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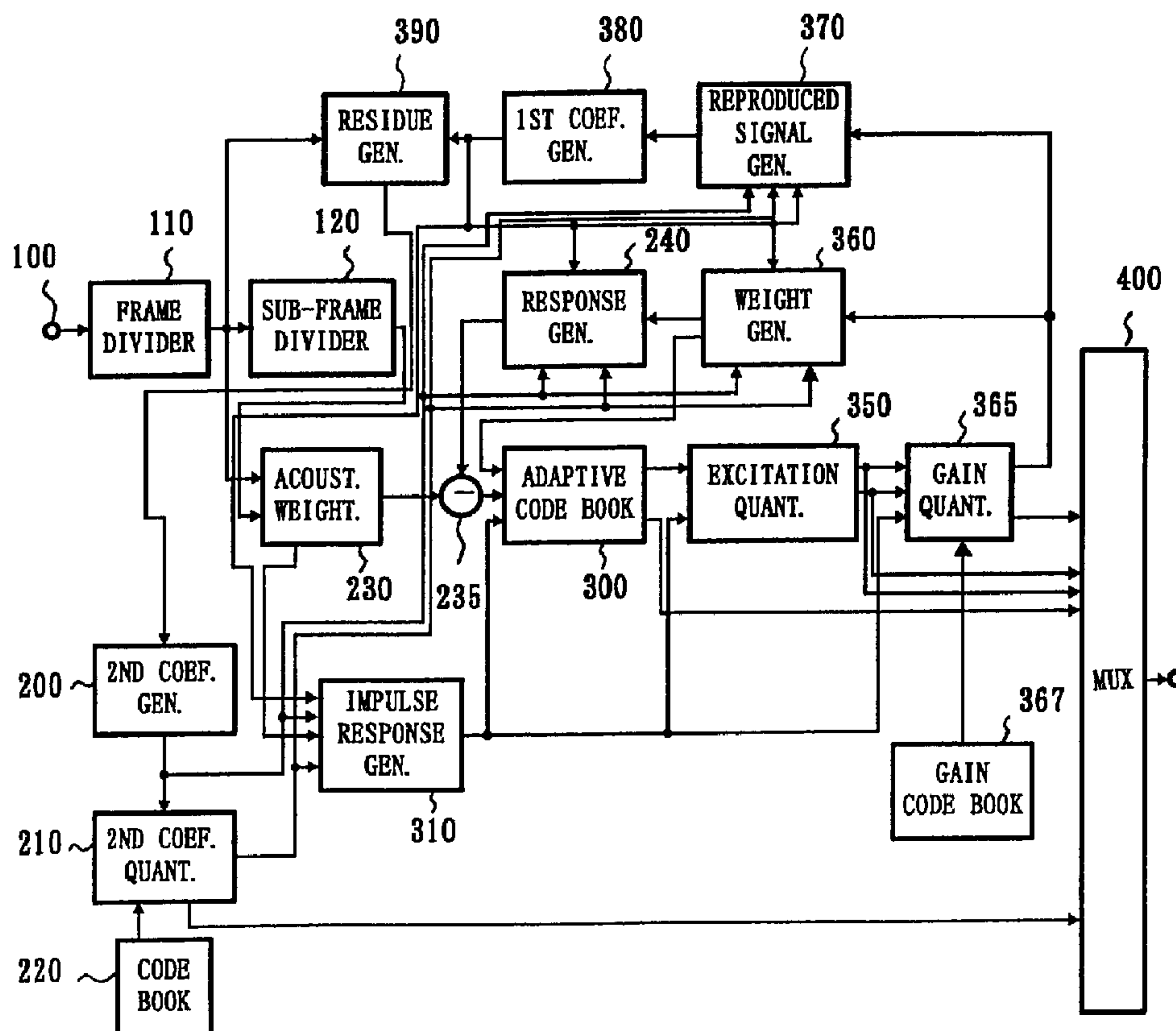
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(54) Titre : CODEUR DE PAROLE DE GRANDE QUALITE ET METHODE DE CODAGE

(54) Title: HIGH QUALITY SPEECH CODER AND CODING METHOD



(57) Abrégé/Abstract:

A first coefficient generating unit derives, from past reproduced speech signal, a first coefficient signal representing a spectral characteristic of the past reproduced speech signal. A residual signal generating unit derives, from speech signal for each frame, a predicted residue signal by using the first coefficients. A second coefficient generator derives second coefficients



(57) Abrégé(suite)/Abstract(continued):

representing a spectral characteristic of the predicted residue signal. A second coefficient quantizing unit quantizes the second coefficients and provides a quantized coefficient signal. An excitation quantizing unit derives an excitation signal concerning the speech signal by using the speech signal, the first coefficient signal, the second coefficient signal and the quantized coefficient signal, quantizes the second source signal thus derived and provides a quantized excitation signal. A reproduced signal generating unit reproduces a speech of the pertinent frame by using the first coefficient signal, the quantized coefficient signal and the quantized excitation signal.

