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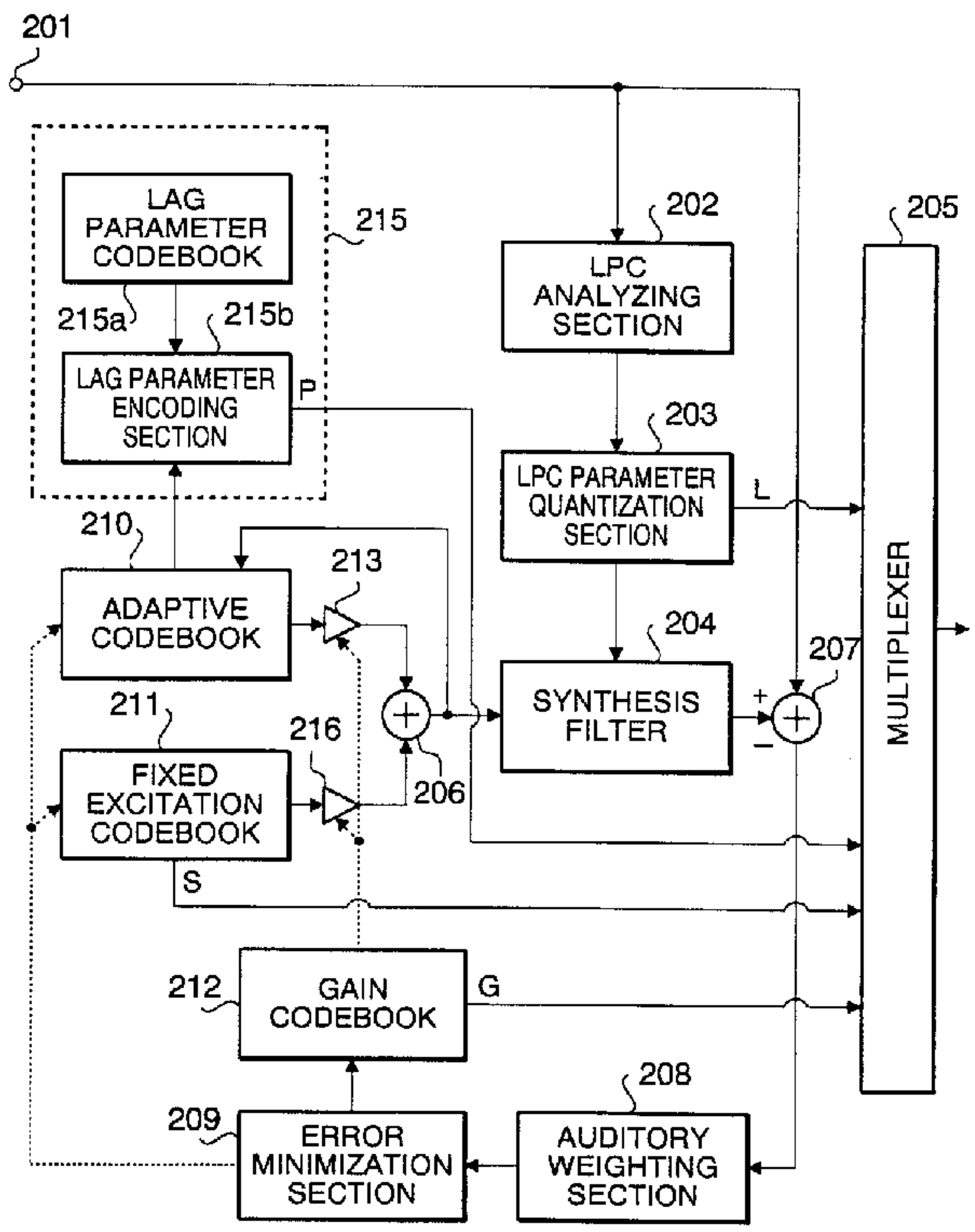
(54) PROCEDE ET DISPOSITIF DE CODAGE A RETARD PARAMETRAL ET PROCEDE DE PREPARATION DE TABLE DE CODAGE

(54) METHOD AND DEVICE FOR CODING LAG PARAMETER AND CODE BOOK PREPARING METHOD



(57) L'invention porte sur un codeur (215b) à retard paramétral qui génère un code correspondant à une valeur de paramètre de retard à l'aide d'une table de codage (215a). Du côté décodage, une valeur de paramètre de retard, correspondant au code du paramètre de retard généré sur la partie décodage, est décodée et

(57) A lag parameter coding means (215b) generates a code corresponding to a lag parameter value by using a lag parameter code book (215a). On the decoding side, a lag parameter value corresponding to the lag parameter code generated on the coding side is decoded by using the same lag parameter code book (215a) and outputted.

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émise par la même table de codage 215 (a). Dans la table de codage (215a) à retard paramètral, est affichée la relation entre la valeur de paramètre de retard et le code (P) correspondant. La relation est ainsi déterminée de façon à augmenter le taux avec lequel la valeur de paramètre de retard décodée, lorsque se produit une erreur binaire dans le code, dévie approximativement vers un multiple entier (y compris une fois) ou un sous-multiple entier de la secon.

In the lag parameter code book (215a), the relation between the lag parameter value and the corresponding code (P) is shown. The relation is so determined as to increase the rate at which the decoded lag parameter value of when a bit error occurs in the code deviates to approximately an integral multiple (including one time) or an integral submultiple of the decoded lag parameter value of when no bit error occurs. As a result, the auditory degradation of quality of decoded sound is suppressed even when the code has a bit error.

ABSTRACT

Lag parameter encoding section 215b finds a code corresponding to a lag parameter value using lag parameter codebook 215a. The decoding side decodes the lag parameter value corresponding to the lag parameter encoded by the encoding side using same lag parameter codebook 215a and outputs it. Lag parameter codebook 215a shows a relationship between a lag parameter value and corresponding code P and is generated in such a way that the rate of lag parameter code, which has a decoded lag parameter value with a bit error close to n times (including one time) or 1/n times(n is integer) of the decoded value without bit error, increases. This makes it possible to suppress deterioration of the perceptual quality of the decoded speech when a bit error occurs with the lag parameter.

