finely realized than the first embodiment because it is performed every frame. That is, it is not the "soft mute" responsive to the error rate

which is obtained for five frames as shown in Figs. 12A to 12C but the muting level responsive to the error rate which is derived in only one frame is set, so that a response speed is obviously high.

A good result is obtained by using a value about 1000 bits as a value of m . The case of $J = 5$ is shown in Figs. 16A to 16C and a state in which after the error rate X' was calculated, the "soft mute" is performed in response to the decoding output start timing of the frame corresponding to the error rate will be understood.

In the second embodiment as well, as the number of times of detection of error, namely, dissident bits increases, the muting level is raised and the level of the audio signal from the audio decoder 40 is suppressed to a low level. An influence, therefore, by the audio sound having a feeling of physical disorder such that it is intermittent at a high frequency can be reduced in accordance with its degree. In the second embodiment, an equivalent error is detected irrespective of the CRC code and the "soft mute" according to it is performed, so that a state where the "soft mute" is desirably performed even to the subband sampling signal to which no CRC code is added can be realized. A conventional inconvenience, therefore, such that the noise acoustic output of the subband sampling signal is conspicuous in the case where the error detection is not performed by the header CRC and ScF-CRC is suppressed.

An embodiment obtained by combining the first and second embodiments can be also realized.

Fig. 17 shows a schematic construction of a DAB receiver according to the third embodiment and portions similar to those in Figs. 10 and 13 are designated by the same reference numerals.

In Fig. 17, the control part 30b has first muting control means 3H. In a manner similar to the processes shown in Fig. 11 mentioned above, the first muting control means 3H generates a first muting control signal (X) to designate the muting level every Nm unit frames on the basis of the error detection information and the header detection information by the header CRC or ScF-CRC from the audio decoder 40.

The means 3A to 3D used in the control part 30a in Fig. 13 also execute the processes shown in Figs. 14 and 15 in cooperation with each other and generate a second muting control signal (X') based on the input/output of the error corrector 38. The second muting control signal designates the muting level every frame.

The control part 30b has muting level adding means 3I. The adding means 3I adds the first and second muting control signals to the muting levels and generates a final muting control signal according to the addition result. The muting control signal formed in this manner is supplied to the digital attenuator 42.

A muting format which is accomplished by the control part 30b is shown in Figs. 18A to 18C. According to Figs. 18A to 18C, it will be understood that for the mut-

ing off period of time of the audio decoder output, the muting operation of the 5-frame period by the first muting control signal and the muting operation of the 1-frame period by the second muting control signal are simultaneously function together. It should be noted that at the period (#1) of the first muting control process, the "soft mute" is not performed in the format of Figs. 12A to 12C but the "soft mute" by the second muting control is executed in the format of the embodiment.

Another embodiment different from the above third embodiment can be also realized although it is obtained by a combination of the first and second embodiments.

Fig. 19 shows a schematic construction of a DAB receiver according to the fourth embodiment and portions similar to those in Fig. 17 are designated by the same reference numerals.

In Fig. 19, in a manner similar to the processes shown in Fig. 5 mentioned above, a control part 30c has the first muting control means 3H for generating the first muting control signal (X) to designate the muting level every Nm unit frames on the basis of the error detection information and the header detection information by the header CRC or ScF-CRC from the audio decoder 40. The control part 30c has second muting control means which is constructed by the means 3A to 3D used in the control part 30a in Fig. 13 and executes the processes shown in Figs. 14 and 15 in cooperation with each other and generates the second muting control signal (X') to designate the muting level every frame on the basis of the input/output of the error corrector 38.

The control part 30c further has switching means 3J for alternatively switching and generating the first and second muting control signals. The switching operation is performed in a manner such that when the error amount X shown by the first muting control signal is equal to or less than the predetermined value X_{th} , the second muting control signal (X') is transmitted to the digital attenuator 42 and when the error amount X shown by the first muting control signal is larger than the predetermined value X_{th} , the first muting control signal (X) is transmitted to the digital attenuator 42. The "soft mute" properly corresponding to the receiving state, consequently, can be performed.

Explaining in detail, when the receiving state is relatively good, the second muting control signal (X') can be treated as a signal according to an estimation error (frequency of occurrence of the dissident bits; refer to the above explanation) having a considerably high reliability. When the receiving state deteriorates to a certain degree, the signal (X') becomes a signal according to an unreliable estimation error. While having the above characteristics, the second muting control signal is updated every frame of a short period as mentioned above, so that a predetermined high response speed for the error occurred in the "soft mute" control can be realized.

On the other hand, the first muting control signal (X) is a signal based on the error due to the CRC which

always has a high reliability irrespective of the receiving state. Since this signal is updated every N_m frames of a long period as mentioned above, the "soft mute" control in which a response speed for the error occurred is low is derived.

To prevent it, in the embodiment, when the first muting control signal is in a relatively good receiving state indicative of the error smaller than a predetermined value, the "soft mute" is executed by using the second muting control signal that is advantageous in the response speed of the muting control and, when the first muting control signal is in a deteriorated receiving state showing the error larger than the predetermined value, the "soft mute" is performed by using the first muting control signal that is advantageous in the reliability of the transmission data for the error.

Since the embodiment has been constructed so as to effect the advantages of the first and second muting control signals while mutually compensating the disadvantages, the "soft mute" which properly acts in accordance with the receiving state and intends to accomplish a good response speed as much as possible can be realized.

The construction of the control part 30c is expressed by software as shown in Fig. 20. Fig. 20 is shown on the assumption that steps S1 to S7 (process for calculating X) shown in Fig. 11 and steps S11 to S21 (process for calculating X') shown in Figs. 14 and 15 are executed in parallel as a prerequisite and is shown as a flowchart for processes corresponding to the operation of the switching means 3J in Fig. 19.

During the parallel execution of the calculating processes of X and X' , the control part 30c discriminates whether the error rate X based on the CRC error has been updated or not (step S51). This discrimination can be accomplished by, for example, comparing the previous value and the present value of X and checking whether they coincide or not. When it is determined that the error rate X was updated, a check is made to see if the error rate X is larger than the predetermined threshold value X_{th} (step S52).

When it is decided in step S52 that the error rate X is larger than the threshold value X_{th} , the control part 30c sets a flag f indicative of such a fact (step S53), generates a muting control signal according to the updated error rate X , and supplies it to the digital attenuator 42 (step S54).

On the contrary, when the error rate X is determined to be equal to or less than the threshold value X_{th} in step S52, the control part 30c resets the flag f (step S55) and discriminates whether the error rate X' based on the input/output of the error corrector 38 has been updated or not (step S56). The discrimination can be also accomplished by, for instance, comparing the previous value and the present value of X' and checking whether they coincide or not. When it is determined in step S56 that the error rate X' was updated, the control part 30c generates a muting control signal according to

the updated error rate X' and supplies it to the digital attenuator 42 (step S57).

When it is decided in step S51 that the error rate X is not updated, a check is made to see if the flag f has been reset or not (step S58). If YES, step S56 follows. A reset state of the flag f denotes the result derived via step S53 and shows that $X \leq X_{th}$, namely, the result in which the receiving state is good to a certain extent has already been discriminated in step S52 just before step S53. The generation of the muting control signal, therefore, by the error rate X' based on the input/output of the error corrector 38 is permitted.

If the flag f has been set in step S58, this denotes the result via step S53 mentioned above and shows that $X > X_{th}$, namely, the result in which the receiving state remarkably deteriorated has already been discriminated in step S52 just before step S58. The processing routine of this flowchart is, therefore, finished without shifting to step S56 of permitting the generation of the muting control signal by the error rate X' .

Even when it is determined in step S56 that the error rate X' is not updated, the processing routine of this flowchart is finished.

A muting format that is accomplished by the control part 30c is shown in Figs. 21A to 21C. According to Figs. 21A to 21C, it will be understood that either one of the coarse adjustment muting operation by the first muting control signal and the fine adjustment muting operation by the second muting control signal is executed at a period ($N_m = 5$ in the embodiment) of the first muting control process.

The fine adjustment muting operation by the second muting control is performed at the first, third, and fifth periods (#1, #3, #5) mentioned above. The coarse adjustment muting operation by the first muting control is performed at the second and fourth periods (#2, #4). This is because at the first, third, and fifth periods (#1, #3, #5), the number of times of error occurrence by the CRC at the periods (#0, #2, #4) just before those periods is small to be 2 or less, so that the receiving state is determined to be good to a certain degree, and the fine adjustment muting operation by the second muting control is executed by preferentially considering the response speed of the "soft mute" control for the generated error. At the second and fourth periods (#2, #4), since the number of times of error occurrence by the CRC at the periods (#1, #3) just before those periods is large to be 3 or more, the receiving state is determined to deteriorate to a certain extent, and the coarse adjustment muting operation by the first muting control is executed by paying importance to the reliability of the error rate serving as a base of the muting level to be set in place of sacrificing the response speed of the "soft mute" control for the generated error. In the embodiment, a condition that the threshold value is set to $2/5 \leq X_{th} < 3/5$ is used as a prerequisite.

In the fourth embodiment, there is an idea that the second muting control is resigned with the progress of

the deterioration of the receiving state. In other words, it teaches that the reliability of the error based on the input/output of the error corrector 38 deteriorates with an increase in error by the CRC. This idea can be also applied to the foregoing third embodiment for performing the muting control based on the addition value of both errors. That is, in order to reduce the dependency of the error based on the input/output of the error corrector 38 on the muting control signal to be supplied to the attenuator 42 as the error by the CRC increases, in the adding means 3I shown in Fig. 17, the coefficient, for example, X' is multiplied by a function $F(X)$ whose value decreases with an increase in value of X and a multiplication result X'' and X are added, thereby deciding the muting level to be held in the final muting control signal.

The above modification example is as shown in Fig. 22 and is shown by a flowchart based on the prerequisite in Fig. 20.

First, a check is made to see if the error rate X based on the CRC error has been updated in a manner similar to above step S51 (step S61). When it is decided that the error rate X was updated, the error rate X' based on the input/output of the error corrector 38 is multiplied by the function $F(X)$ of the error rate X , thereby deriving multiplication result X'' (step S62). A muting control signal according to a value obtained by adding the multiplication result X'' to the error rate X is generated (step S63).

When it is determined in step S61 that the error rate X is not updated, whether the error rate X' has been updated or not is discriminated in a manner similar to step S56 (step S64). When it is decided in step S61 that the error rate X' was updated, the processing routine advances to step S62 and the multiplication result X'' is derived with respect to the updated X' and, after that, the corresponding muting control signal is generated.

When it is decided in step S64 that the error rate X' is not updated after all, since this means that both of X and X' are not updated, the processes of this flowchart are finished without shifting to step S62.

Now, assuming that the function $F(X)$ is a function whose value decreases as the value of X is large, as mentioned above, it will be understood that the dependency of the error rate X' based on the input/output of the error corrector 38 for the muting control signal to be supplied to the attenuator 42 decreases as the error rate X due to the CRC increases. An operation and an effect similar to those in the fourth embodiment, consequently, can be obtained.

As a function $F(X)$, moreover, a function of two values such that it is set to 0 when $X > X_{th}$ and to a predetermined value when $X \leq X_{th}$ can be also used. According to this function, when the CRC error is large, the setting of the muting level based on the error of the input/output of the error corrector 38 is perfectly stopped and, when the CRC error is small, the setting of the muting level based on the input/output error of the

error corrector 38 can be adjusted by only a predetermined ratio.

In each of the foregoing embodiments, the "soft mute" has been performed at the muting level which is substantially unconditionally determined for the error rate X , X' , and/or X'' , the muting level can be also set on the basis of not only the error rates but also the code rate as shown in the following embodiment.

In the DAB, as already mentioned above, the code rate is variable in accordance with the transmission data and the data concerning the code rate is included in the FIC data signal. The code rate corresponds to the protection level, the low code rate corresponds to the high protection level, and the high code rate corresponds to the low protection level. The low protection level corresponds to the low error correcting ability. In this case, it is preferable to set the muting level to be higher than that in the case of the high protection level. When the protection level is high, on the contrary, it is desirable to set a relatively small muting level. By deciding the muting level for the error rate, therefore, on the basis of the code rate data in the FIC data signal, in more detail, by determining the change ratio of the muting level for the error rate in accordance with the protection level corresponding to the relevant code rate data, the good "soft mute" according to the actual situation can be realized.

Fig. 23 shows a schematic construction of a DAB receiver according to the fifth embodiment and portions similar to those in Figs. 10 and 13 are designated by the same reference numerals.

A construction shown in Fig. 23 is based on the construction of Fig. 13. Adjusting means 3K for adjusting an output of the comparing means 3F and generating a new muting control signal is provided in a modified control part 30d. This adjustment is performed in accordance with the code rate data from the FIC decoder 3A.

By the above construction, the muting control signal according to not only the error based on the input/output of the error corrector 38 but also the code rate can be generated. The proper muting level adapted to the reproduction acoustic format of the transmission data can be designated to the attenuator 42. Similar adjusting means can be also provided for the construction other than Fig. 13 mentioned above.

In the above explanation, although the digital attenuator provided at the front stage of the D/A converter has been used as means for actually muting, the invention is not limited to it. It will be obviously understood that an analog attenuator is provided at the post stage of the D/A converter and can be used or other equivalent means can be also used.

Although each of the above embodiments has been described with respect to N_m , m , and other parameters with the specific numerical values, it will be obviously understood that the invention is not limited to those values.