ABSTRACT OF THE DISCLOSURE

A first coefficient generating unit derives, from past reproduced speech signal, a first coefficient signal representing a spectral characteristic of the past reproduced speech signal. A residual signal generating unit derives, from speech signal for each frame, a predicted residue signal by using the first coefficients. A second coefficient generator derives second coefficients representing a spectral characteristic of the predicted residue signal. A second coefficient quantizing unit quantizes the second coefficients and provides a quantized coefficient signal. An excitation quantizing unit derives an excitation signal concerning the speech signal by using the speech signal, the first coefficient signal, the second coefficient signal and the quantized coefficient signal, quantizes the second source signal thus derived and provides a quantized excitation signal. A reproduced signal generating unit reproduces a speech of the pertinent frame by using the first coefficient signal, the quantized coefficient signal and the quantized excitation signal.

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HIGH QUALITY SPEECH CODER AND CODING METHOD

BACKGROUND OF THE INVENTION

The present invention relates to a speech coder for high quality coding an input speech signal at
5 low bit rates.

As a well-known systems for high quality coding input speech signal, CELP (Code Excited Linear Predictive Coding) is disclosed in M. Schroeder and B. Atal, "Code-excited linear prediction: High
10 quality speech at very low bit rates", Proc. ICASSP, pp. 937-940, 1985 (hereinafter referred to as Literature 1), Kleij et al, "Improved speech quality and efficient vector quantization in SELP", Proc. ICASSP, pp. 155-158, 1988 (hereinafter referred to
15 as Literature 2), and so forth.

On the transmitting side of such a coding system, spectral parameters representing spectral characteristics of speech signal are extracted from the same by linear predictive (LPC) analysis of a
20 predetermined degree (for instance 10-th degree), and quantized to provide quantized parameters. In addition, each frame of the speech signal is divided into a plurality of sub-frames (for instance of 5 ms), codebook parameters (a delay parameter and a
25 gain parameter corresponding to the pitch cycle) are extracted for each sub-frame on the basis of past excitation signal by using the spectral parameters, and sub-frame speech signal is predicted by pitch

prediction with reference to the adaptive codebook.

The excitation signal thus obtained through the pitch prediction, is then quantized by selecting an optimum excitation codevector from an excitation codebook (or vector quantization codebook) which is constituted by predetermined kinds of noise signals and by calculating an optimum gain. The excitation codevector selection is performed such as to minimize error power between a signal synthesized from the selected noise signals and a residue signal. Index indicative of kind of the selected codevector, a gain, quantized spectral parameters and extracted adaptive codebook parameters, are multiplexed in a multiplexer, and the resultant multiplexed data is transmitted. The receiving side is not described.

As a method of improving the analysis accuracy of the speech signal spectral parameter on the basis of the CELP, has been proposed one, on the transmitting side of which spectral parameters of reproduced speech signal are developed by analyzing past reproduced speech signal in a higher degree than the conventional degree, and used to quantize the speech. As this method, LD-CELP (Low-Delay CELP) is well known and described in, for instance, J-H Chen et al, "A low-delay CELP coder for the CCITT 16 kb/s speech coding standard", IEEE Journal of Selected Areas on Communications, vol. 10, pp.

830-849, June 1992 (hereinafter referred to as Literature 3). In the LD-CELP, on the receiving side as well as the transmitting side, spectral parameters are developed from the past reproduced speech signal by analysis thereof and used. This
5 provided for a merit that no spectral parameter need be transmitted even when the degree of analysis is greatly increased.

Such well-known speech coding/decoding method
10 is disclosed in, for example, in Patent-Laid Open 4-344699.

In the speech coding method disclosed in Literatures 1 and 2, since the spectral parameters are analyzed with a constant degree (for example,
15 10-degree) for each frame, if the analysis degree is increased to twice (for example, 20-degree) in order to increase the spectral analysis degree, it requires twice number of transmission bits, increasing bit rate.

20 In the speech coding method disclosed in Literature 3, it requires to transmit the speech parameters of the analysis degree is increased. The spectral parameter matching is degraded at portions where the signal characteristic is changed with
25 time, degrading the characteristic and speech quality. This is due to the use of spectral parameters analyzed from the past produced signal. In particular, the increase of analysis degree

degrades the matching characteristic of the reproduced signal developed on the transmission side and the reproduced signal on the received side, when error is caused on the transmission side, remarkably degrading the speech quality on the receiving side because of mismatching between the reproduced signal obtained from the reproduced signals on the transmission side and the receiving side.

10 SUMMARY OF THE INVENTION

An object of the present invention is therefore to provide a speech coder and coding method capable of improving speech quality with relatively small amount of calculations. An aspect of the invention which seeks to satisfy the

15 aforesaid capability of speech quality includes a speech coder comprising a divider operable to divide an input speech signal into a plurality of frames having a predetermined time length; a first coefficient analyzing unit operable to derive first coefficients representing a spectral characteristic of a past

20 speech reproduction signal and provide the first coefficients as a first coefficient signal; a residue generating unit operable to derive a predicted residue signal from the input speech signal by using the first coefficient signal; a second coefficient analyzing unit operable to derive second

25 coefficients representing a spectral characteristic of the predicted residue signal and provide the second coefficients as a second coefficient signal; a coefficient quantizing unit operable to quantize the second coefficients represented by the second coefficient signal and provide the quantized second

30 coefficients as a quantized coefficient signal; an excitation signal generating unit operable to derive an excitation signal in accordance with the input speech signal in a particular frame, the first coefficient signal, the second coefficient signal and the quantized coefficient signal, the excitation

35 signal generating unit including a quantizer operable to quantize the excitation signal and provide the quantized

signal as a quantized excitation signal; and a speech reproducing unit operable to reproduce speech of the particular frame by using the first coefficient signal, the quantized coefficient signal and the quantized excitation signal to produce a speech reproduction signal; the past speech reproduction signal being derived from the speech reproduction signal.