DESCRIPTION

METHOD AND APPARATUS FOR LAG PARAMETER ENCODING AND METHOD FOR MAKING CODEBOOK

Technical Field

The present invention relates to a speech processing apparatus applicable to a digital cellular telephone and personal computer, etc. and relates in particular to a method and apparatus for lag parameter encoding, that is, encoding of lag parameters expressing pitch period or related parameter which is one of parameters indicating the features of speech signals, and a method for making codebooks used for these.

Background Art

One of important parameters expressing the features of speech signals is pitch period and lag parameters. These parameters are used as encoded parameters in speech encoding processing for efficiently encoding speech signals and synthesis parameters in speech synthesis. When transmitting or storing a lag parameter, it is necessary to encode the parameter value to a code corresponding to that value according to a specific rule.

The lag parameter encoding method for speech encoding is described in the international organization ITU-T Recommendation G.729 (8 kbps CS-ACELP speech encoding method).

The lag parameter encoded according to the Recommendation is transmitted together with codes of other encoded parameters. The lag parameter in this conventional example is a value (lag value) indicating which segment of a signal contained in a codebook called "adaptive codebook" is used when making an excitation signal used for synthesizing a decoded speech in the CS-ACELP algorithm, a speech encoding algorithm of this conventional example. This lag value T consists of integer part T1 (T1=19 to 143) and fraction part frac/3 (frac=-1, 0, 1).

This lag value T is encoded by a encoder as code P (P=0 to 255) in expression (1) below using T1 and frac above.

$$P=\begin{cases} 3\times(T1-19)+frac-1, & T1=19\sim 85, frac=-1,0,1 \\ (T1-85)+197, & T1=86\sim 143, frac=0 \end{cases} \quad \cdots (1)$$

On the other hand, decoded lag value T1 and frac are decoded by a decoder based on code P according to a rule opposite to expression (1).

The lag parameter is an amount of delay from time t1 of a speech signal to t0 preceding T1, at which a waveform is similar to the waveform at t1. That is, the lag parameter is typically a parameter indicating a pitch period in a periodic waveform and is a pitch period of speech itself. However, the lag parameter is a parameter which has a wide concept in a sense that it includes an amount of delay up to a position where a waveform is simply similar in non-periodic signal such as the speech onset.

However, in the lag parameter code obtained by the conventional lag parameter encoding method above, if a bit error occurs in the process of transmission or storage, the decoded lag value is by far different from the correct lag value free of errors, which may cause great deterioration of the decoded speech.

Generally, one of the methods for suppressing deterioration of the quality due to bit errors in a code is to provide a certain correlation between distortion like an Euclidean distance between parameter values of encoded parameters and a distance(Hamming distance) between codes indicating those parameter values and reduce an influences of a bit error.

If an Euclidean distance and differential value, etc. between those lag values are used as a measure of distortion between the parameter values of lag parameters, they are valid as long as such a value is small. However, if the value exceeds a certain value, it is no longer possible to keep the correspondence with perceptual distortion and the use of the general method above is not so effective for encoding/decoding processing of the lag parameters.

To handle such bit errors, a method of detecting bit errors and preventing use of lag values containing errors is also available, but error detection of this method itself is complicated, and moreover adding a redundant bit such as a check bit to a low bit rate communication method such as speech communication is not appropriate.

The present invention has been made in view of such a situation and it is an objective of the present invention to provide an excellent method and apparatus for encoding lag parameters and a method for making codebooks capable of suppressing, in the event of a bit error with lag parameter codes, deterioration of the perceptual speech quality caused thereby.

Disclosure of Invention

To solve the above problem, the present invention performs encoding of lag parameters using a codebook defined as shown below. The codebook is generated in such a way that the rate of lag parameter code, which has a decoded lag parameter value with a bit error close to n times (including one time) or 1/n times(n is integer) of the decoded value without bit error, increases.

The codebook is generated in such a way that a total sum of distortion of decoded values between codes with Hamming distance within a prescribed number of bits in the codebook is minimized or nearly minimized, and by using such a distortion measure that distortion is measured smaller between one decoded lag parameter value and another value with n times or 1/n times (n is integer) of that value.

As a result, the codebook is generated in such a way that the rate of lag parameter code, which has a decoded lag parameter value with a bit error close to n times (including one time) or 1/n times(n is integer) of the decoded value without bit error, increases, allowing the speech signal to be encoded/decoded with less deterioration of the perceptual speech quality.

Brief Description of Drawings

FIG.1 is a schematic block diagram of a radio transmitter to which a method and apparatus for lag parameter encoding of Embodiment 1 of the present invention is applied;

FIG.2 is a schematic block diagram of a speech encoding section of the radio communication apparatus of Embodiment 1;

FIG.3 is a block diagram of the main part of the speech encoding section of the radio communication apparatus of Embodiment 1;

FIG.4 is a block diagram of the main part of a speech decoding section of the radio communication apparatus of Embodiment 1; and

FIG.5 is a drawing showing the procedure for making a codebook applicable to the radio communication apparatus of Embodiment 1.

Best Mode for Carrying out the Invention

(Embodiment 1)

With reference to FIG.1 to FIG.5, Embodiment 1 of the present invention is explained below.

FIG.1 is a schematic block diagram of a radio transmitter to which the present invention is applied.

Speech transmission processing is carried out as follows: A speech signal input from microphone 101 is converted from analog to digital by A/D converter 102, output to speech encoding section 103 and encoded according to a CELP algorithm, for example. The encoded output is modulated by modulator/demodulator 104 according to a CDMA system, etc., and transmitted via radio transmission section 105 and antenna 106.

Speech reception processing is carried out as follows: A modulated signal received via antenna 107 and radio reception section 108 is demodulated by modulator/demodulator 104, then decoded by speech decoding section 109, converted from digital to analog by D/A converter 110 and output from speaker 111 as a speech.

The present invention is applied to part of adaptive codebook search processing used in speech encoding section 103 and speech decoding section 109 of the radio communication apparatus above.

FIG.2 is a schematic block diagram of speech encoding section 103 of the radio communication apparatus and shows a general configuration of a CELP type speech encoder/decoder. The A/D converted speech signal is input from terminal 201 and output to LPC analyzing section 202. LPC analyzing section 202 carries out a linear predictive analysis based on the input speech signal and outputs linear predictive coefficients. LPC parameter quantization section 203 quantizes the linear predictive coefficients(L) and outputs the quantization result to synthesis filter 204 and multiplexer 205.