Although each of the above embodiments has been

described with respect to the receiver adapted to a specific DAB system, the invention is not fundamentally restricted to only this system. In brief, the invention can be applied to any system having a system for receiving a digital audio signal and is valid for a data signal such as a foregoing subband sampling signal such that no error detection signal is added. Particularly, the format such that the error of the relevant data signal is presumed on the basis of the input/output of the error corrector 38 in the embodiments can be said to be extremely valid to a digital signal of a format having no error detection signal.

Step S22 (refer to Fig. 9) is provided in the second muting control and the audio signal of the corresponding frame is certainly soft-muted. It is, however, not always necessary to perfectly coincide the frame of the error detection with the frame as a target of the "soft mute". Even if the control part doesn't obtain the timing for "soft mute" on the basis of the frame output start information from the decoder 40, therefore, the "soft mute" can be also performed at a time point when X' is obtained. For the purpose of perform the accurate "soft mute", it is preferable to match the frames on the time base as shown in step S22.

In Figs. 14 and 15, although J error rates X_j have been obtained, the number of error rates can be also set to be $J = 1$ or it is also possible to obtain a bit error with respect to partial data in the frame and to perform the "soft mute" by this error with regard to the audio signal of the whole frame.

Although the embodiments have been described above while limiting the various means, furthermore, many modifications are possible within a purview where those skilled in the art can design.

According to the invention, a good muting operation can be accomplished even for a digital audio signal having no error detection code.

Claims

1. An information data signal receiver for receiving a transmission wave in which a specific information data signal and an identification data signal to identify that said specific information data signal is valid are transmitted by a same frequency band, comprising:

receiving and demodulating means for receiving said transmission wave and demodulating it to a predetermined digital signal;

decoding means for decoding said digital signal;

error detecting means for obtaining an error amount of said digital signal which is recognized in a decoding process by said decoding means;

evaluating means for evaluating said error amount;

identification data detecting means for detecting said identification data signal from a decoded output of said decoding means; and control means for reproducing and outputting said specific information data signal concerning said identification data signal on the basis of an evaluation result of the error amount by said evaluating means and said identification data signal.

2. A receiver according to claim 1, wherein:

said transmission wave includes a train of data blocks containing a digital audio signal, a digital additional information signal associated with said digital audio signal, and an error detection signal for said digital additional information signal;

said decoding means has an audio decoder for decoding said data blocks and detecting a data error of said digital additional information signal by said error detection signal;

said error detecting means generates an error rate signal according to a ratio of the number of detecting times of said data error by said audio decoder for the number of decoding processing times of said data blocks in said audio decoder; said evaluating means discriminates whether said error rate signal is smaller than a predetermined value or not; and

said control means performs a control so as to reproduce and generate the specific information data signal concerning said identification data signal in the case where said evaluating means determines that said error rate signal is smaller than the predetermined value and said identification data detecting means detects said identification data signal.

3. A receiver according to claim 2, wherein said error detecting means updates said error rate signal every predetermined number of said data blocks.

4. A receiver according to claim 2 or 3, wherein when the data error of said digital additional information signal is detected by said error detection signal, said audio decoder itself mutes an audio signal of a decoding output corresponding to the detected data error.

5. A receiver according to claim 1, wherein:

said transmission wave includes a train of data blocks in which at least a digital audio signal was subjected to a convolution encoding for error correction;

said decoding means has error correcting means for decoding the convolution encoding

of said data blocks and performing an error correction to said data blocks and an audio decoder for decoding the data blocks corrected by said error correcting means;

said error detecting means has buffer means for buffering an input signal of said error correcting means, reencoding means for convolution encoding an output signal of said error correcting means, and comparing means for comparing an output signal of said buffer means with an encoded output signal of said reencoding means with respect to a predetermined number of bits and generating an estimation error rate signal according to the number of dissident bits between them which is obtained by said comparison;

said evaluating means discriminates whether said estimation error rate signal is smaller than a predetermined value or not; and

said control means performs a control so as to reproduce and generate specific information data signal concerning said identification data signal in the case where said evaluating means determines that said estimation error rate signal is smaller than the predetermined value and said identification data detecting means detects said identification data signal.

6. A receiver according to claim 5, wherein said predetermined bit number is equal to a number obtained by dividing the number of constructing bits of said data block or a number smaller than said number of constructing bits, and said error detecting means updates said estimation error rate signal every said predetermined number of bits.
7. A receiver according to claim 2, wherein said data block is an audio frame according to an MPEG audio.
8. A receiver according to claim 1, wherein:

said transmission wave includes a train of data blocks which contains a digital audio signal, a digital additional information signal associated with said digital audio signal, and an error detection signal for said digital additional information signal and in which at least said digital audio signal was subjected to a convolution encoding for error correction;

said decoding means has error correcting means for decoding the convolution encoding of said data blocks and performing an error correction to said data blocks and an audio decoder for decoding the data blocks corrected by said error correcting means and detecting a data error of said digital additional information signal by said error detection signal;

said error detecting means has means for generating a first error signal according to a ratio of the number of detecting times of said data error by said audio decoder for the number of decoding processing times of said data blocks in said audio decoder, buffer means for buffering an input signal of said error correcting means; reencoding means for convolution encoding an output signal of said error correcting means, and comparing means for comparing an output signal of said buffer means with an encoded output signal of said reencoding means with respect to a predetermined number of bits and generating a second error signal according to the number of dissident bits between them which is obtained by said comparison;

said evaluating means discriminates whether each of said first and second error signals is smaller than a corresponding predetermined value or not; and

said control means performs a control so as to reproduce and generate specific information data signal concerning said identification data signal in the case where said evaluating means determines that each of said first and second error signals is smaller than said corresponding predetermined value and said identification data detecting means detects said identification data signal.

9. A receiver according to claim 2, wherein the reproduction output control of said specific information data which is executed by said control means is performed by designating a target of the decoding processor decoding output in said audio decoder to said specific information data signal.
10. A system using an information data signal receiver, comprising:

an information data signal receiver for receiving a transmission wave in which a specific information data signal and an identification data signal to identify that said specific information data signal is valid are transmitted by a same frequency band, said receiver having receiving and demodulating means for receiving said transmission wave and demodulating it to a predetermined digital signal, decoding means for decoding said digital signal, error detecting means for obtaining an error amount of said digital signal which is recognized in a decoding process by said decoding means, evaluating means for evaluating the error amount, identification data detecting means for detecting said identification data signal from a decoded output of said decoding means, and control means for reproducing and outputting said specific information data signal.

mation data signal concerning said identification data signal on the basis of an evaluation result of the error amount by said evaluating means and said identification data signal;

an audio signal output source different from said receiver;

selecting means for selectively outputting either one of an output audio signal of said audio output source and an output audio signal of said receiver in accordance with said reproduction output control signal; and
means for generating an acoustic output in accordance with a selection output of said selecting means.

11. A digital audio signal receiver having a muting function for a reproduced audio sound, comprising:

receiving and demodulating means for receiving a transmission wave including a digital audio signal and demodulating it to a predetermined digital signal;

decoding means for decoding said digital signal;

control means for obtaining an error amount of said digital signal which is recognized in a decoding process by said decoding means; and

muting means for muting an audio signal to be reproduced on the basis of said digital signal at a muting level according to said error amount.

12. A receiver according to claim 11, wherein:

said transmission wave includes a train of data blocks containing said digital audio signal, a digital additional information signal associated with said digital audio signal, and an error detection signal for said digital additional information signal;

said decoding means has an audio decoder for decoding said data blocks and detecting a data error of said digital additional information signal by said error detection signal;

said control means generates a muting control signal according to a ratio of the number of detecting times of said data error by said audio decoder for the number of decoding processing times of said data blocks in said audio decoder; and

said muting means has means for changing a dynamic range of the audio signal which is obtained from said audio decoder in accordance with said muting control signal.

13. A receiver according to claim 12, wherein said control means updates said muting control signal every predetermined number of said data blocks.

14. A receiver according to claim 12, wherein when the data error of said digital additional information signal is detected by said error detection signal, said audio decoder itself mutes an audio signal of a decoding output corresponding to the detected data error.

15. A receiver according to claim 11, wherein:

said transmission wave includes a train of data blocks in which at least said digital audio signal was subjected to a convolution encoding for error correction;

said decoding means has error correcting means for decoding the convolution encoding of said data blocks and performing an error correction to said data blocks and an audio decoder for decoding the data blocks corrected by said error correcting means;

said control means has buffer means for buffering an input signal of said error correcting means, reencoding means for convolution encoding an output signal of said error correcting means, and comparing means for comparing an output signal of said buffer means with an encoded output signal of said reencoding means with respect to a predetermined number of bits and generating a muting control signal according to the number of dissident bits between them which is obtained by said comparison;

said muting means has means for changing a dynamic range of an audio signal which is obtained from said audio decoder in accordance with said muting control signal.

16. A receiver according to claim 15, wherein said predetermined bit number is equal to a number obtained by dividing the number of constructing bits of said data block or a number smaller than said number of constructing bits, and said control means updates said muting control signal every said predetermined number of bits.

17. A receiver according to claim 12, wherein said data block is an audio frame according to an MPEG audio.

18. A receiver according to claim 11, wherein:

said transmission wave includes a train of data blocks which contains said digital audio signal, a digital additional information signal associated with said digital audio signal, and an error detection signal for said digital additional information signal and in which at least said digital audio signal was subjected to a convolution encoding for error correction;

said decoding means has error correcting means for decoding the convolution encoding of said data blocks and performing an error correction to said data blocks and an audio decoder for decoding the data blocks corrected by said error correcting means and detecting a data error of said digital additional information signal by said error detection signal;

said control means has means for generating a first muting control signal according to a ratio of the number of detecting times of said data error by said audio decoder for the number of decoding processing times of said data blocks in said audio decoder, buffer means for buffering an input signal of said error correcting means; reencoding means for convolution encoding an output signal of said error correcting means, comparing means for comparing an output signal of said buffer means with an encoded output signal of said reencoding means with respect to a predetermined number of bits and generating a second muting control signal according to the number of dissident bits between them which is obtained by said comparison, and adding means for adding said first and second muting control signals and generating a final muting control signal; and said muting means has means for changing a dynamic range of an audio signal which is obtained from said audio decoder in accordance with said final muting control signal.

19. A receiver according to claim 18, wherein said adding means multiplies said second muting control signal by a coefficient and adds a multiplication result to said first muting control signal, thereby generating said final muting control signal, and changes said coefficient in accordance with said first muting control signal.

20. A receiver according to claim 11, wherein:

said transmission wave includes a train of data blocks which contains said digital audio signal, a digital additional information signal associated with said digital audio signal, and an error detection signal for said digital additional information signal and in which at least said digital audio signal was subjected to a convolution encoding for error correction;

said decoding means includes error correcting means for decoding the convolution encoding of said data blocks and performing an error correction to said data blocks and an audio decoder for decoding the data blocks corrected by said error correcting means and detecting a data error of said digital additional information signal by said error detection signal;

said control means includes means for generating a first muting control signal according to a ratio of the number of detecting times of said data error by said audio decoder for the number of decoding processing times of said data blocks in said audio decoder, buffer means for buffering an input signal of said error correcting means; reencoding means for convolution encoding an output signal of said error correcting means, comparing means for comparing an output signal of said buffer means with an encoded output signal of said reencoding means with respect to a predetermined number of bits and generating a second muting control signal according to the number of dissident bits between them which is obtained by said comparison, and switching output means for generating said second muting control signal as a final muting control signal when an error level corresponding to said first muting control signal is equal to or less than a predetermined level and generating said first muting control signal as a final muting control signal when the error level corresponding to said first muting control signal exceeds a predetermined level; and

said muting means includes means for changing a dynamic range of an audio signal which is obtained from said audio decoder in accordance with said final muting control signal.

21. A receiver according to claim 11, wherein said transmission wave includes at least data concerning a code rate showing a code rate of said digital audio signal, and said control means further changes said muting level in accordance with the code rate shown by said data concerning the code rate.