According to an aspect of the present invention, there is provided a speech coder comprising a divider for dividing an input speech signal into a plurality of frames having a predetermined time length, a first coefficient analyzing unit for deriving first coefficients representing a spectral characteristic of past reproduced signal from the reproduced speech signal and providing the first coefficient as a first coefficient signal, a residue generating unit for deriving a predicted residue from the speech signal by using the first coefficient signal, a second coefficient analyzing unit for deriving second coefficients representing a spectral characteristic of the predicted residue signal from the predicted residue signal and providing the second coefficients from the second coefficient signal, a coefficient quantizing unit for quantizing the second coefficients represented by the second coefficient signal and providing the quantized coefficient as a quantized coefficient signal, an excitation signal generating unit for deriving an excitation signal concerning the speech signal in the pertinent frame by using the speech signal, the first coefficient signal, the second coefficient signal and the quantized coefficient signal, quantizing the excitation signal, and providing the quantized signal as a quantized excitation signal, and a speech reproducing unit for reproducing a speech of the pertinent frame by

using the first coefficient signal, the quantized coefficient signal and the quantized excitation signal and providing a speech reproduction signal.

According to another aspect of the present invention, there is provided a speech coder comprising a divider operable to divide an input speech signal into a plurality of frames having a predetermined time length; a first coefficient analyzing unit operable to derive first coefficients representing a spectral characteristic of a past speech reproduction signal and provide the first coefficients as a first coefficient signal; a residue generating unit operable to derive a predicted residue from the input speech signal by using the first coefficients and provide a predicted gain signal representing a predicted gain calculated from the predicted residue; a judging unit operable to determine whether the predicted gain represented by the predicted gain signal is above a predetermined threshold and provide a judge signal representing the result of the determination; a second coefficient analyzing unit operative, when the judge signal represents a predetermined value, to derive second coefficients representing a spectral characteristic of the predicted gain from the predicted gain signal and provide the second coefficients as a second coefficient signal; a coefficient quantizing unit operable to quantize the second coefficients represented by the second coefficient signal and provide the quantized second coefficients as a quantized coefficient signal; an excitation signal generating unit operable to produce a quantized excitation signal in accordance with the input speech signal by quantizing the speech signal, the second coefficient signal and the quantized coefficient signal, the excitation signal generating unit using the

second coefficients to produce the quantized excitation signal depending on the value of the judge signal; and a speech reproducing unit operable to produce a speech reproduction signal of a pertinent frame by using the second coefficients, the quantized coefficient signal and the quantized excitation signal, the speech reproducing unit using the first coefficients to produce the speech reproduction signal depending on the value of the judge signal; the past speech reproduction signal being derived from the speech reproduction signal.

According to another aspect of the present invention, there is provided a speech coder comprising a divider for dividing an input speech signal into a plurality of frames having a predetermined time length; a mode judging unit for selecting one of a plurality of different modes by extracting a feature quantity from the input speech signal and providing a mode signal representing the selected mode; a first coefficient analyzing unit operative, when a predetermined one of the modes exists as represented by the mode signal, to derive first coefficients representing a spectral characteristic of a past speech reproduction signal and providing the first coefficients as a first coefficient signal; a residue generating unit for deriving a predicted residue signal for each frame of the input speech signal by using the first coefficient signal; a second coefficient analyzing unit for deriving second coefficients representing a spectral characteristic of the predicted residue signal and providing the second coefficients as a second coefficient signal; a coefficient quantizing unit or quantizing the second coefficients represented by the second coefficient signal and providing the quantized second coefficients as a quantized coefficient signal; an excitation signal

generating unit for deriving an excitation signal in accordance with the input speech signal, the first coefficient signal and the quantized coefficient signal; and a speech reproducing unit for producing a speech reproduction signal by using the first coefficient signal, the quantized coefficient signal and the quantized excitation signal; the past speech reproduction signal being derived from the speech reproduction signal.

10 The invention taught herein also encompasses a speech coding method comprising the steps of dividing an input speech signal into a plurality of frames having a predetermined time length; deriving first coefficients representing a spectral characteristic of a past speech reproduction signal and providing the first coefficients as a first coefficient signal; deriving a predicted residue signal from the input speech signal by using the first coefficient signal; deriving second coefficients representing a spectral characteristic of the predicted residue signal and providing the second coefficients as a second coefficient signal; quantizing the second coefficients represented by the second coefficient signal and providing the quantized coefficients as a quantized coefficient signal; deriving an excitation signal in accordance with the input speech signal in a particular frame, the first coefficient signal, the second coefficient signal and the quantized coefficient signal, quantizing the excitation signal, and providing the quantized signal as a quantized excitation signal; reproducing speech of the particular frame by using the first coefficient signal, the quantized coefficient signal and the quantized excitation signal to produce a speech reproduction signal; and the past speech

reproduction signal being derived from the speech reproduction signal.