Synthesis filter 204 constitutes a filter with a given characteristic according to the linear predictive coefficients above, filters an excitation signal input from adder 206 and outputs the result to adder 207. This adder 207 calculates an error between the input speech signal from terminal 201 and the output from synthesis filter 204 and outputs the error signal to perceptual weighting section 208. Perceptual weighting section 208 carries out weighting processing corresponding to the perceptual sense of the error signal and outputs the result to error minimization section 209.

Error minimization section 209 selects code vectors for adaptive codebook 210 and fixed excitation codebook 211 so that the error signal output from perceptual weighting section 208 may be minimized and selects a gain for gain codebook 212 as well.

Adaptive codebook 210 is an excitation signal table which stores past excitation vectors and selectively outputs a specific code vector selected by error minimization section 209. Multiplier 213 multiplies the output by a gain select by gain codebook 212 and outputs the result to adder 206.

By the way, this adaptive codebook 210 includes a buffer storing a history for a certain period of the excitation vector finally determined as the output of adder 206 and outputs a lag value indicating which segment of the signal sequence stored in said buffer should be extracted to lag parameter encoding section 215 according to the code vector selected by error minimization section 209. This lag parameter encoding section 215 includes lag parameter codebook 215a generated beforehand according to a prescribed rule and lag parameter encoding section 215b and encodes the lag value of adaptive codebook 210 according to a certain rule and outputs it to multiplexer 205. This lag parameter encoding section 215 will be described in detail later.

Fixed excitation codebook 211 selectively outputs a specific fixed excitation code vector selected by error minimization section 209. Multiplier 216 multiplies the output by a gain set by gain codebook 212 and outputs the result to adder 206.

Adder 206 finds a sum of the outputs of multiplier 213 and multiplier 216 and outputs it to synthesis filter 204 as an excitation vector. At the same time, the output is fed back to adaptive codebook 210 and accumulated.

In this way, error minimization section 209 measures error signals for all excitation vectors stored in adaptive codebook 210 and outputs the output (P) of lag parameter encoding section 215b, the output (S) of fixed excitation codebook 211 and the output (G) of gain codebook 212 corresponding to a minimum value of the error signal from perceptual weighting section 208 to multiplexer 205. Multiplexer 205 multiplexes quantized linear predictive coefficient (L) and outputs (P), (S) and (G) above and outputs the result to modulator 104 in FIG.1.

Speech decoding section 110 of the radio communication apparatus (FIG.1) also consists of a general CELP type speech decoder, but its explanation is omitted here.

Then, lag parameter encoding section 215 to which the present invention is applied is explained in detail below.

FIG.3 and FIG.4 show a configuration of the main part of lag parameter encoding section 215 to which the present invention is applied and FIG. 3 shows a functional block on the lag parameter encoding section side and FIG.4 shows a functional block on the lag parameter decoding section side. Such a lag parameter encoding section is not limited to cellular telephones but is applicable to all apparatuses carrying out speech encoding/decoding.

As shown in FIG.3, lag parameter encoding section 215 comprises lag parameter codebook 215a and lag parameter encoding section 215b that encodes lag values with reference to this lag parameter codebook 215a. Lag parameter codebook 215a is a table that stores input lag values with corresponding output codes and is generated beforehand according to a certain rule.

In the same way, as shown in FIG.4, the lag parameter decoding section of the speech decoder comprises same lag parameter codebook 215a as that in the lag parameter encoding section above and lag parameter decoding section 401 that decodes lag parameters corresponding to the encode received/input with reference to this lag parameter codebook 215a.

Lag parameter encoding section 215 configured as shown above is further explained in detail.

Lag parameter codebook 215a is a table that shows a relationship between lag parameter value T and corresponding code P. For example, if the codebook size is N, lag value T corresponding to code P (=0 to N-1) is stored. Furthermore, it is also possible to find intermediate code P0 (0 to N-1) from a calculation expression such as expression (1) for lag parameter encoding in the ITU-T Recommendation G.729 (8 kbps CS-ACELP) mentioned in the conventional technology and store a table of correspondence with final code P (=0 to N-1) corresponding to P0.

Lag parameter codebook 215a of the present invention is characterized by a configuration such that the codebook is generated in such a way that the rate of lag parameter code, which has a decoded lag parameter value with a bit error close to n times (including one time) or 1/n times(n is integer) of the decoded value without bit error, increases. The method of its generation will be explained later.

A lag parameter is a parameter related to a pitch period included in a speech signal. There are cases where a correct lag value is not obtained due to a bit error, etc. However, the inventor of the present invention has found that if a wrong decoded lag value is close to n times(including one time) or 1/n times (n is integer) of the correct lag value, perceptual deterioration can be relatively small. This is because the spectrum of the speech signal decoded or synthesized using that wrong lag value includes the frequency component of the correct pitch period as its part as long as the above condition is satisfied.

As shown above, Embodiment 1 is capable of reducing deterioration of the perceptual speech quality when a bit error occurs with a code taking advantage of the characteristic that perceptual deterioration is small in the case of a lag parameter with a bit error having a value close to n times(including one time) or 1/n times (n is integer) of the correct decoded value.

Then, the method of making the lag parameter codebook above used in the present invention is explained. This lag parameter codebook is generated in such a way that the rate of lag parameter code, which has a decoded lag parameter value with a bit error close to n times (including one time) or 1/n times(n is integer) of the decoded value without bit error, increases.

FIG.5 shows the processing procedure for making the lag parameter codebook above.

First, initial codebook Table(i) (i=0 to N-1; N: codebook size) is set in step 501. Here, Table(i) denotes a decoded value (either a scalar value or vector value). If this codebook is a lag parameter codebook, Table(i) may be set to indicate intermediate code P0 for code i as explained in lag parameter codebook 215a of Embodiment 1. Furthermore, the correspondence between a code and decoded value in the initial codebook can be determined arbitrarily.

Then, in step 502, for all combinations of codes whose Hamming distance dH is within a specific number of bits (suppose it is MB) ($dH \leqq MB$) in Table(i), distortion of the decoded value between those combined codes is calculated one by one and total sum D0 is obtained.