22. A receiver according to claim 21, wherein said muting level is set to be higher as said code rate increases.

FIG.1

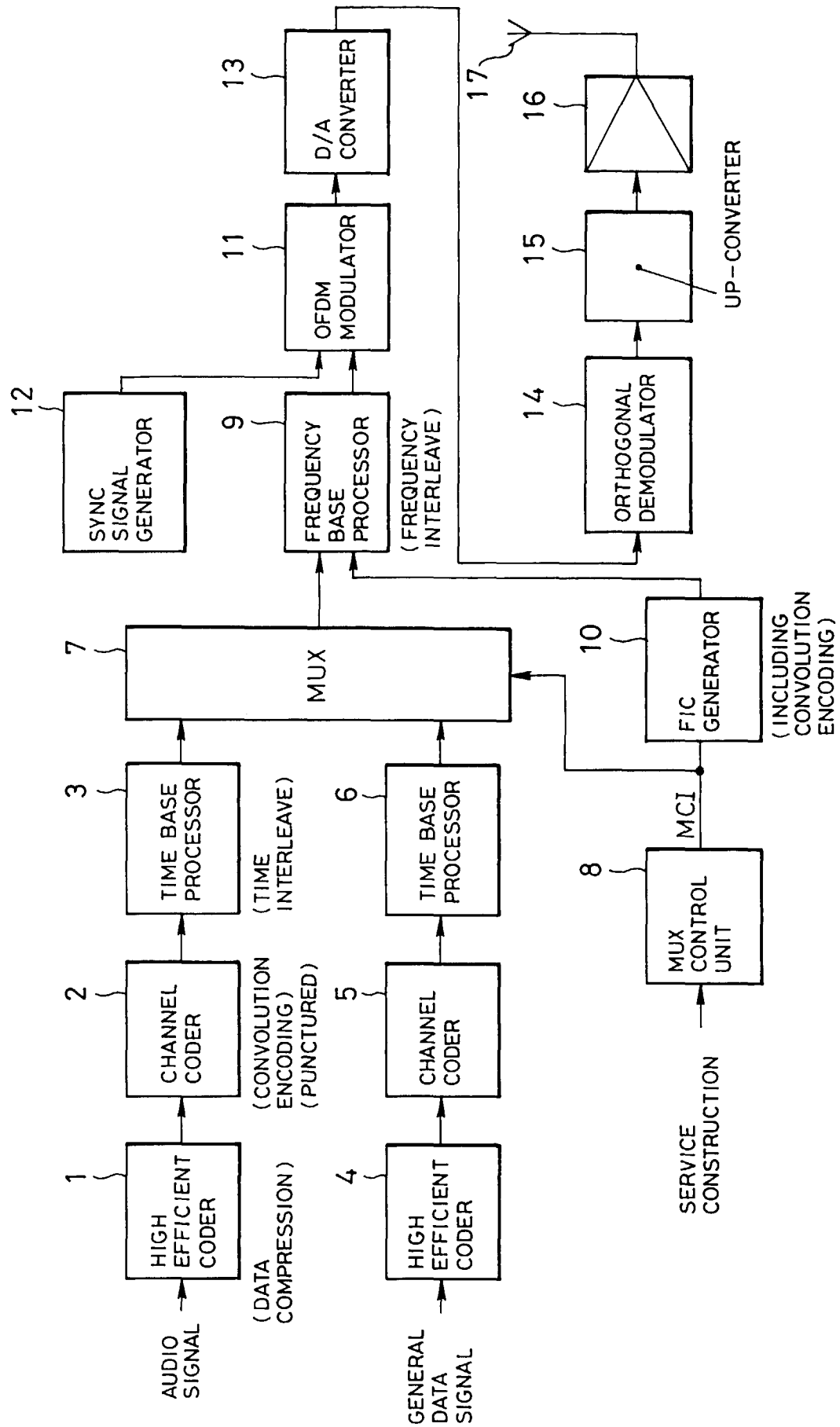


FIG. 2

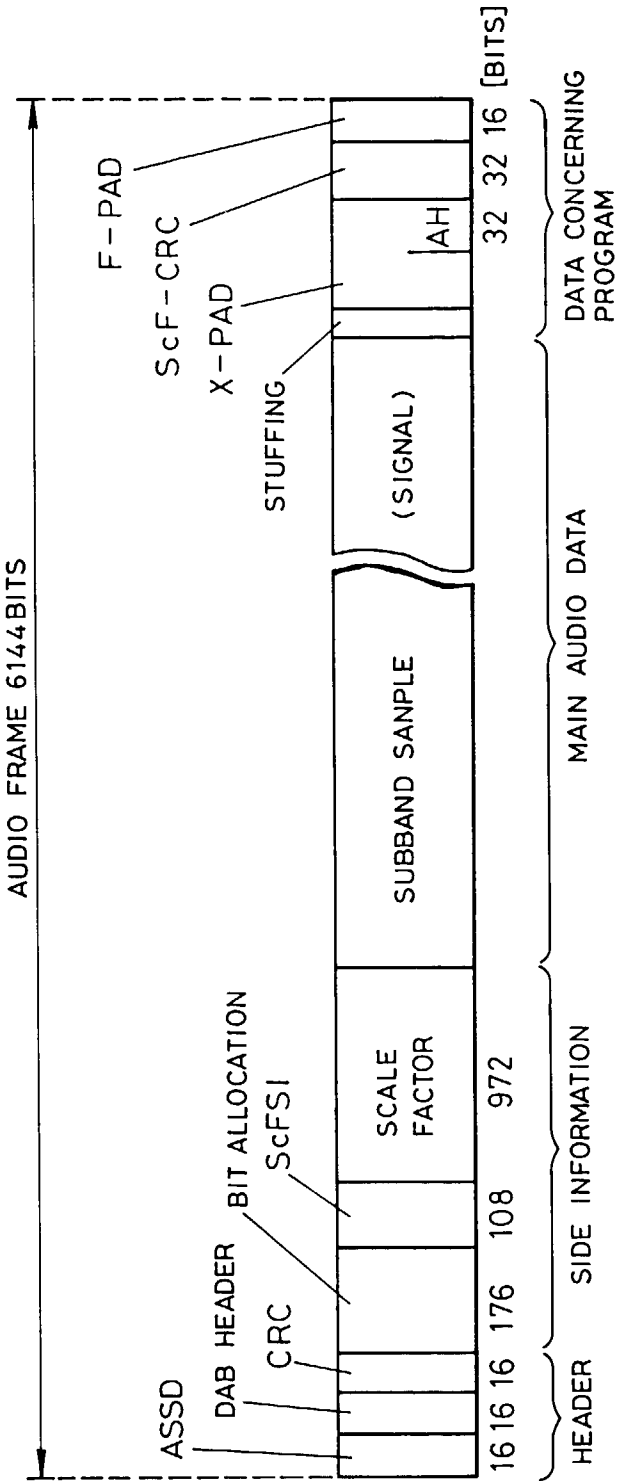


FIG. 3