According to another aspect of the present invention, there is provided a speech coding method comprising the steps of dividing an input speech signal into a plurality of frames having a predetermined time length; deriving first coefficients representing a spectral characteristic of a past speech reproduction signal and providing the first coefficients as a first coefficient signal; deriving a predicted residue from the input speech signal by using the first coefficients and providing a predicted gain signal representing a predicted gain calculated from the predicted residue; determining whether the predicted gain represented by the predicted gain signal is above a predetermined threshold and providing a judge signal representing the result of the determination; deriving second coefficients representing a spectral characteristic of the predicted gain from the predicted gain signal and providing the second coefficients as a second coefficient signal, the deriving and providing steps operative when the judge signal represents a predetermined value; quantizing the second coefficients represented by the second coefficient signal and providing the quantized second coefficients as a quantized coefficient signal; producing a quantized excitation signal according to the input speech signal by quantizing the speech signal, the second coefficient signal and the quantized coefficient signal, wherein the second coefficients are used to produce the quantized excitation signal depending on the value of the judge signal; and making a speech reproduction signal of the particular frame by using the second coefficients, the quantized coefficient signal and the quantized

excitation signal, wherein the first coefficients are used to produce the speech reproduction signal depending on the value of the judge signal.

According to another aspect of the present invention, there is provided a speech coding method comprising the steps of dividing an input speech signal into a plurality of frames having a predetermined time length; selecting one of a plurality of different modes by extracting a feature quantity from the input speech signal and providing a mode signal representing the selected mode; deriving first coefficients representing a spectral characteristic of a past speech reproduction signal and providing the first coefficients as a first coefficient signal; deriving a predicted residue signal for each frame of the input speech signal by using the first coefficient signal when a predetermined mode is represented by the mode signal; deriving second coefficients representing a spectral characteristic of the predicted residue signal and providing the second coefficients as a second coefficient signal; quantizing the second coefficients represented by the second coefficient signal and providing the quantized second coefficients as a quantized coefficient signal; deriving an excitation signal in accordance with the input speech signal, the first coefficient signal and the quantized coefficient signal; producing a speech reproduction signal by using the first coefficient signal, the quantized coefficient signal and the quantized excitation signal; the past speech reproduction signal being derived from the speech reproduction signal.

According to another aspect of the present invention, there is provided a coder for producing an output speech signal from an input speech signal, comprising a frame divider adapted to divide the input

speech signal into time frames of a predetermined length; a first signal generator having a linear prediction analyzer to produce first linear prediction coefficients (FLPCs) from a predetermined number of samples of an output speech feedback signal, the FLPCs being of a predetermined degree; a residue signal generator adapted to produce a predictive residue signal as a function of inverse filtering a predetermined number of samples of the input speech signal and the FLPCs; a second signal generator having a linear prediction analyzer to produce second linear prediction coefficients (SLPCs) from a predetermined number of samples of the predictive residue signal, the SLPCs being of a predetermined degree, the second signal generator having a linear spectrum pair (LSP) analyzer to produce LSP parameters from the SLPCs; a quantizer adapted to produce a quantized signal obtained by quantizing the LSP parameters; an excitation unit having an excitation quantizer, the excitation unit being adapted to produce a quantized excitation signal based on the input speech signal, the FLPCs, the SLPCs, and the quantized signal; and a speech reproducing unit adapted to produce a speech reproduction signal for each frame and the output speech feedback signal using the FLPCs, the quantized signal and the quantized excitation signal.

30

Other objects and features will be clarified from the following description with reference to attached drawings.

5 BRIEF DESCRIPTION OF THE DRAWINGS

Fig. 1 is a block diagram showing the basic construction of the speech coder in a first embodiment of the present invention;

10 Fig. 2 is a detailed construction of the excitation quantizer 350 in Fig. 1;

Fig. 3 is a block diagram showing the basic construction of a speech coder in a second embodiment of the present invention;

15 Fig. 4 is a block diagram showing the basic construction of a speech coder in a third embodiment to the present invention; and

Figs. 5 to 7 show modifications of the embodiments of the speech coder shown in Figs. 1, 3 and 4, respectively.

20 PREFERRED EMBODIMENTS OF THE INVENTION

Preferred embodiments of the present invention will now be described will now be described with reference to the drawings.

25 Fig. 1 is a block diagram showing the basic construction of the speech coder in a first embodiment of the present invention.

In this embodiment, speech signal $x(n)$ is provided from an input terminal 100 to a frame

divider 110. The fame divider 110 divides the speech signal $x(n)$ into frames (of 10 ms, for instance). A sub-frame divider 120 divides each frame speech signal into sub-frames (of 5 ms, for instance) each shorter than the frames.

A first coefficient signal generator (or first coefficient analyzer) 380 calculates first coefficients, which are given as linear prediction coefficients a_{1i} ($i = 1, \dots, P1$) of predetermined degree $P1$ (for instance $P1 = 20$) degree through linear prediction analysis using a predetermined number of samples of past frame reproduced speech signal $s(n - L)$, and provides the calculated first coefficient as a first coefficient signal. The linear prediction analysis may be performed by a well-known process, such as LPC analysis or Burg analysis. Here, it is assumed that the Burg analysis is used. The Burg analysis is detailed in, for instance, Nakamizo, "Signal analysis and system identification", issued by Corona Co., Ltd., 1988, pp. 82-87 (hereinafter referred to as Literature 4), and hence not described.