Here, distortion of the decoded value between codes depends on the parameter indicated by the code, but an Euclidean distance between decoded values or the like is used. The method of expressing a distortion measure of lag parameters is one of the features of the present invention. It will be described further in Embodiment 2.

Then, in step 503, a code pair i_a and i_b whose

Hamming distance dH exceeds said specific number of bits MB (dH>MB) is randomly selected from codebook Table (i). In step 504, after the decoded values are mutually exchange between said code pair, total sum D of distortion of decoded values between codes with a Hamming distance within said specific number of bits is calculated.

Then in step 505, it is judged whether total sum D of distortion in step 504 is smaller than total sum Do of distortion calculated before. If it is smaller, the decoded values are exchange between said code pair and the total sum of distortion is updated in step 506.

In step 507, the convergence of said total sum D0 of distortion is judged and the operations from said steps 503 to 507 are repeated until said total sum of distortion converges.

Making the lag parameter codebook using the processing above can reduce the total sum of distortion measure of decoded values between codes within a specific Hamming distance, making the decoded values when a bit error occurs with a code closer to the correct decoded value, suppressing deterioration of the perceptual speech quality.

Especially, by limiting the minimization of the total sum of distortion to between codes with a Hamming distance within a specific number of bits, it is possible to suppress deterioration more effectively when a bit error occurs with fewer bits occurs. By randomly selecting code pair i_a and i_b whose Hamming distance exceeds a specific number of bits, it is possible to achieve higher efficiency and reduce the total sum of distortion. Thus, even if a bit error occurs it is possible to suppress deterioration of the perceptual speech quality.

In step 503 above, code pair i_a and i_b selected randomly from codebook Table (i) is limited to those whose Hamming distance exceeds a specific number of bits, but the present invention is not limited to this.

(Embodiment 2)

Embodiment 2 is implemented on the same hardware and software as those in Embodiment 1. The difference from the method for making the lag parameter codebook applied in Embodiment 1 is a change in the distortion measure.

The procedure for making a codebook is the same as that in FIG.5 shown in Embodiment 1. What is different from Embodiment 1 is the use of a measure shown in expression (2) as distortion of decoded values between codes used in steps 502 and 504.

$$d(fa,fb)=\min(w1\times d0(fb,fa),\ w2\times d0(fb,2\times fa),\ w3\times d0(fb,3\times fa))$$

where $fa=Fs/Ta(Hz)$ $\cdots$ (2)

$fb=Fs/Tb(Hz)$

$fb\geqq fa$

$d0(fx,fy)=|fx-fy|/(fx\times fy)^{1/2}$

Where, Ta and Tb are decoded lag values (unit: sample) of target codes i_a and i_b; fa and fb, frequency values (Hz) for Ta and Tb; Fs, sampling frequency (Hz); and d(fa, fb), distortion of decoded values between code pairs.

Expression (2) does not simply express distortion of lag parameter values by anything similar to an Euclidean distance. Expression (2) is an example of definition taking account of a difference between one lag value and n times(n is integer) of another lag value(w1, w2 and w3 are weighting constants corresponding to distortion from n times(n is integer) of that value) and another definition implementing a similar concept can also be used.

Using such a distortion measure, one decoded value of codes with a Hamming distance within a specific number of bits becomes a value close to n times(n is integer) of the other decoded value. As already explained, the lag parameter is a parameter related to a pitch period contained in the speech signal. If the decoded lag value is close to n times(including one time) or 1/n times (n is integer) of the correct lag value due to a bit error, etc., the spectrum of the speech signal decoded or synthesized using that value contains the frequency component of the correct pitch period as a part, and therefore perceptual deterioration can be relatively small.

It is possible to further reduce the total sum of distortion by defining a value close to such n times(n is integer) as small distortion and making a codebook through minimization of distortion by limiting to between codes with a Hamming distance within a specific number of bits. Therefore, if a lag parameter codebook is generated by the method described above, it is possible to suppress deterioration of the perceptual quality more effectively in the event of a bit error even for parameters like lag parameters susceptible to dislocation of decoded values due to the error.

As described above, the invention of the lag parameter encoding method of the present invention is a method for encoding lag parameters which are parameters for encoding speech signals, and is intended to encode lag parameters using a lag parameters codebook generated in such a way that the rate of lag parameter code, which has a decoded lag parameter value with a bit error close to n times (including one time) or 1/n times(n is integer) of the decoded value without bit error, increases.

In the invention of the lag parameter decoding method of the present invention, the lag parameter coded by the encoding side using the encoding method described above is decoded using the same lag parameter codebook as that on the encoding side.

As shown above, it is possible to suppress deterioration of the perceptual speech quality when a bit error occurs with a code by taking advantage of the characteristic that perceptual deterioration is small in the case of a lag parameter with a bit error having a value close to n times(including one time) or 1/n times (n is integer) of the correct decoded value by using a codebook generated in such a way that the rate of lag parameter code, which has a decoded lag parameter value with a bit error close to n times (including one time) or 1/n times(n is integer) of the decoded value without bit error, increases.

Furthermore, the invention of the codebook generation method of the present invention is the method for making a codebook in such a way that a total sum of distortion of decoded values between codes with Hamming distance within a prescribed number of bits in the codebook is minimized or nearly minimized. Setting the decoded value when a bit error occurs with a code to a value close to the correct decoded value can suppress deterioration of the perceptual speech quality and limiting the target of minimization of the total sum of distortion to between codes with a Hamming distance within a specific number of bits can more effectively suppress deterioration of the speech quality when a bit error occurs with fewer bits.

When making the codebook above, the initial codebook can be designed to comprise the steps of calculating a total sum of distortion of decoded values between codes with a Hamming distance within a prescribed number of bits, randomly selecting a code pair from the codebook, calculating a total sum of distortion of decoded values between codes with a Hamming distance within a prescribed number of bits after exchanging the decoded values between said code pair, exchanging said decoded values and updating the total sum of distortion if said total sum of distortion of decoded values is smaller than said total sum of distortion calculated before and judging the convergence of said total sum of distortion, and repeat said steps of randomly selecting a code pair, exchanging decoded values and updating a total sum of distortion and judging the convergence of a total sum of distortion until said total sum of distortion converges.