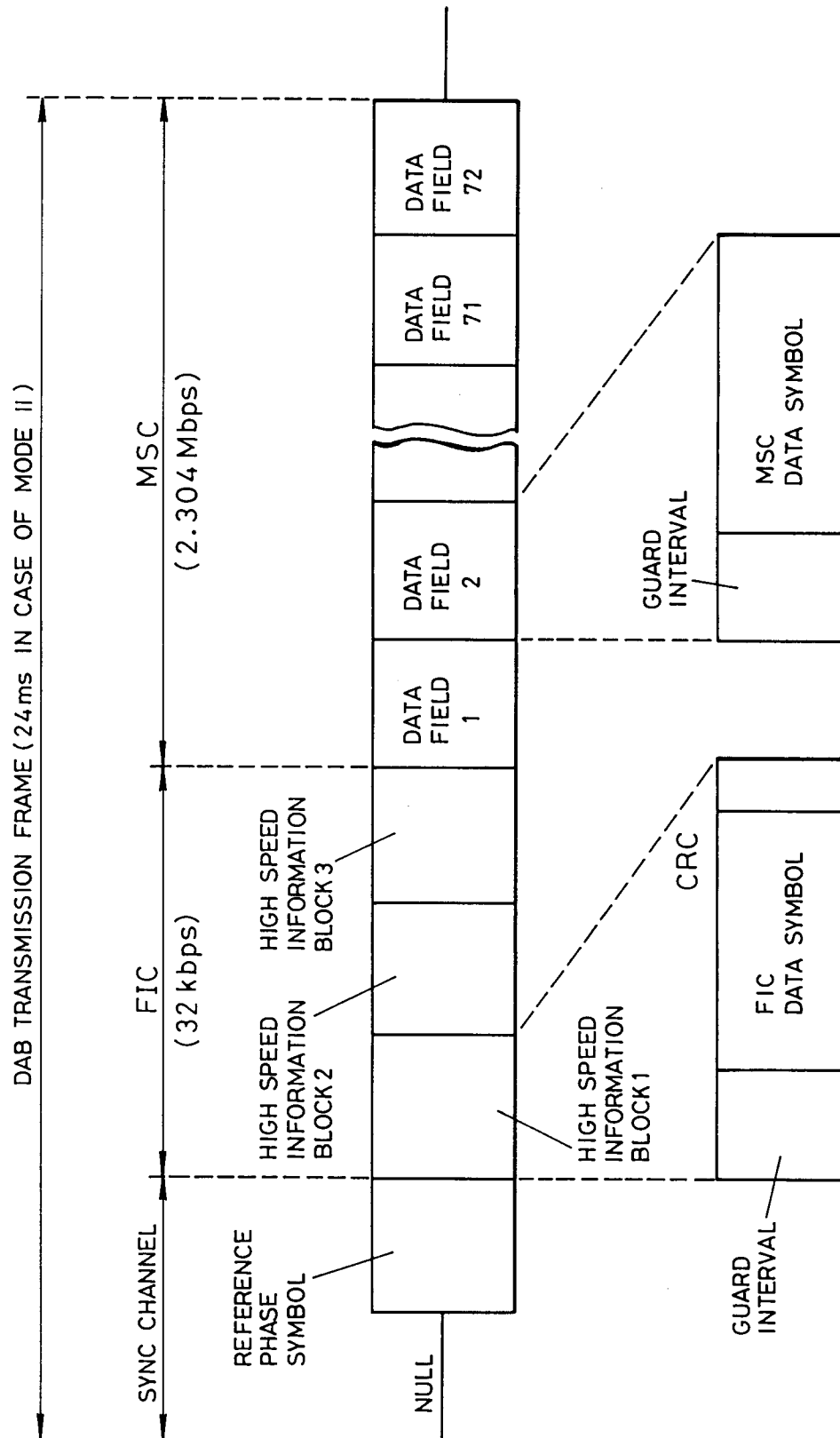


FIG. 4

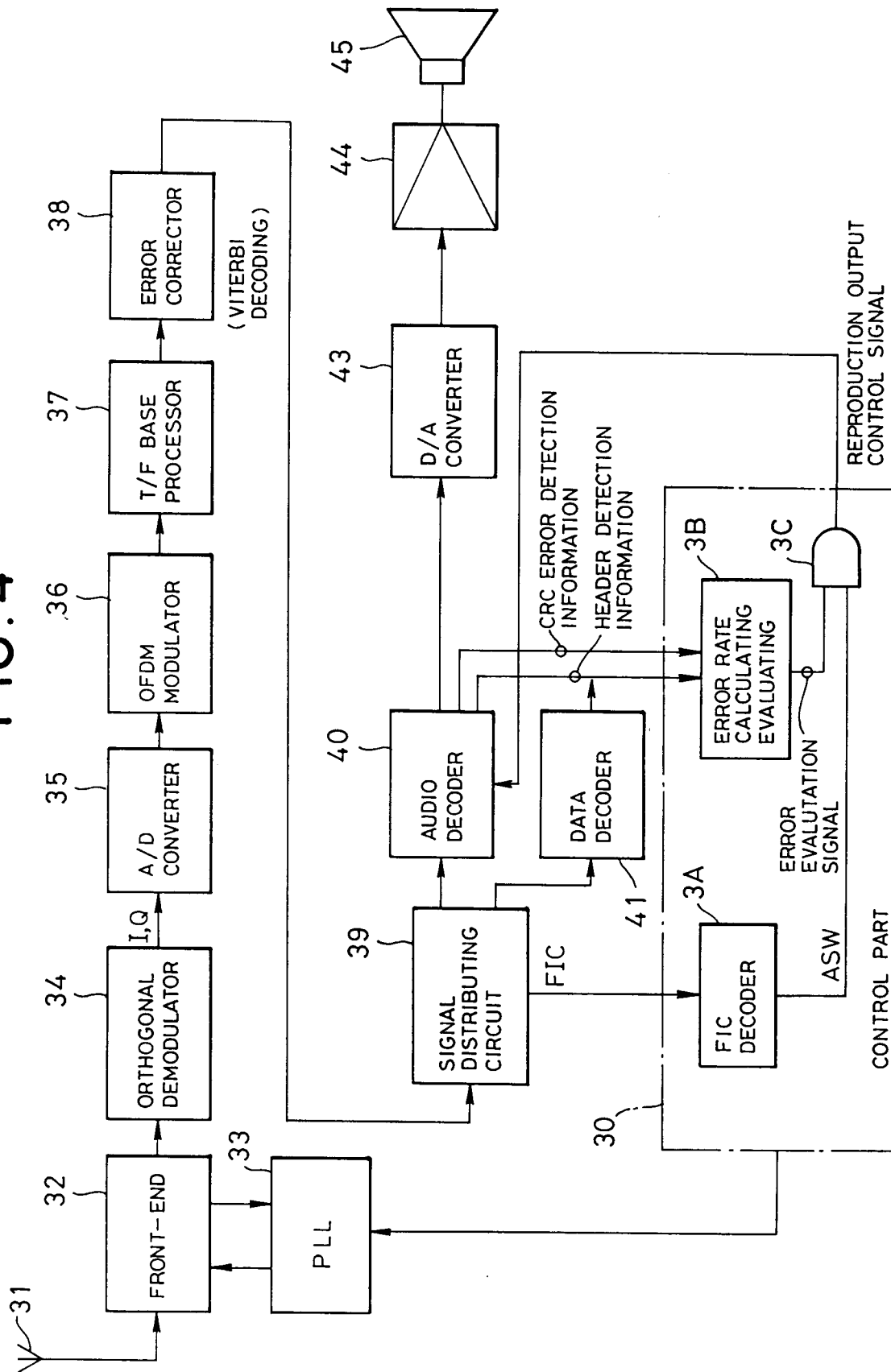


FIG. 5

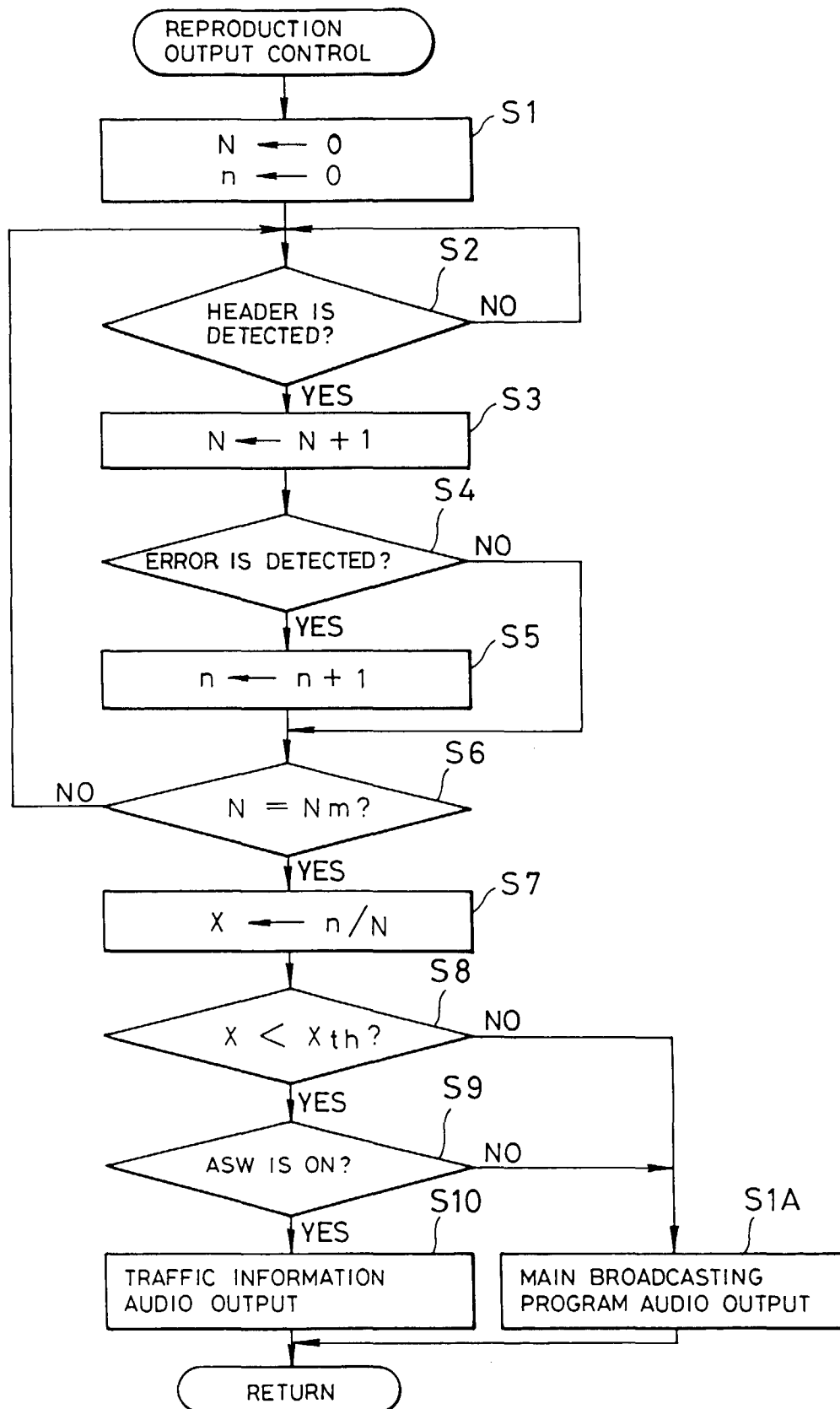


FIG. 6

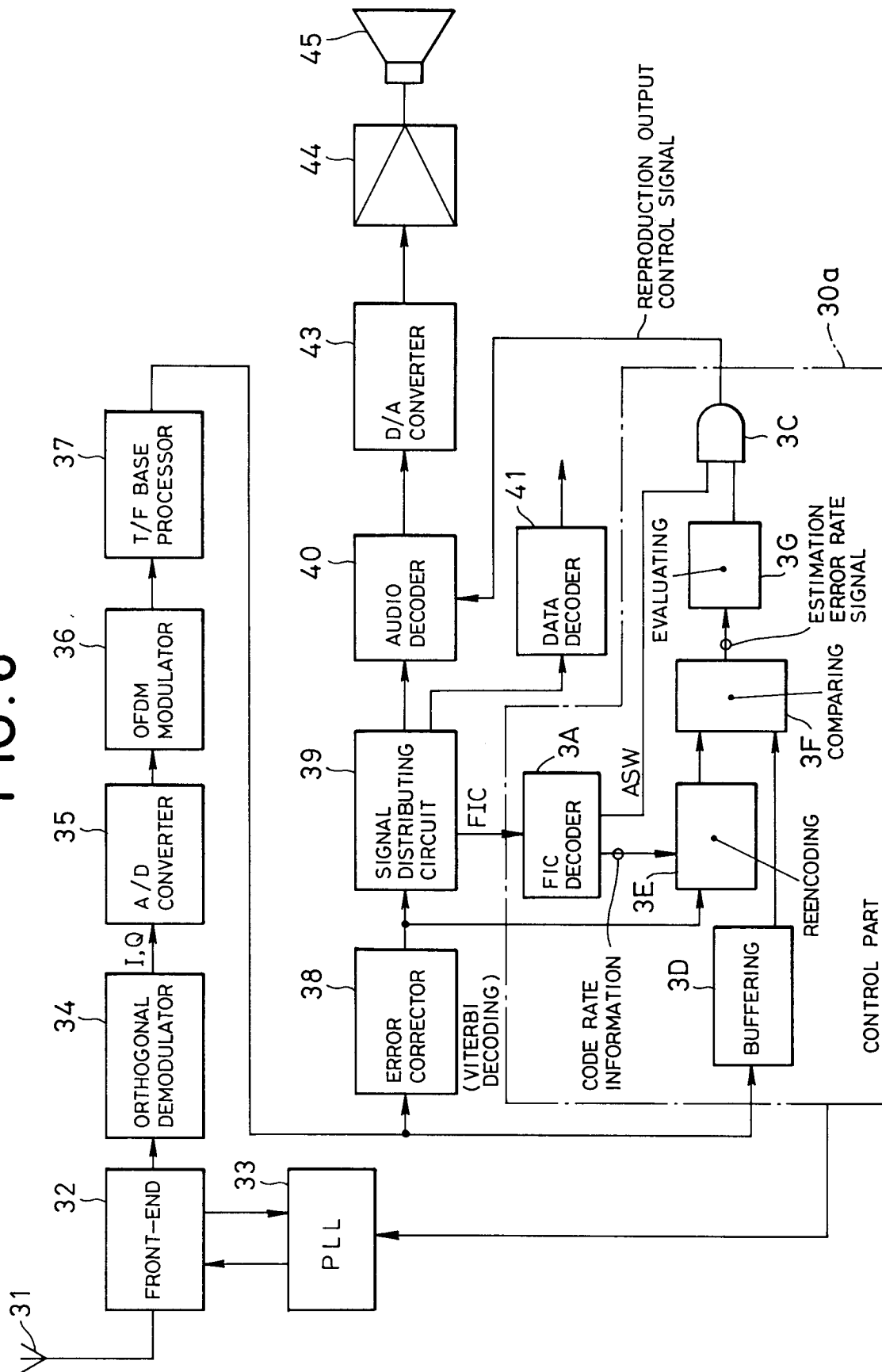


FIG. 7