A residue signal generator (or residue calculator) 390 calculates predictive residue signal $e(n)$ given by the following equation (1) as a result of calculation of inverse filtering of a predetermined number of samples of the speech signal $x(n)$.

$$e(n) = x(n) - \sum_{i=1}^{P1} \alpha_{1i} x(n-i) \quad (1)$$

A second coefficient generator (or second
 5 coefficient analyzer) 200 calculates second
 coefficient α_{2j} ($j = 1, \dots, P2$) of $P2$ -th degree, by
 linear predictive analysis of a predetermined number
 of samples of the predictive residue signal $e(n)$.
 The second coefficient generator 200 converts the
 10 second coefficient α_{2j} into LSP parameters which are
 suited for quantization and interpolation, and
 provides these LSP parameters as a second
 coefficient signal. The conversion of the linear
 predictive coefficients into LSP, may be performed
 15 by adopting techniques disclosed in Sugamura et al,
 "Speech data compression on the basis of linear
 spectrum pair (LSP) speech analysis synthesis
 system". The Transactions of Institute of
 Electronics and Communication Engineers of Japan,
 20 J64-A, pp. 599-606, 1981 (hereinafter referred to as
 Literature 5).

A second coefficient quantizer (or coefficient
 quantizer) 210 efficiently quantizes the LSP
 parameters, represented by the second coefficient
 25 signal, using a codebook 220, selects codevector D_j
 which minimizes a distortion given by the following
 equation (2), and provides an index of the selected
 codevector D_j as a quantized coefficient signal

representing the quantized coefficients to a multiplexer 400.

$$D_j = \sum_{i=1}^{P2} W(i) [LSP(i) - QLSP(i)_j]^2 \quad (2)$$

5

where $LSP(i)$, $QLSP(i)_j$ and $W(i)$ are i -th LSP, j -th codevector stored in the codebook 220 and weighting coefficient, respectively, before the quantization.

In the following description, it is assumed
 10 that the vector quantization is employed, and that the LSP parameter representing the second coefficients are quantized. The LSP parameters may be quantized by vector quantization in a well-known method. Specific methods that can be utilized are
 15 disclosed in Japanese Laid-Open Patent Publication No. 4-171500 (Japanese Patent Application No. 2-297600, hereinafter referred to as Literature 6), Japanese Laid-Open Patent Publication No. 4-363000 (Japanese Patent Application No. 3-261925,
 20 hereinafter referred to as Literature 8), and T. Nomura et al, "LSP Coding Using VQ-SVQ with Interpolation in 4,075 kbps M-LCELP Speech Coder", Proc. Mobile Multimedia Communications, pp. B. 2.5, 1993 (hereinafter referred to as Literature 9), and
 25 are not described.

The second coefficient quantizer 210 provides a quantized coefficient signal, representing linear prediction coefficient α'_{2j} ($j = 1, \dots, P2$) obtained

from the quantized LSP parameter, to an impulse response generator 310.

An acoustical weighting circuit 230 calculates linear prediction coefficient β_i of predetermined degree P through Brug analysis from the speech signal $x(n)$ from the frame divider 110. Using this linear prediction coefficients, a filter having a transfer characteristic $H(z)$ given by the following equation (3) is formed. The acoustical weighting of the speech signal $x(n)$ from the sub-frame divider 120 is performed to provide resultant weighted speech signal $x_w(n)$.

$$H(z) = \frac{1 - \sum_{i=1}^P \beta_i \gamma_1^i z^{-i}}{1 - \sum_{i=1}^P \beta_i \gamma_2^i z^{-i}} \quad (3)$$

where γ_1 and γ_2 are acoustical weighting factor control constants selected to adequate values such that $0 < \gamma_2 < \gamma_1 \leq 1.0$. The linear prediction coefficient β_i is provided to an impulse response generator 310.

The impulse response generator 310 calculates impulse response $h_w(z)$ of an acoustic weighting filter, the z transform of which is given by the following equation (4) for predetermined number L of instants, and provides the calculated impulse response to an adaptive codebook circuit 300, an excitation quantizer 350 and a gain quantizer 365.

$$H_w(z) = \frac{1 - \sum_{i=1}^P \beta_i \gamma_1^i z^{-i}}{1 - \sum_{i=1}^P \beta_i \gamma_2^i z^{-i}} \frac{1}{1 - \sum_{i=1}^{P1} \alpha_{1i} z^{-i}} \frac{1}{1 - \sum_{i=1}^{P2} \alpha'_{2i} z^{-i}} \quad (4)$$

5 A response signal generator 240 calculates
 response signal $x_z(n)$ of one sub-frame for the input
 signal of $d(n) = 0$, from coefficients provided from
 the first and second coefficient generators 380 and
 200 and second coefficient quantizer 210 and using
 10 stored filter memory values, and provides the
 calculated response signal $x_z(n)$ to a subtracter 235.
 The response signal $x_z(n)$ is given by equation (5).

$$x_z(n) = d(n) - \sum_{i=1}^P \beta_i \gamma_1^i d(n-i) + \sum_{i=1}^P \beta_i \gamma_2^i y_1(n-i) + \sum_{i=1}^{P1} \alpha_{1i} y_2(n-i) + \sum_{i=1}^{P2} \alpha'_{2i} x_z(n-i) \quad (5)$$

15

The subtracter 235 subtracts the response
 signal $x_z(n)$ from the weighted speech signal $x_w(n)$
 for one frame, and provides the result $x'_w(n)$, given
 as $x'_w(n) = x_w(n) - x_z(n)$, to the adaptive codebook
 20 circuit 300.