It is also preferable to use such a distortion measure that distortion is measured smaller between one decoded lag parameter value and another value with n times or 1/n times (n is integer) of that value, and using such a distortion measure makes it possible to suppress deterioration of the perceptual speech quality when a bit error occurs with a code using the characteristic that perceptual deterioration is small in the case of a lag parameter with a bit error having a value close to n times(including one time) or 1/n times (n is integer) of the correct decoded value.

Furthermore, it is possible to realize a lag parameter encoding/decoding method for encoding/decoded lag parameters using the encoding and decoding methods above or a codebook generated by one of the codebook generation methods above.

It is also possible to implement the present invention as a speech encoder comprising a codebook indicating the correspondence between the parameter values of lag parameters which are encoded parameters of speech signals and codes and a lag parameter encoder that encodes lag parameters using said codebook. It is further possible to implement the present invention as a speech decoder comprising a lag parameter decoder that decodes lag parameters codes encoded by the above encoder using the same codebook as that on the encoding side. Furthermore, it is possible to implement the lag parameter encoder/decoder with a single apparatus.

It is also possible to implement the encoding method above by computer software. More specifically, it is possible to configure a system that comprises a computer-readable medium and a program instruction means for instructing a computer processor to encode lag parameters using a lag parameter codebook generated in such a way that the rate of lag parameter code, which has a decoded lag parameter value with a bit error close to n times (including one time) or 1/n times(n is integer) of the decoded value without bit error, increases, wherein said program instruction means is stored in said medium in an executable format and loaded to a computer memory when executed by said processor to operate the computer.

Of course, the decoding method above can also be implemented by computer software likewise.

It is also possible to use the encoding software above by storing it in various types of storage medium. It is a mechanically readable storage medium storing a program that instructs the computer to encode lag parameters using a lag parameter codebook generated in such a way that the rate of lag parameter code, which has a decoded lag parameter value with a bit error close to n times (including one time) or 1/n times(n is integer) of the decoded value without bit error, increases. It is then downloaded to the computer to operate the computer, thus implementing the encoding method above.

Of course, the decoding software above can also be used by storing it in various types of storage medium likewise.

The present invention can also be implemented as a codebook generation apparatus that comprises a computer-readable medium and a program instruction means that instructs a computer processor to generate a codebook in such a way that a total sum of distortion of decoded values between codes with a Hamming distance within a prescribed number of bits in the codebook is set to a minimum value or a value close to a minimum value, wherein said program instruction means is stored in said storage medium in an executable format and loaded to a computer memory when executed by said processor to operate the computer.

The present invention is applicable when no error detection is performed, and of course can be used with error detection as well. It is further applicable to all speech encoding/decoding methods carrying out encoding of lag parameters.

This application is based on the Japanese Patent Application No.HEI 10-29332 filed on January 27, 1998, entire content of which is expressly incorporated by reference herein.

Industrial Applicability

The encoder, decoder and encoding and decoding methods of the present invention are applicable to a wide range of equipment equipped with a speech encoder and speech decoder. The use of the present invention for a radio communication apparatus such as digital cellular telephone is particularly preferable because it can efficiently suppress deterioration of the perceptual speech quality.

What is claimed is:

1. A lag parameter encoding method, the method for encoding lag parameters that are encoded parameters of speech signals, which encodes lag parameters using a lag parameter codebook generated in such a way that the rate of lag parameter code, which has a decoded lag parameter value with a bit error close to n times (including one time) or 1/n times(n is integer) of the decoded value without bit error, increases.

2. A lag parameter decoding method, which decodes lag parameter codes encoded by the encoding method according to claim 1 on the encoding side using the same lag parameter codebook as that on the encoding side.

3. A codebook generation method, which generates a codebook in such a way that a total sum of distortion of decoded values between codes with Hamming distance within a prescribed number of bits in the codebook is minimized or nearly minimized.

4. The codebook generation method according to claim 3, comprising the steps of:

calculating a total sum of distortion of decoded values between codes with a Hamming distance within a prescribed number of bits in an initial codebook;

randomly selecting a code pair from a codebook;

calculating a total sum of distortion of decoded values between codes with a Hamming distance within said prescribed number of bits after exchanging the decoded values between said code pair;

exchanging said decoded values and updating a total sum of distortion if said total sum of distortion of decoded values is smaller than said total sum of distortion calculated before; and

judging the convergence of said total sum of distortion,

wherein said steps of randomly selecting a code pair, exchanging decoded values and updating a total sum of distortion and judging the convergence of a total sum of distortion are repeated until said total sum of distortion converges.

5. The codebook generation method according to claim 3, which uses such a distortion measure that distortion is measured smaller between one decoded lag parameter value and another value with n times or 1/n times (n is integer) of that value.

6. The codebook generation method according to claim 4, which uses such a distortion measure that distortion is measured smaller between one decoded lag parameter value and another value with n times or 1/n times (n is integer) of that value.

7. A lag parameter encoding/decoding method, which performs encoding/decoding of lag parameters using the encoding method according to claim 1 and the decoding method according to claim 2.

8. A lag parameter encoding/decoding method, which performs encoding/decoding of lag parameters using the codebook generated by the codebook generation method

according to claim 3.

9. A lag parameter encoding/decoding method, which performs encoding/decoding of lag parameters using the codebook generated by the codebook generation method according to claim 4.

10. A lag parameter encoding/decoding method, which performs encoding/decoding of lag parameters using the codebook generated by the codebook generation method according to claim 5.

11. A lag parameter encoding/decoding method, which performs encoding/decoding of lag parameters using the codebook generated by the codebook generation method according to claim 6.

12. A lag parameter encoder, comprising:

a codebook that shows the correspondence between parameter values of lag parameters which are encoded parameters of speech signals and codes; and

a lag parameter encoder that encodes lag parameters using said codebook,

wherein said codebook is generated in such a way that the rate of lag parameter code, which has a decoded lag parameter value with a bit error close to n times (including one time) or 1/n times(n is integer) of the decoded value without bit error, increases.

13. A lag parameter decoder, comprising a lag parameter decoder that decodes the lag parameter codes encoded by the encoder according to claim 12 on the encoding side using the same codebook as that on the encoding side.