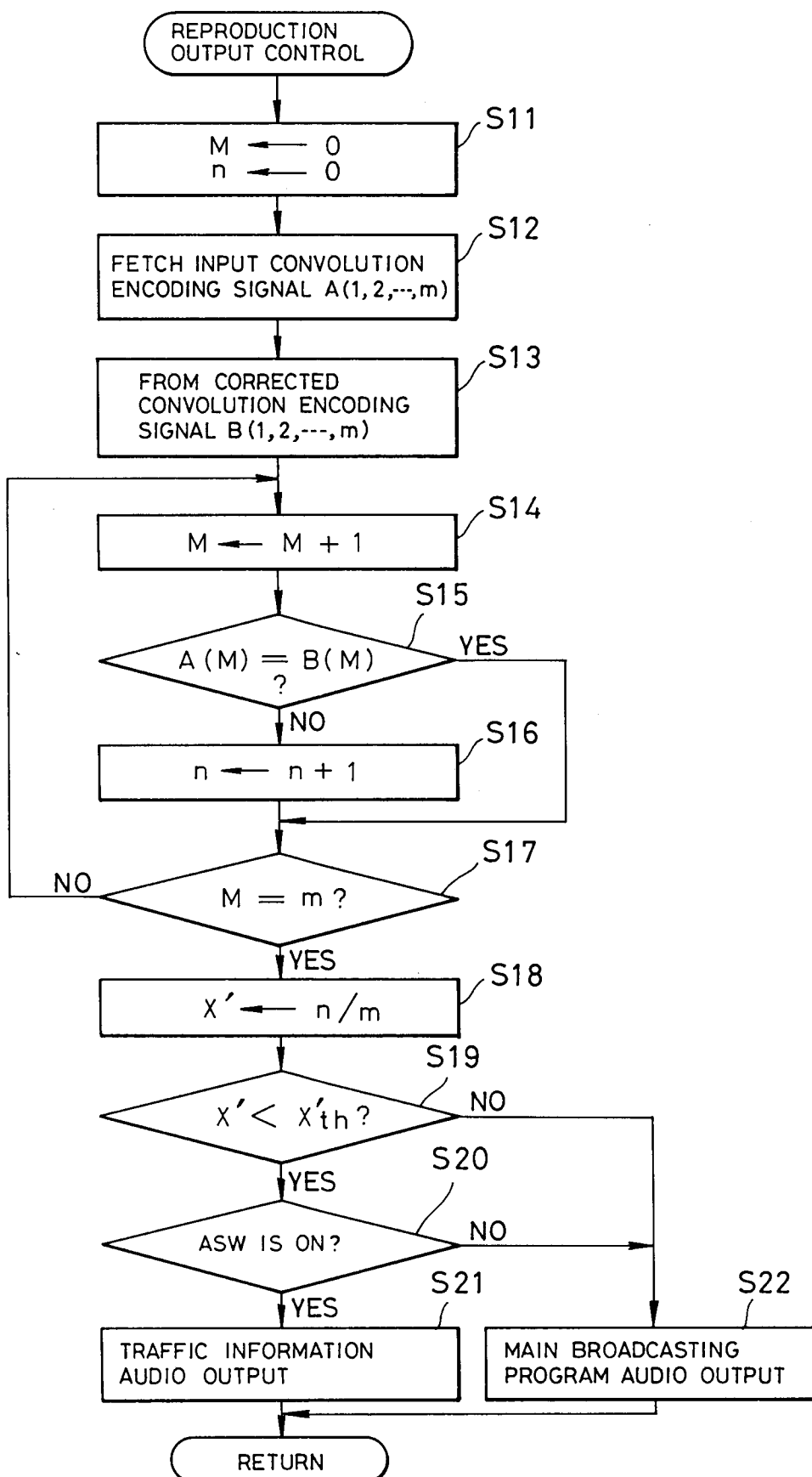


FIG. 8

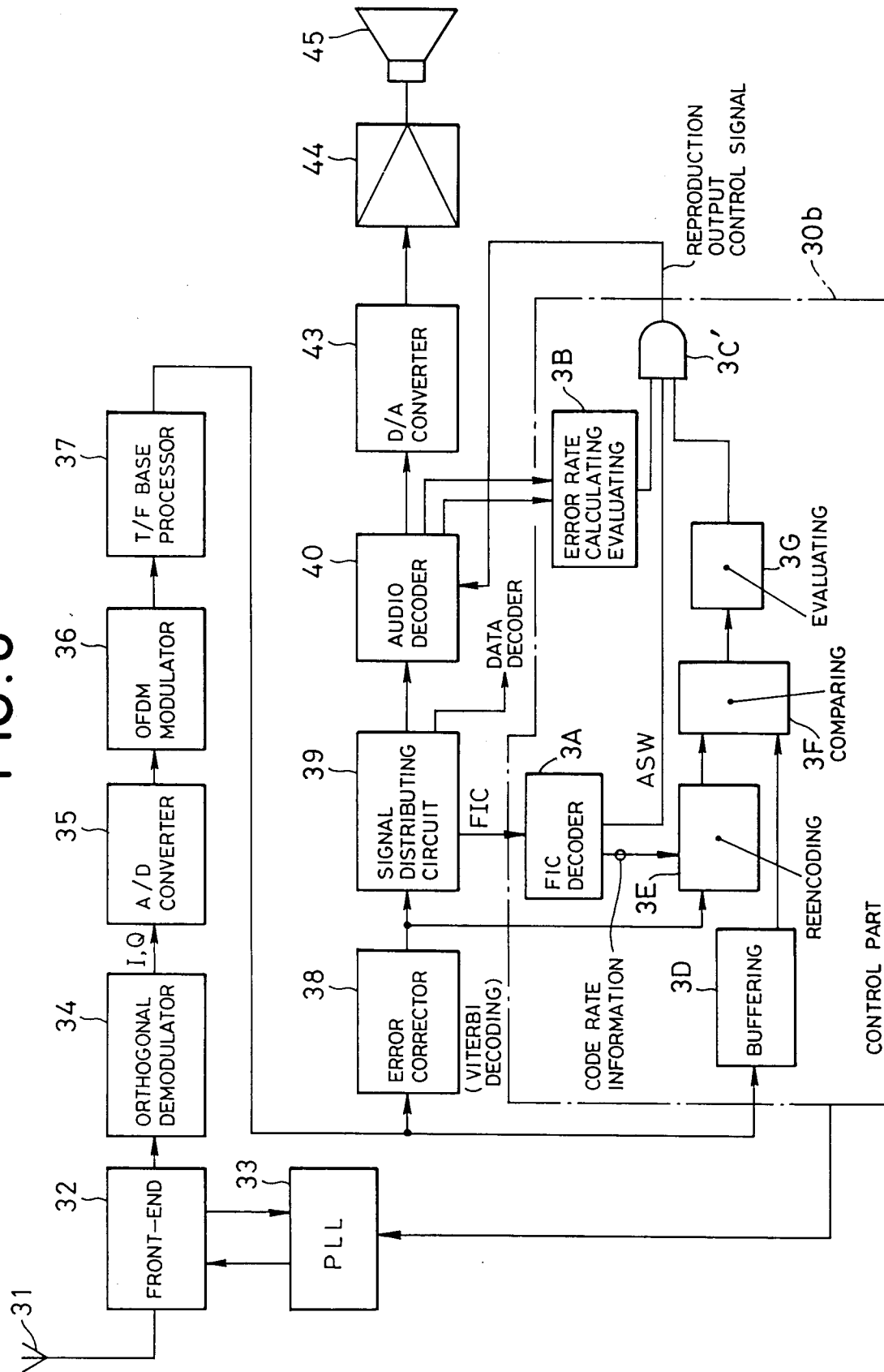


FIG. 9

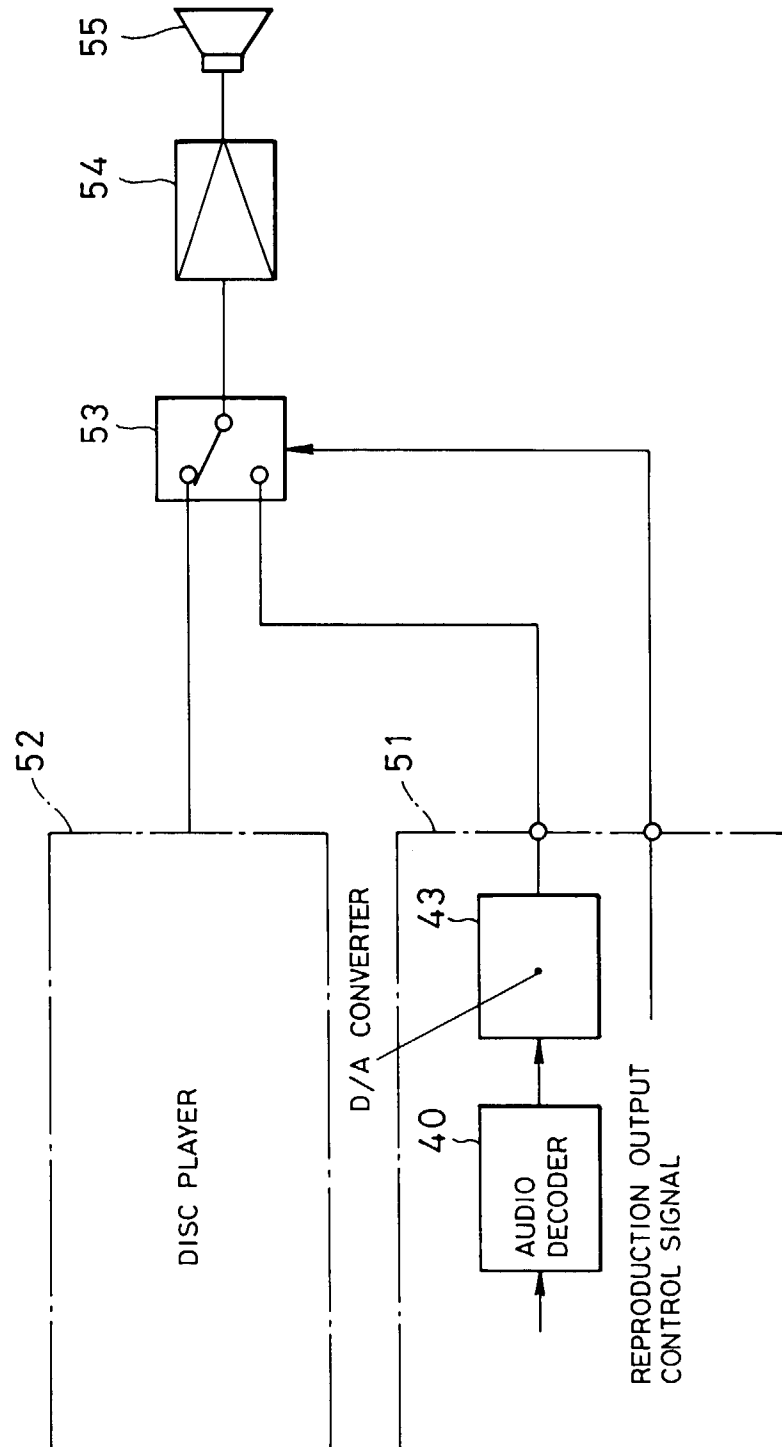


FIG. 10

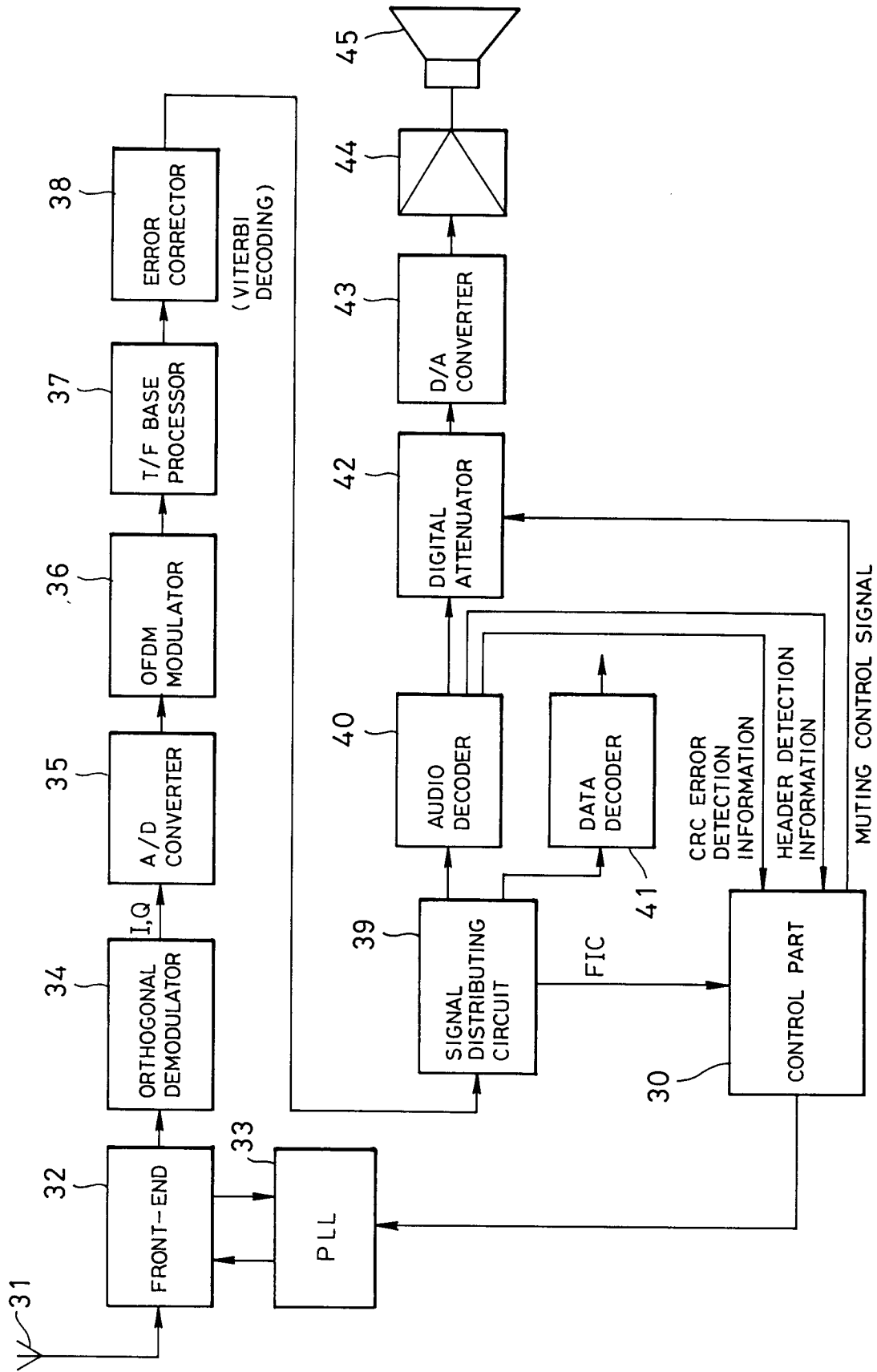
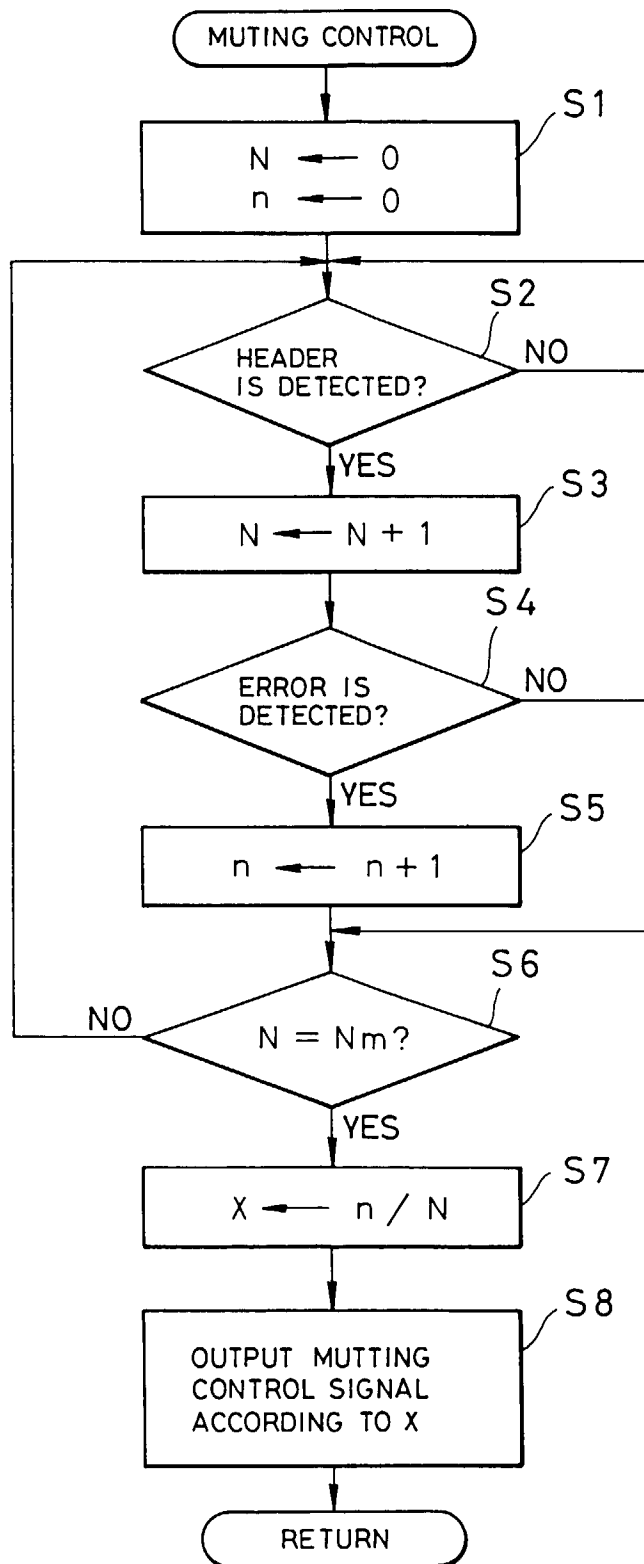