The adaptive codebook circuit 300 is provided
 with past excitation signal $v(n)$ from a weighting
 signal generator 360 to be described later, the
 output signal $x'_w(n)$ from the subtracter 235, and
 25 acoustic-weighted impulse signal $h_w(n)$ from the
 impulse response generator 310, calculates delay T
 corresponding to the pitch cycle according to a
 codevector which minimizes the distortion D_T given by

the following equation (6), and outputs an index representing the delay T to the multiplexer 400.

$$D_T = \sum_{n=0}^{N-1} x_w'^2(n) - \left[\sum_{n=0}^{N-1} x_w'(n) y_w(n-T) \right]^2 / \sum_{n=0}^{N-1} y_w^2(n-T) \quad (6)$$

5

where $y_w(n-T) = v(n-T) * h_w(n)$ represents a pitch prediction signal, and symbol $*$ represents convolution operation.

Gain η is calculated in accordance with equation (7).

$$\eta = \sum_{n=0}^{N-1} x_w'(n) y_w(n-T) / \sum_{n=0}^{N-1} y_w^2(n-T) \quad (7)$$

To improve the extraction accuracy of the delay T for woman's voice and child's voice, the delay T may be derived not from an integral number of samples but from a decimal number of samples. A specific method to this end may be adopted by having reference to, for instance, P. Kroon et al., "Pitch predictors with high temporal resolution", Proc. ICASSP, pp. 661-664, 1990 (hereinafter referred to as Literature 10).

The adaptive codebook circuit 300 further provides pitch prediction residue signal $z_w(n)$ given as $z_w(n) = x_w'(n) - \eta v(n-T) * h_w(n)$, obtained by pitch prediction using selected delay T and gain η , and also a pitch prediction signal obtained by using selected delay T , to the excitation quantizer (or

excitation calculator) 350.

The excitation quantizer 350 assigns M non-zero amplitude pulses to each sub-frame, and sets a pulse position retrieval range of each pulse. For example, assuming the case of determining the positions of five pulses in a 5-ms sub-frame (i.e., 40 samples), the candidate pulse positions in the pulse position retrieval range of the first pulse are 0, 5, ..., 35, those of the second pulse are 1, 6, ..., 36, those of the third pulse are 2, 7, ..., 37, those of the fourth pulse are 3, 8, ..., 38, and those of the fifth pulse are 4, 9, ..., 39.

Fig. 2 shows the detailed construction of the excitation quantizer 350. A first correlation function generator 353 receives $z_w(n)$ and $h_w(n)$, and calculates first correlation function $\psi(n)$ given by the following equation (8). A second correlation function generator 354 receives $h_w(n)$, and calculates second correlation function $\phi(p, q)$ given by the following equation (9).

$$\phi(n) = \sum_{i=n}^{N-1} z_w(i) h_w(i-n), \quad n = 0, \dots, N-1 \quad (8)$$

$$\phi(p, q) = \sum_{n=\max(p, q)}^{N-1} h_w(n-p) h_w(n-q), \quad p, q = 0, \dots, N-1 \quad (9)$$

A pulse polarity setting circuit 355 extracts and provides polarity data of the first correlation function $\psi(n)$ for each candidate pulse position. A

pulse position retrieving circuit 356 calculates
 function D given as $D = C_k^2/E_k$ for each of the
 candidate pulse position combinations noted above,
 and selects a position which maximizes the function
 5 as an optimum position.

Denoting the number of pulses per sub-frame by
 M, C_k and E are expressed by the following equations
 (10) and (11), respectively.

$$10 \quad C_k = \sum_{k=1}^M \text{sign}(k)\phi(m_k) \quad (10)$$

$$E = \sum_{k=1}^M \text{sign}(k)^2 \phi(m_k, m_k) + 2 \sum_{k=1}^{M-1} \sum_{i=k+1}^M \text{sign}(k)\text{sign}(i)\phi(m_k, m_i) \quad (11)$$

where $\text{sign}(k)$ represents the polarity of k-th pulse
 15 and the polarity extracted in the pulse polarity
 setting circuit 355. In this way, the excitation
 quantizer 350 provides data of the polarities and
 positions of M pulses to the gain quantizer 365.
 The excitation quantizer 350 also provides a pulse
 20 position index, obtained by quantizing each pulse
 position with a predetermined number of bits, and
 also pulse polarity data to the multiplexer 400.