14. A lag parameter encoder/decoder, which performs encoding/decoding of lag parameters using the encoder according to claim 12 and the decoder according to claim 13.

15. A speech encoder/decoder, comprising the lag parameter encoder/decoder according to claim 14.

16. A radio communication apparatus, comprising the speech encoder/decoder according to claim 15.

17. A encoder, comprising:

a computer-readable medium; and

a program instruction means for instructing a computer processor to encode lag parameters using a lag parameter codebook generated in such a way that the rate of lag parameter code, which has a decoded lag parameter value with a bit error close to n times (including one time) or 1/n times(n is integer) of the decoded value without bit error, increases,

wherein said program instruction means is stored in said medium in an executable format and loaded to a computer memory when executed by said processor to operate the computer.

18. A decoder, comprising:

a computer-readable medium; and

a program instruction means for instructing a computer processor to decode lag parameters using a lag parameter codebook generated in such a way that the rate of lag parameter code, which has a decoded lag parameter value with a bit error close to n times (including one time) or 1/n times(n is integer) of the decoded value without bit error, increases,

wherein said program instruction means is stored in said medium in an executable format and loaded to a computer memory when executed by said processor to operate the computer.

19. A codebook generation apparatus, comprising:

a computer-readable medium; and

a program instruction means for instructing a computer processor to generate a codebook in such a way that a total sum of distortion of decoded values between codes with Hamming distance within a prescribed number of bits in the codebook is minimized or nearly minimized,

wherein said program instruction means is stored in said medium in an executable format and loaded to a computer memory when executed by said processor to operate the computer.

FIG. 1

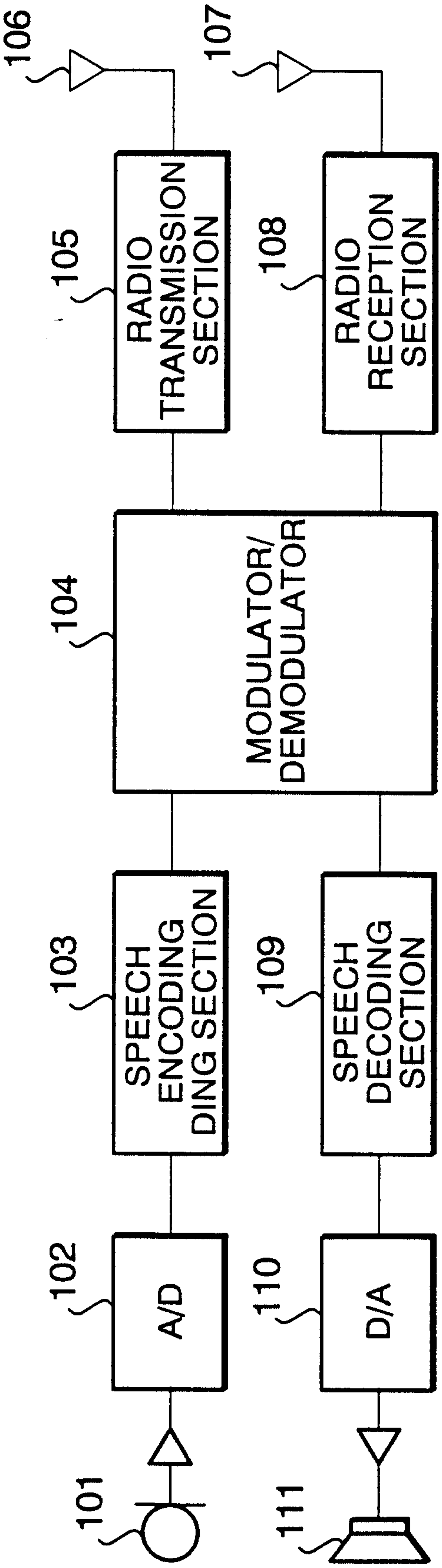

101
102
A/D
103
SPEECH ENCODING DING SECTION
104
MODULATOR/ DEMODULATOR
105
RADIO TRANSMISSION SECTION
106
107
108
RADIO RECEPTION SECTION
109
SPEECH DECODING SECTION
110
D/A
111