FIG.11



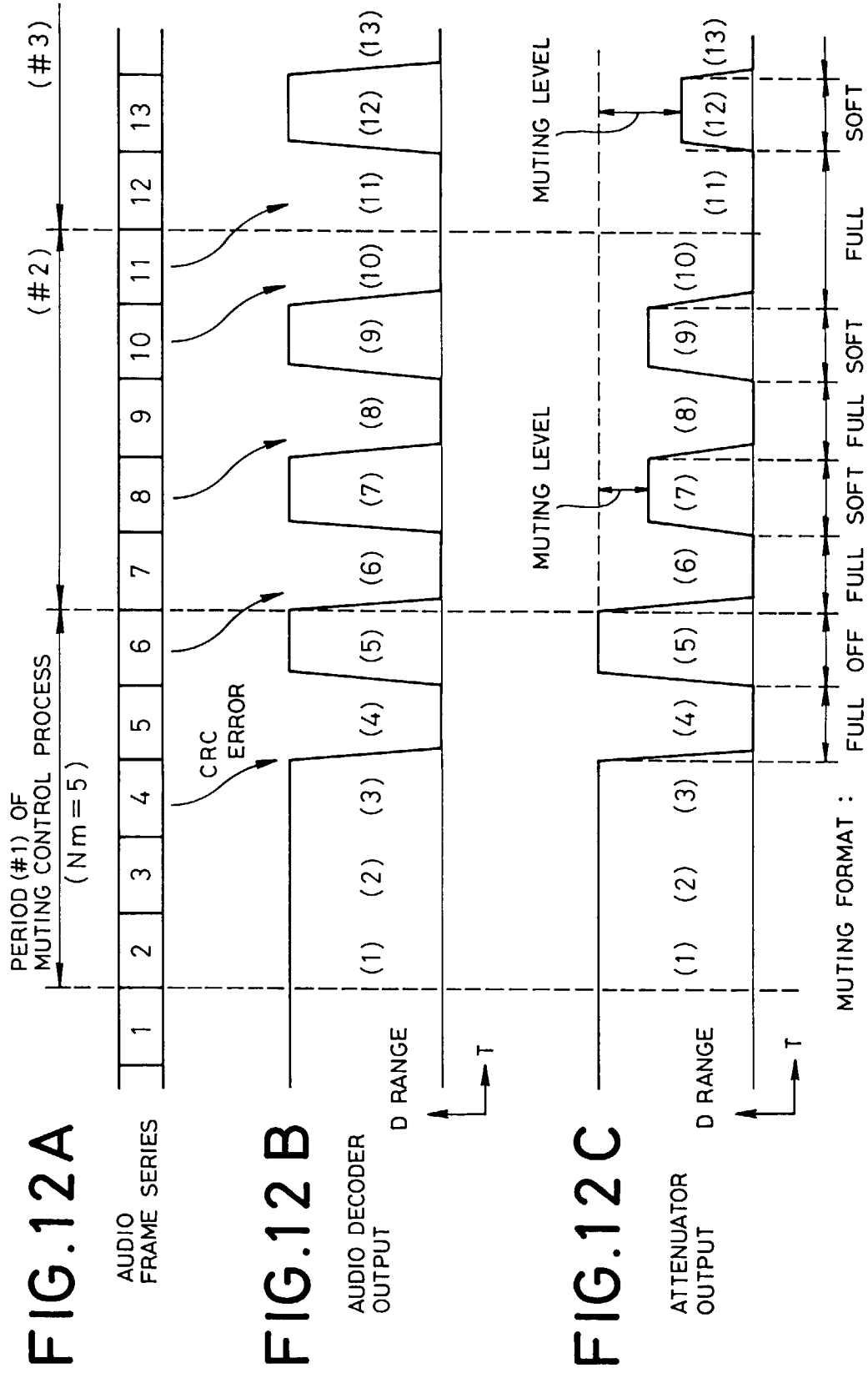


FIG. 13

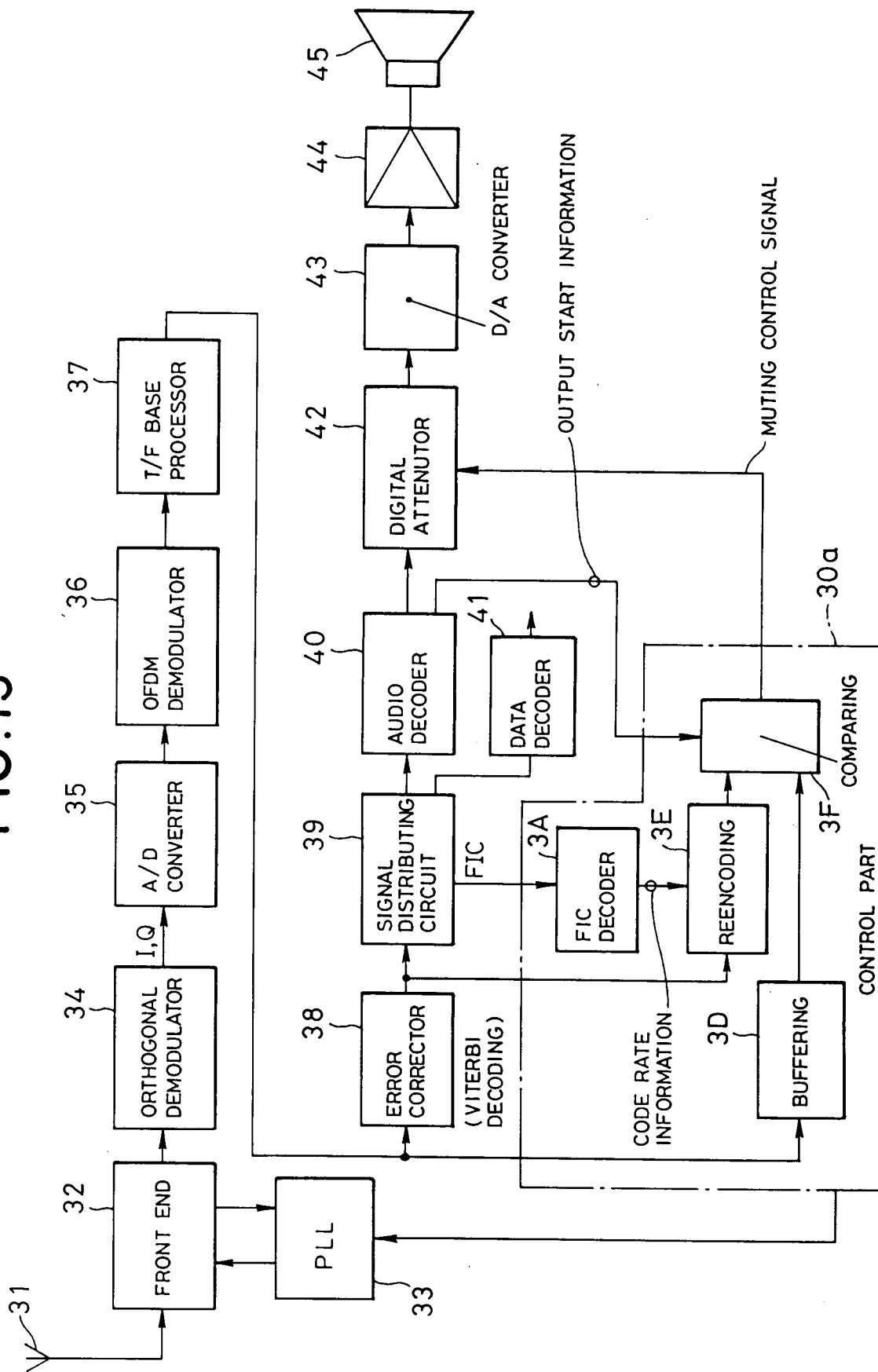


FIG.14

(2ND)

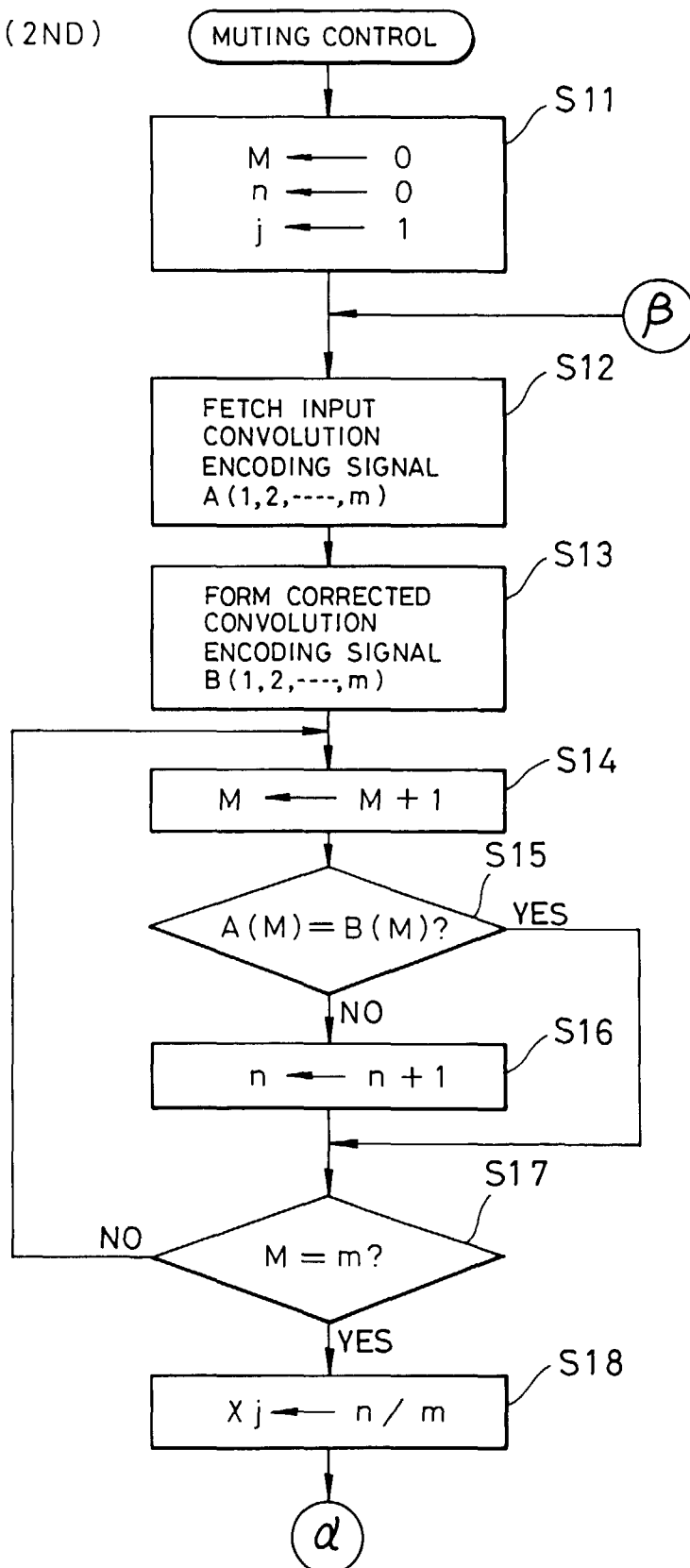
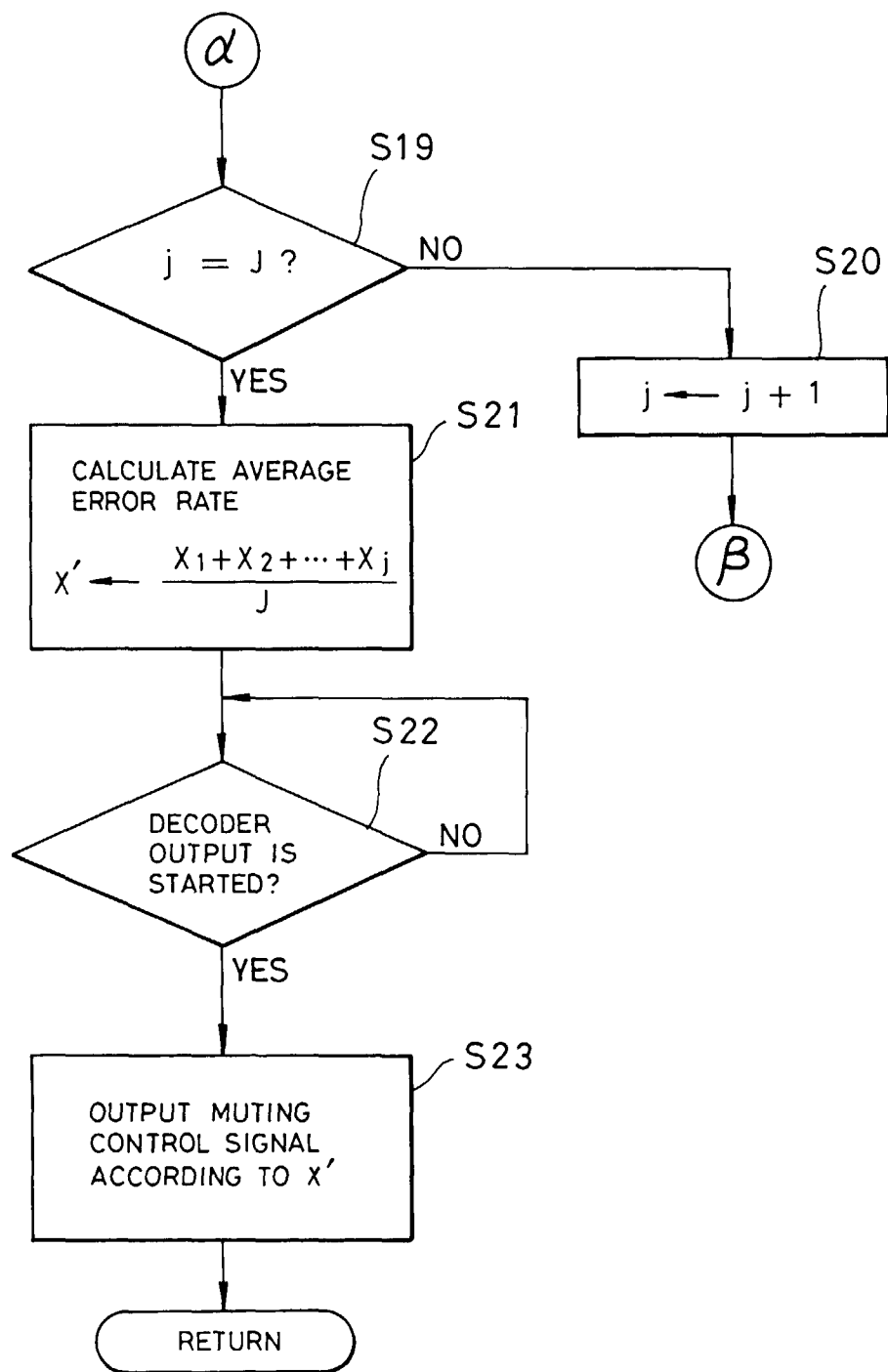