The gain quantizer 365 reads out gain
 codevectors from a gain codebook 367 and selects a
 25 gain codevector which maximizes the value of the
 following equation (12), and finally it selects a
 combination of amplitude codevector and gain
 codevector which minimizes the value of distortion

D_t .

$$D_t = \sum_{n=0}^{N-1} [x'_w(n) - \eta'_t v(n-T) * h_w(n) - G'_t \sum_{k=1}^M \text{sign}(k) h_w(n-m_k)]^2 \quad (12)$$

5 Here, two kinds of gains such as gain η' of the adaptive codebook and gain G' of excitation expressed by pulses are simultaneously vector-quantized. Where η'_t and G'_t constitute t-th element in two-dimensional gain codevectors stored in the gain codevector 367. The gain quantizer 365 selects
10 a gain codevector which minimizes the value of the distortion D_t by repetitively executing the above calculation for each gain codevector, and provides an index representing the selected gain codevector
15 to the multiplexer 400.

A reproduced speech signal generator (or speech reproducing unit) 370 provides a reproduced speech signal produced by speech reproduction, which is performed by storing speech signal $s(n)$ ($n = 0, \dots, N-1$, N being the number of samples in a frame) for
20 one frame. Filter transfer characteristic $H'(z)$ in this operation is as shown in equation (13).

$$H'(z) = \frac{1}{1 - \sum_{i=1}^{P1} \alpha_{1i} z^{-i}} \frac{1}{1 - \sum_{i=1}^{P2} \alpha'_{2i} z^{-i}} \quad (13)$$

A filter using the first coefficient α_{1i} and a filter using the quantized second coefficient α'_{2i}

both have recursive structures.

The weighting signal generator 360 noted above receives the individual indexes, reads out corresponding codevectors, and calculates drive
5 excitation signal $v(n)$ given by equation (14).

$$v(n) = \eta' v(n-T) + G' \sum_{k=1}^M \text{sign}(k) \delta(n-m_k) \quad (14)$$

The drive excitation signal $v(n)$ is provided to
10 the above adaptive codebook circuit 300. The weighting signal calculator 360 then generates a response signal $s_w(n)$ given by the following equation (15) for one sub-frame through the response calculation from output parameters from the first
15 coefficient generator 380, output parameters from the second coefficient generator 200 and output parameters from the second coefficient quantizer 210, and provides the response signal $s_w(n)$ thus generated to the response signal generator 240.

$$s_w(n) = v(n) - \sum_{i=1}^P \beta_i \gamma_1^i v(n-i) + \sum_{i=1}^P \beta_i \gamma_2^i P_1(n-i) + \sum_{i=1}^{P_1} \alpha_{1i} p_2(n-i) + \sum_{i=1}^{P_2} \alpha'_{2i} s_w(n-i) \quad (15)$$

In the first embodiment of the speech coder, the individual components operate as described
25 above. The reproduced speech signal generator 370, weighting signal generator 360 and response signal generator 240 all use recursive filters for filtering the first coefficient signal.

In this speech coder, the first coefficients representing a spectral characteristic of the past reproduced speech signal is first developed, the predicted residue signal is developed by prediction of the pertinent frame speech signal from the first coefficients, the second coefficients representing a spectral characteristic of the predicted residue signal is developed, then the second coefficients are quantized to develop the quantized coefficient signal, and the excitation signal is obtained from the first coefficient signal, quantized coefficient signal and speech signal. Thus, while the sole second coefficient signal is transmitted, the prediction is performed in the sum of the degrees of the first and second coefficients. It is thus possible to greatly improve the speech signal spectrum approximation accuracy. In addition, in the event of error generation on the transmission line, the sound quality is less deteriorated compared to the prior art because the second coefficients are less immune to errors. With this speech coder, it is thus possible to obtain, with the same bit rate as in the prior art, compressed decoded speech of higher quality with relatively less calculation effort.

Fig. 3 is a block diagram showing the basic construction of a speech coder in a second embodiment of the present invention.

Compared to the preceding first embodiment of the speech coder, this embodiment further comprises a predicted gain generator 410 and a judging circuit 390, and the functions of some parts in the first
 5 embodiment are changed, these parts being designated by different reference numerals.

In this speech coder, the predicted gain generator 410 calculates predicted gain G_p , given by the following equation (16), from the speech signal
 10 and the predicted residue signal from the residue signal generator 390, and provides a predicted gain signal representing the calculation result of the predicted gain G_p to the judging circuit 420.

$$15 \quad G_p = \frac{\sum_{n=0}^{N-1} x^2(n)}{\sum_{n=0}^{N-1} e^2(n)} \quad (16)$$

The residual signal generator 390 and predicted
 20 gain generator 410 constitute a residue generator, which derives the predicted residue from the speech signal by using the first coefficient signal and provides the predicted gain signal representing the calculation result of the predicted gain
 25 corresponding to the derived predicted residue.

The judging circuit 420 compares the predicted gain G_p with a predetermined threshold and judges whether the predicted gain G_p is greater than the

threshold, and provides a judge signal representing judge data, which is "1" when G_p is less and "0" when G_p is greater, to a second coefficient generator 510, an impulse response generator 530, a response
5 generator 540, a weighting signal generator 550, a reproduced speech signal generator 560, and the multiplexer 400.

The second coefficient generator 510 receives the judge signal, and when the judge data thereof is
10 "1", it calculates the second coefficient from the predicted residue signal, and provides the calculation result as a second coefficient signal. When the judge data is "0", the second coefficient generator 510 generates speech signal from the frame
15 divider 110, calculates the second signal therefrom, and provides the result as the second coefficient signal.