FIG. 2

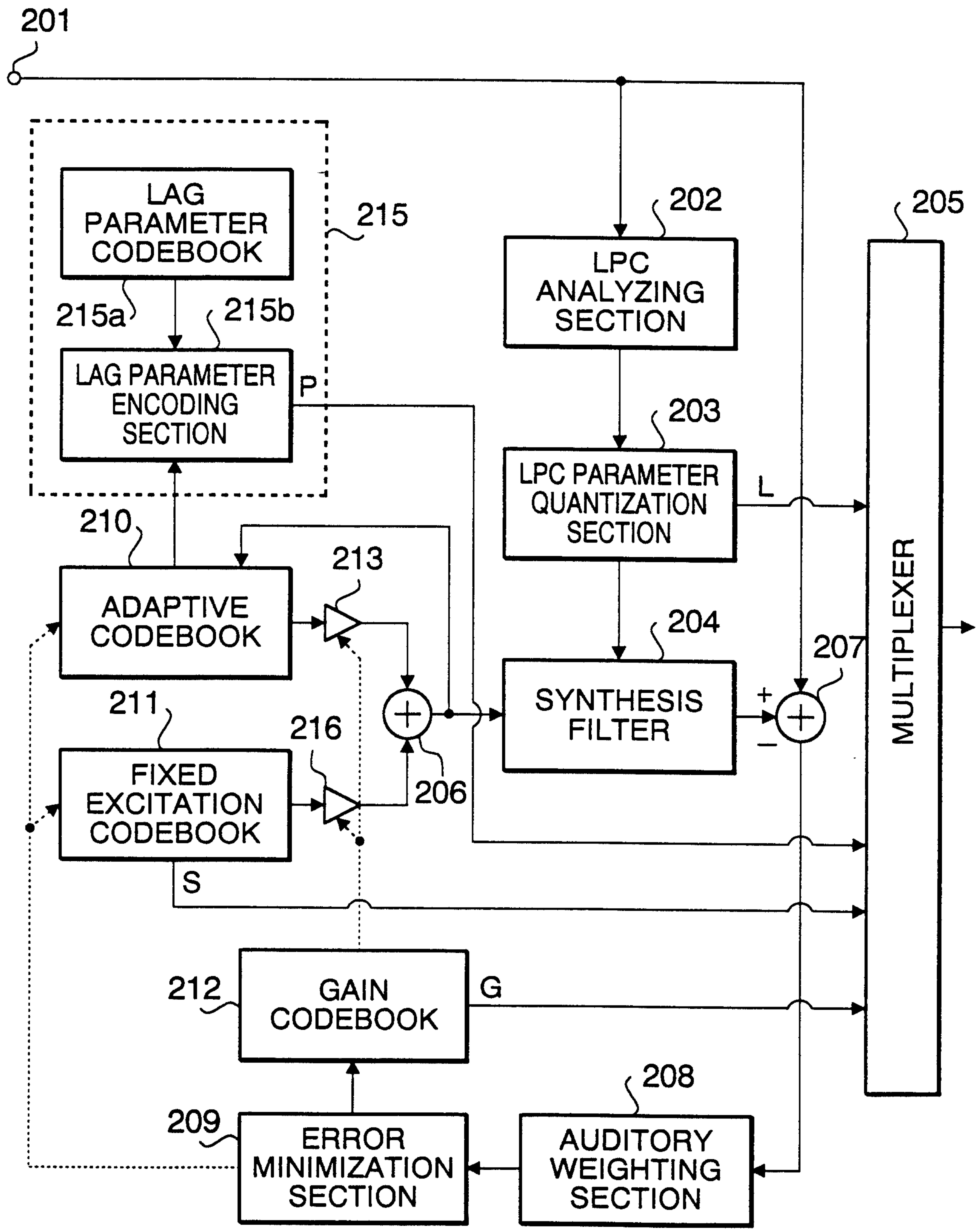

201
LAG PARAMETER CODEBOOK
215
215a
215b
LAG PARAMETER ENCODING SECTION
P
210
ADAPTIVE CODEBOOK
213
211
216
FIXED EXCITATION CODEBOOK
206
S
212
GAIN CODEBOOK
G
209
ERROR MINIMIZATION SECTION
202
LPC ANALYZING SECTION
203
LPC PARAMETER QUANTIZATION SECTION
L
204
SYNTHESIS FILTER
207
+
−
208
AUDITORY WEIGHTING SECTION
205
MULTIPLEXER