FIG. 15



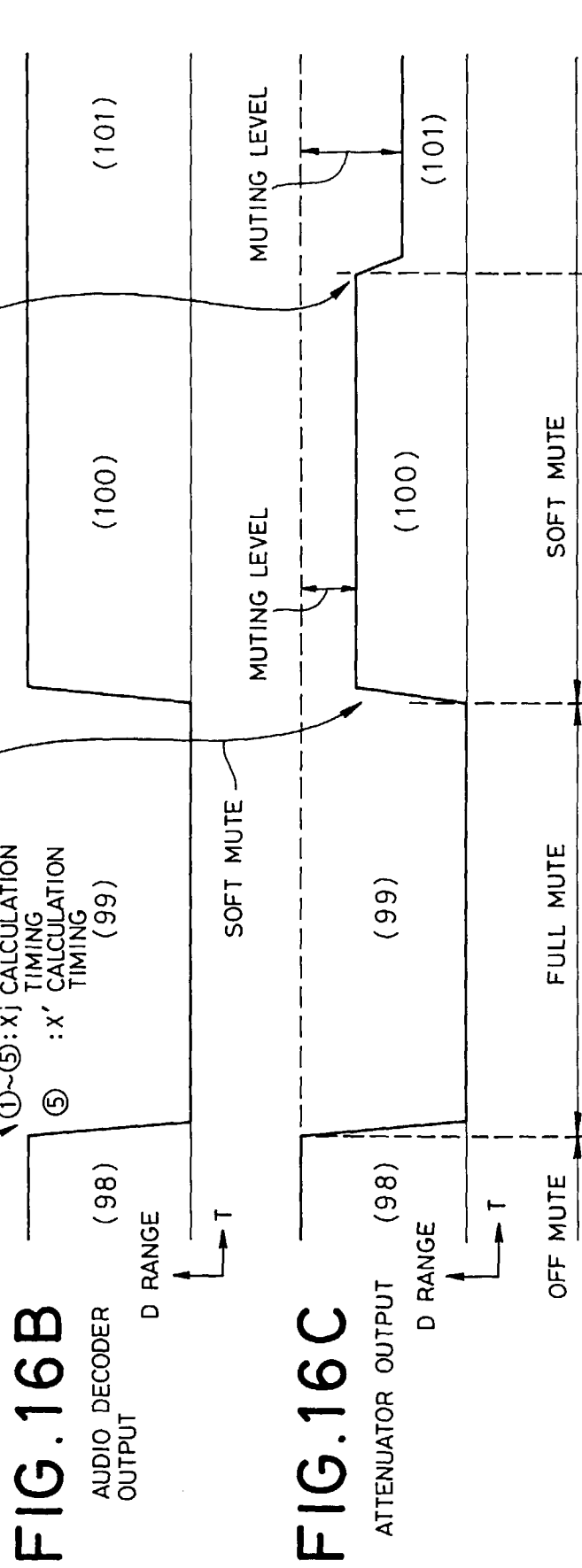
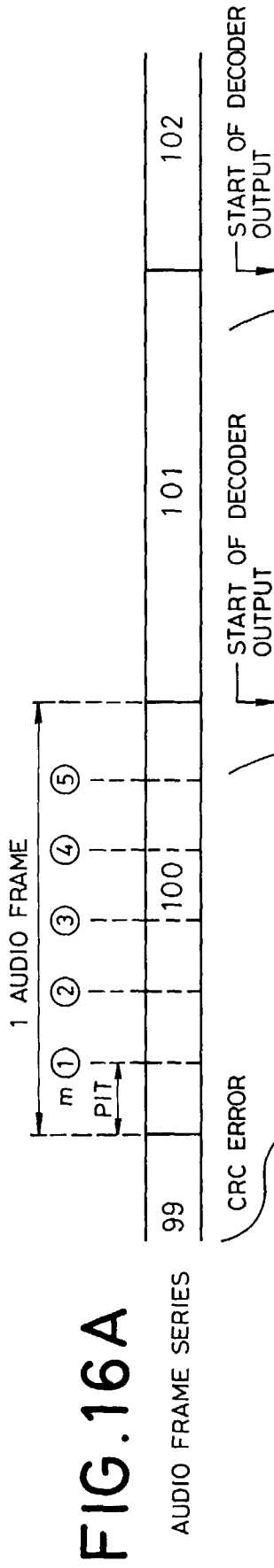
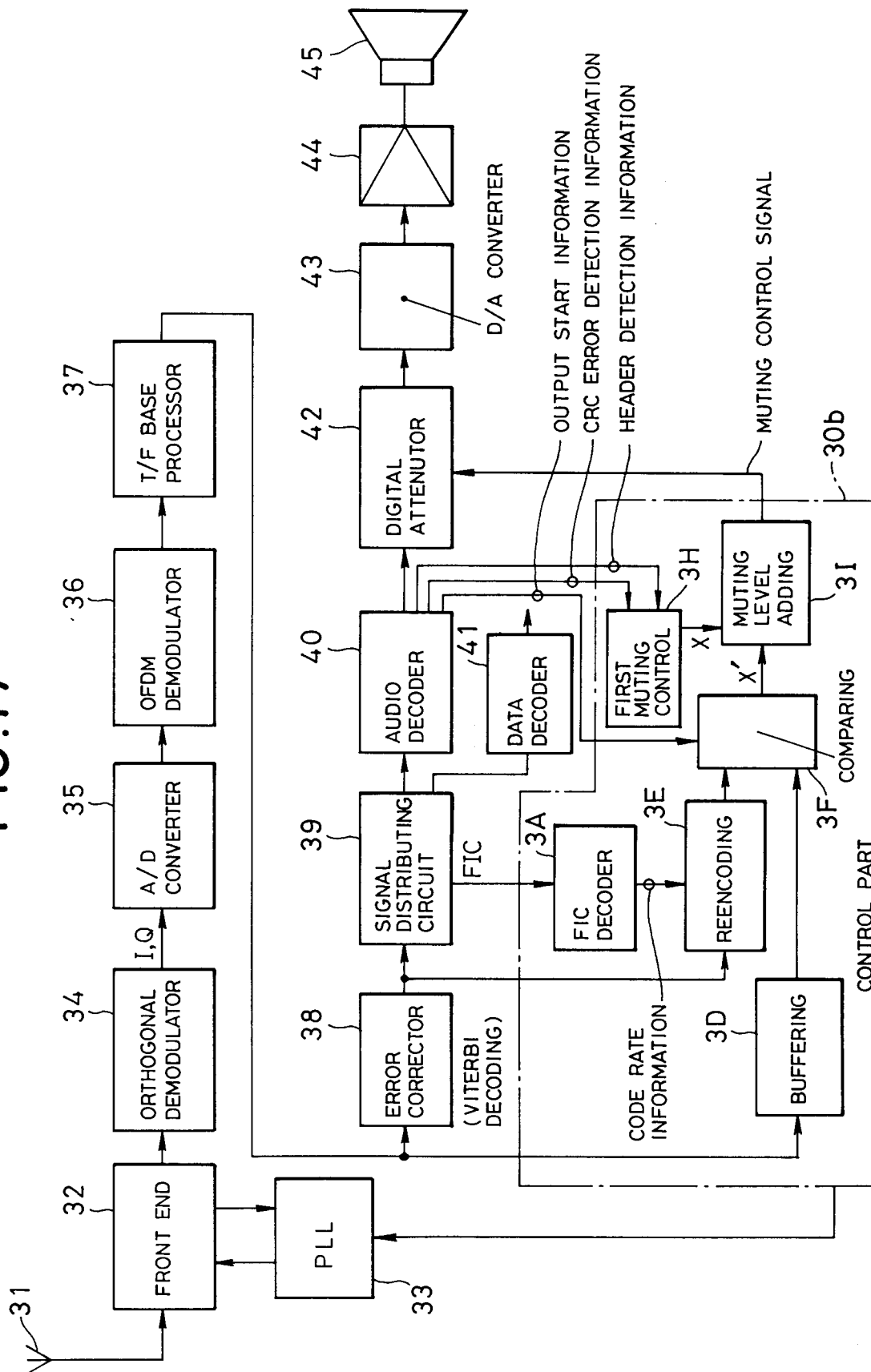