FIG. 3

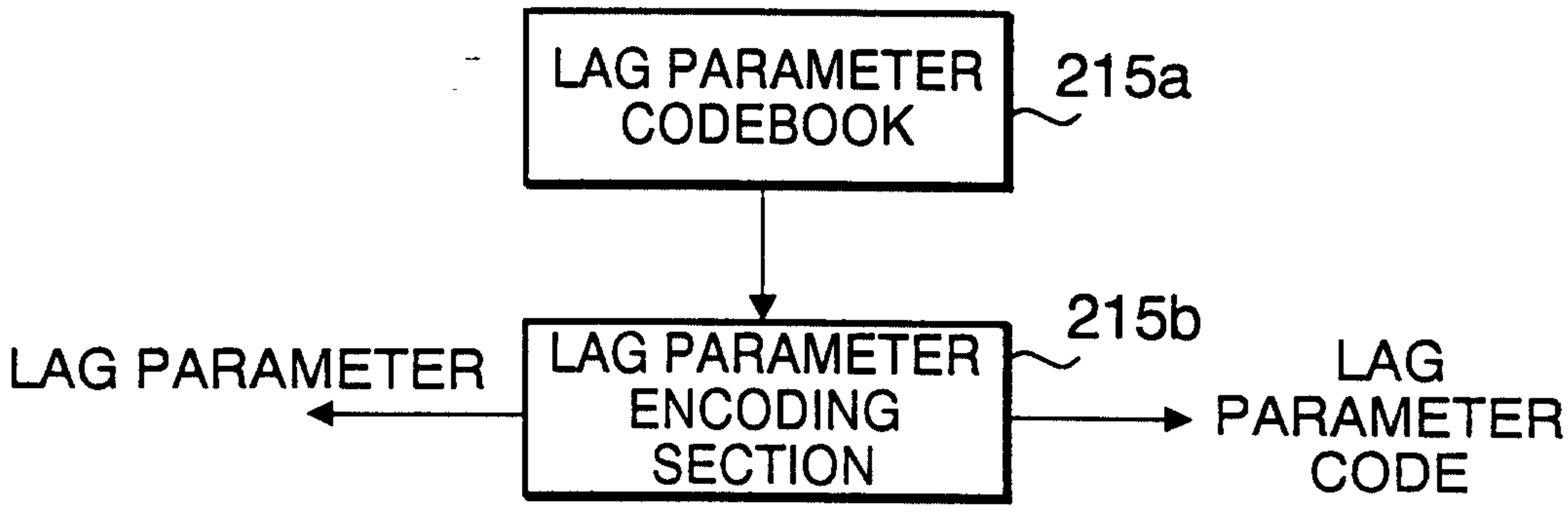

LAG PARAMETER CODEBOOK
215a
215b
LAG PARAMETER
LAG PARAMETER ENCODING SECTION
LAG PARAMETER CODE

FIG. 4

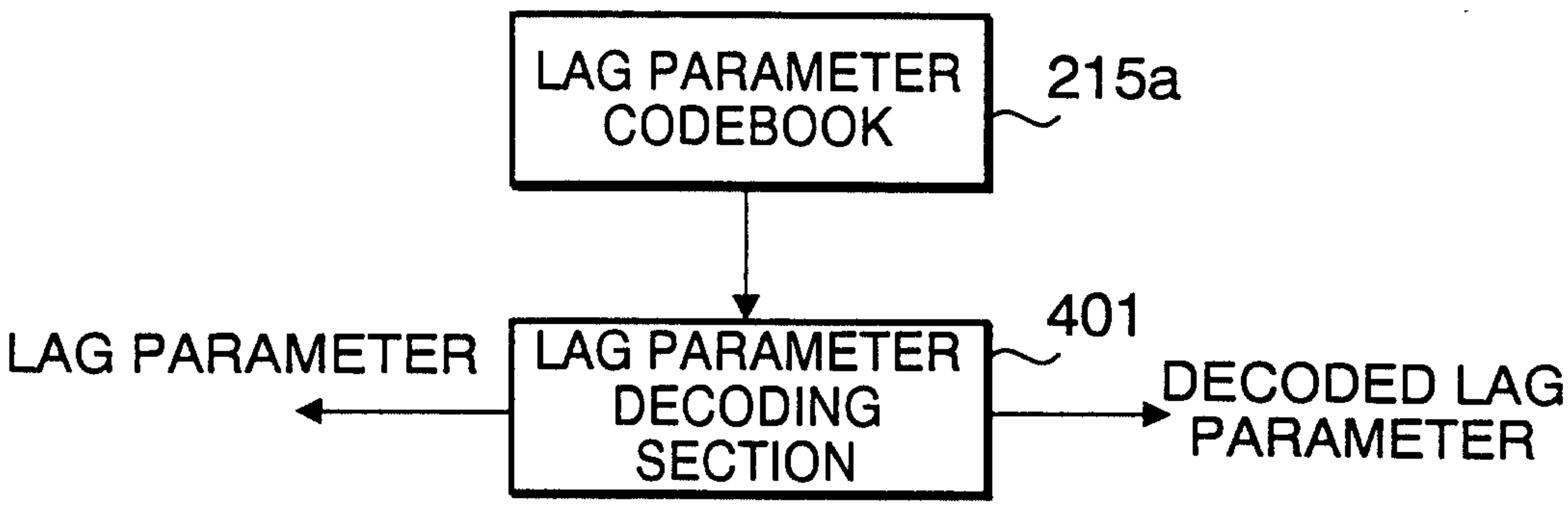
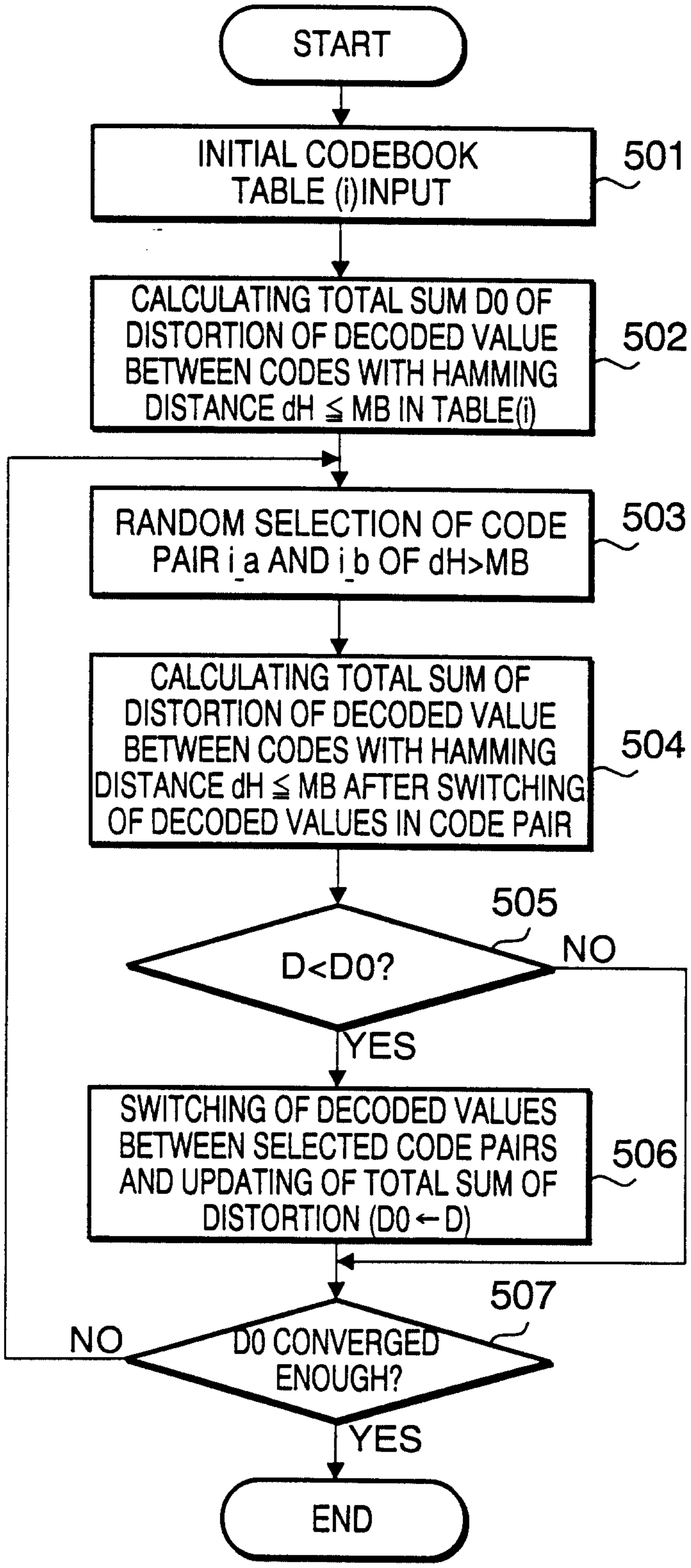

LAG PARAMETER CODEBOOK
215a
401
LAG PARAMETER
LAG PARAMETER DECODING SECTION
DECODED LAG PARAMETER

FIG. 5


START
INITIAL CODEBOOK TABLE (i)INPUT
501
CALCULATING TOTAL SUM D0 OF DISTORTION OF DECODED VALUE BETWEEN CODES WITH HAMMING DISTANCE dH ≦ MB IN TABLE(i)
502
RANDOM SELECTION OF CODE PAIR i_a AND i_b OF dH>MB
503
CALCULATING TOTAL SUM OF DISTORTION OF DECODED VALUE BETWEEN CODES WITH HAMMING DISTANCE dH ≦ MB AFTER SWITCHING OF DECODED VALUES IN CODE PAIR
504
505
D<D0?
NO
YES
SWITCHING OF DECODED VALUES BETWEEN SELECTED CODE PAIRS AND UPDATING OF TOTAL SUM OF DISTORTION (D0←D)
506
507
D0 CONVERGED ENOUGH?
NO
YES
END