FIG. 17



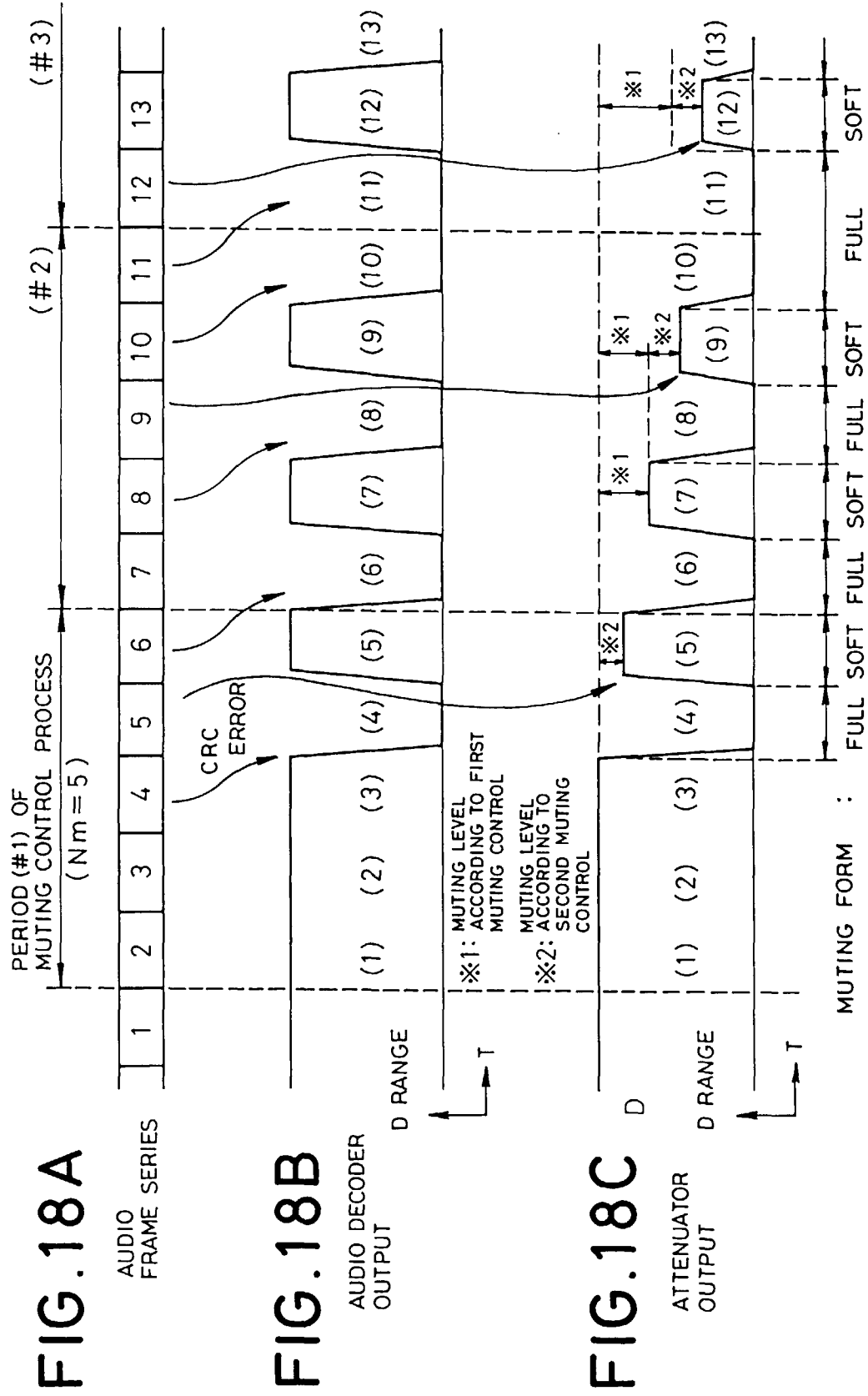


FIG. 19

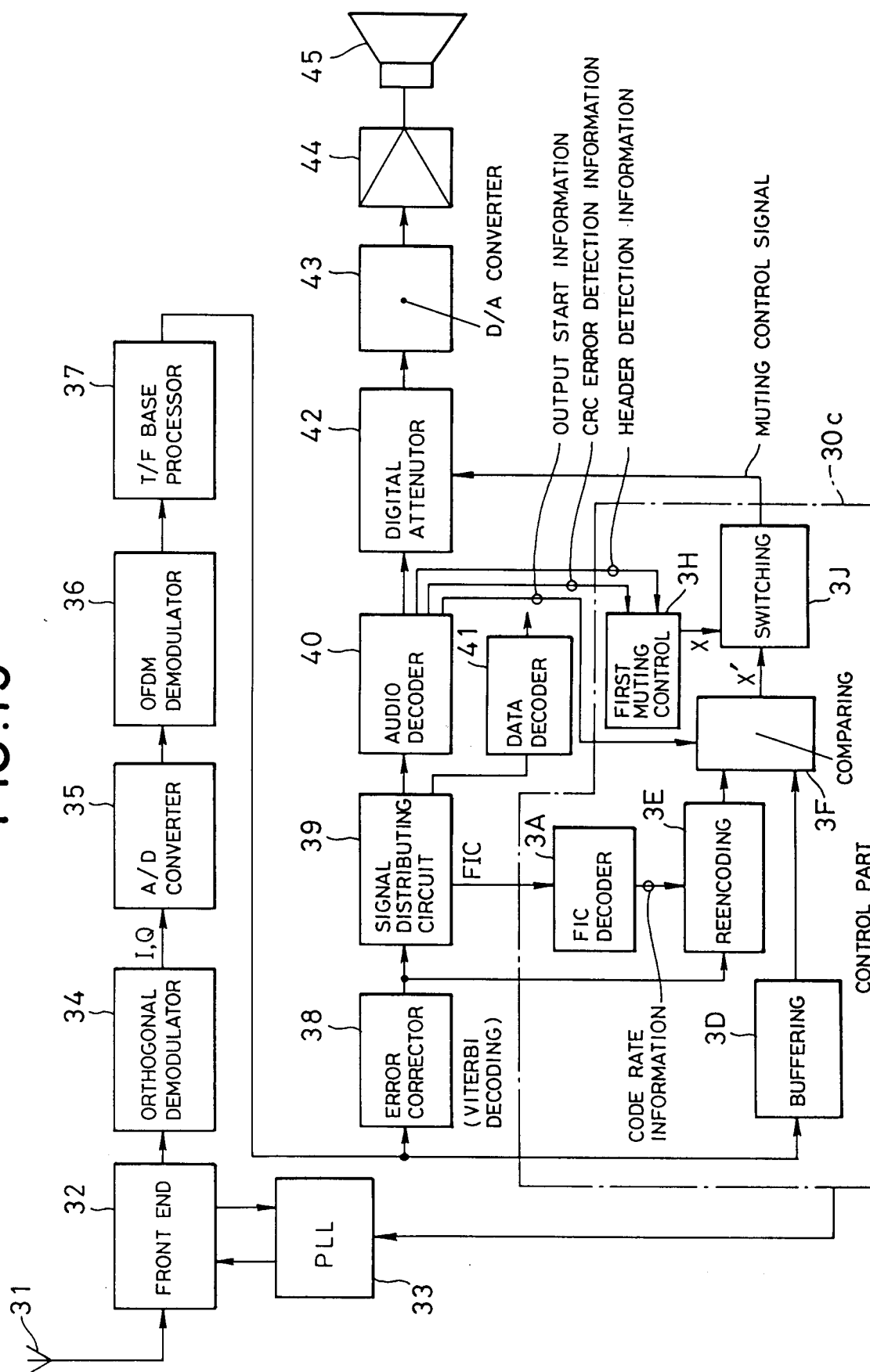
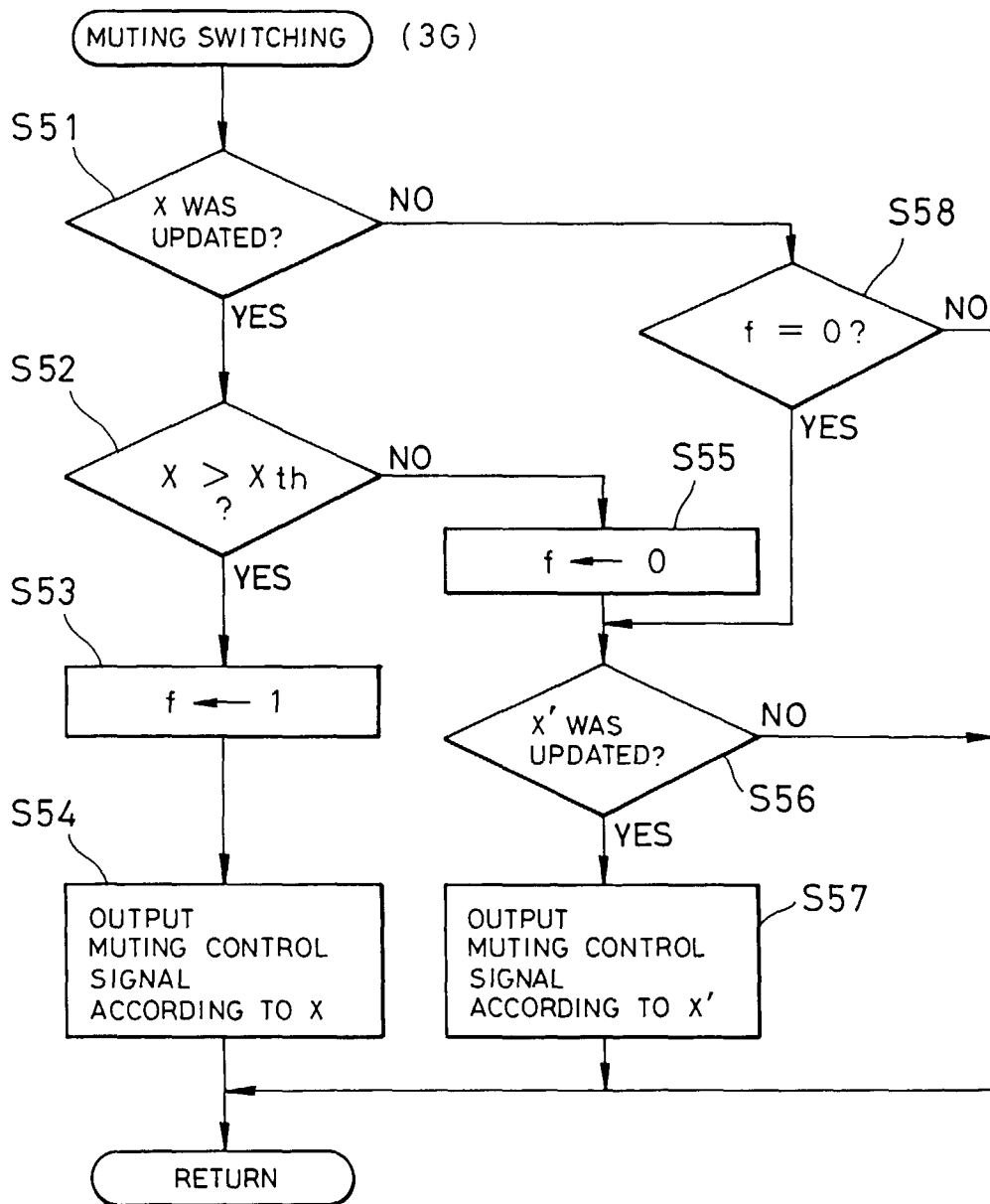


FIG. 20



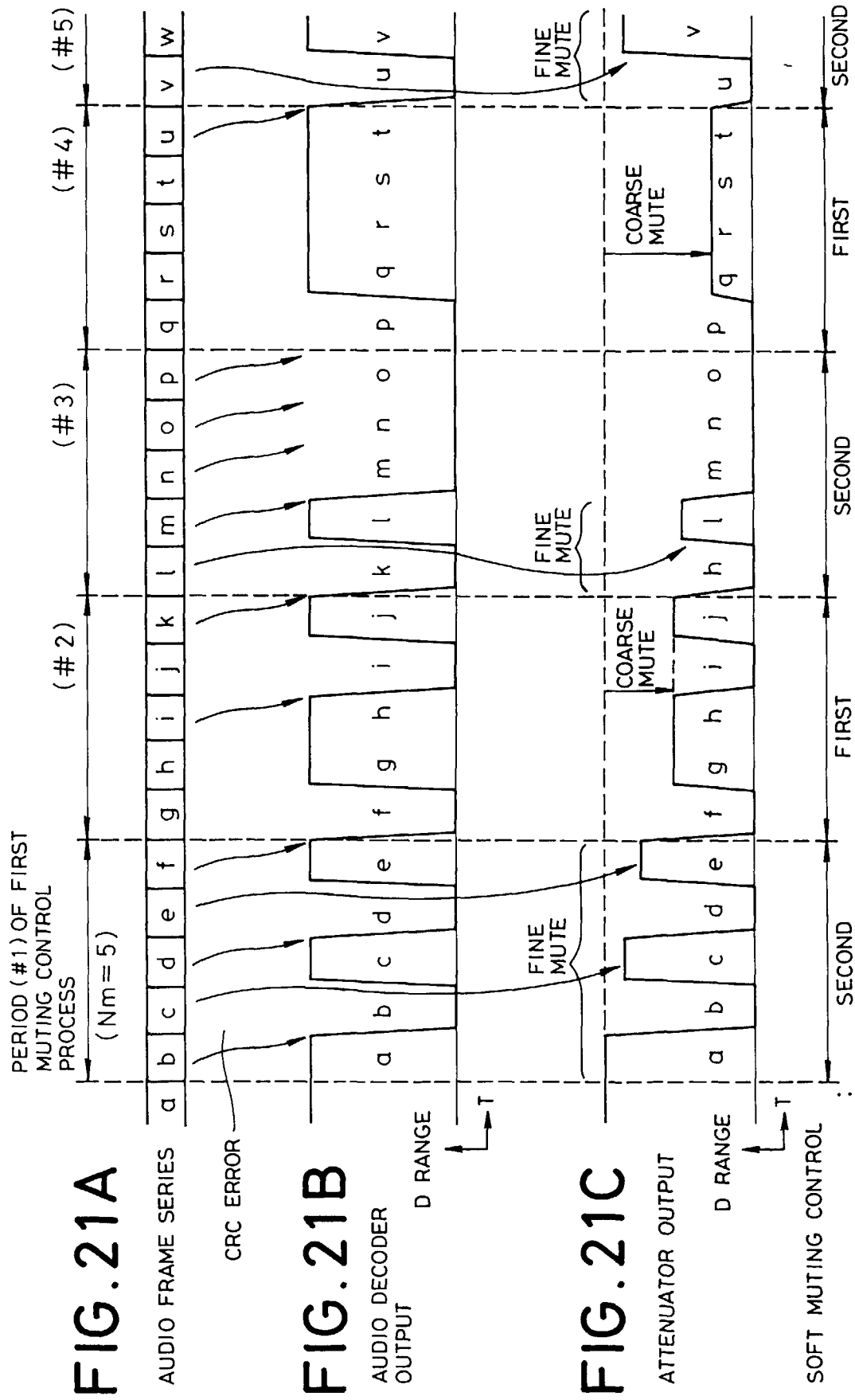


FIG.22

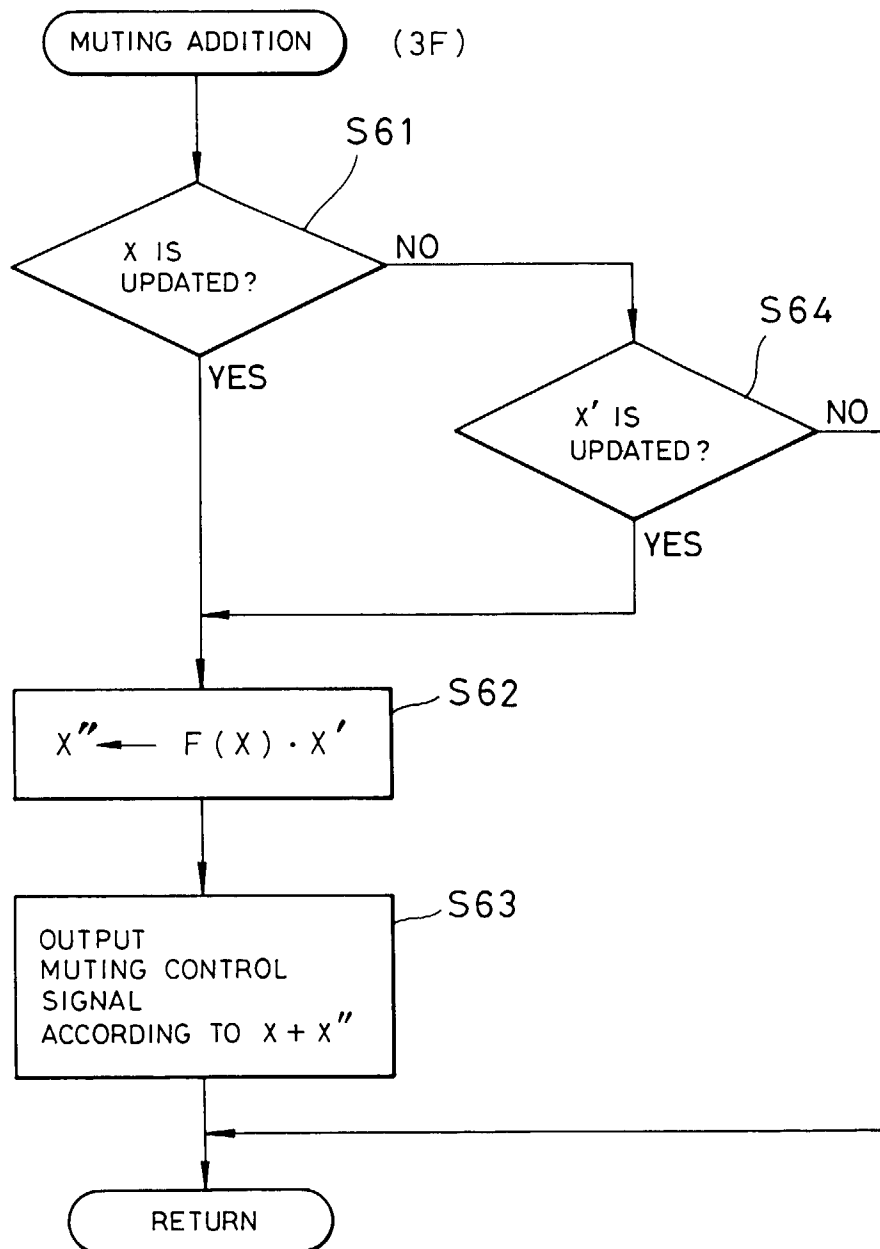


FIG. 23