As for the impulse generator 530, response signal generator 540, weighting signal generator 550
20 and reproduced speech signal generator 560, a judge as to whether the first coefficients are to be used is performed according to the judge data. When the judge data is "1", the first coefficient signal from the first coefficient signal generator 380, the
25 second coefficient signal from the second coefficient signal generator 510, and the quantization coefficient signal from the second coefficient quantizing circuit 210 are used. When

the judge data is "0", the first coefficient signal from the first coefficient generator 380 is not used.

The other parts than those described above have the same functions as in the first embodiment. In the above second embodiment of the speech coder, the individual parts have the functions as described above. The above reproduced signal generator 560, weighting signal generator 550 and response signal generator 540 each use a recursive filter for filtering the first coefficient signal.

In this speech coder, the predicted gain based on the first coefficient is calculated, and the first coefficients are used in combination with the second coefficient when and only when the predicted gain is above the threshold. Thus, it is possible to prevent deterioration of the overall sound quality even in a section, in which the prediction based on the first coefficient is deteriorated. In addition, even when an error occurs on the transmission line, the occurrence frequency of reproduced speech difference between the transmitting and receiving sides is reduced, so that it is possible to obtain high quality speech as a whole compared to the quality obtainable in the prior art.

Fig. 4 is a block diagram showing the basic construction of a speech coder in a third embodiment

to the present invention.

Compared to speech coder in the previous first embodiment, this speech coder further comprises a mode judging circuit 500, and the functions of some parts are changed, those parts being designated by different reference numerals. Again parts like those in the first embodiment are designated by like reference numerals, and are not described.

In this speech coder, the mode judging circuit 500 receives the speech signal frame by frame from the frame divider 110, extracts a feature quantity from the received speech signal, and provides a mode selection signal containing mode judge data representing a selected one of a plurality of modes to a first coefficient generator 520, a second coefficient generator 510 and the multiplexer 400.

The mode judging circuit 500 uses a feature quantity of the present frame for the mode judge. The feature quantity may be the frame mean pitch predicted gain. The pitch predicted gain is calculated according to the following equation (17).

$$G = 10 \log_{10} [1/L \sum_{i=1}^L (P_i / E_i)] \quad (17)$$

where L is the number of sub-frames contained in the frame, and P_i and E_i are the speech power and the pitch predicted error power of i-th frame as given by the following equations (18) and (19).

$$P_i = \sum_{n=0}^{N-1} x_i^2(n) \quad (18)$$

$$E_i = P_i - \left[\sum_{n=0}^{N-1} x_i(n)x_i(n-T) \right]^2 / \left[\sum_{n=0}^{N-1} x_i^2(n-T) \right] \quad (19)$$

5

where $x_i(n)$ is the speech signal in the i -th sub-frame, and T the optimum delay corresponding to the maximum predicted gain. The mode judging circuit 500 classifies the modes into a plurality of different kinds (for instance R kinds) by comparing the frame mean pitch predicted gain with a plurality of predetermined thresholds. The number R of different mode kinds may be 4. The modes may correspond to a no-sound section, a transient section, a weak vowel steady-state section, a strong vowel steady-state section, etc.

The first coefficient generator 520 receives the mode selection signal, and when and only when the mode discrimination data thereof represents a predetermined mode, calculates the first coefficient from the past reproduced speech signal. Otherwise, the first coefficient generator 520 does not calculate the first coefficients.

The second coefficient calculator 510 receives the mode selection signal, and when and only when mode discrimination data thereof represents a predetermined mode, it calculates the second coefficient from the predicted error signal from the

predicted residue signal generator 390. Otherwise, the second coefficient calculator 510 calculates the second coefficient from the speech signal from the frame divider 110.

5 The other parts as those described above have the same functions as in the first embodiment. In the speech coder in the third embodiment, the individual parts have the same functions as described above.

10 In this speech coder, one of a plurality of modes is discriminated by extracting a feature quantity from the speech signal. In a predetermined mode (for instance, one in which the speech signal characteristics are less subject to changes with
15 time, such as a steady-state section of a vowel), the second coefficients are calculated from the predicted residue signal after deriving the first coefficients, and the first and second coefficients are used in combination. Thus, it is possible
20 without need of predicted gain judge to prevent deterioration of prediction based on the first coefficient, and improve the sound quality compared to the prior art. In addition, even when an error occurs in the transmission line, the occurrence
25 frequency of reproduced speech difference between the transmitting and receiving sides is reduced, so that it is possible to obtain high quality speech as a whole compared to the quality obtainable in the

prior art.

The above embodiments of the speech coder according to the present invention can be variously modified. Figs. 5 and 7 show modifications of the 5 embodiments of the speech coder shown in Figs. 1 and 4, respectively. In these modifications, non-recursive filters are used in view of the recursive filters used for filtering the first coefficient signal in the reproduced signal 10 generator 370, weighting signal generator 360, and a response signal generator 240. Fig. 6 is a modification of the embodiment shown in Fig. 3. In this modification, non-recursive filters are used in lieu of the recursive filters used for filtering the 15 first coefficient signal in the reproduced signal generator 560, weighting signal generator 350 and response signal generator 540. In either case, the reproduced speech signal generator 600, weighting signal generator 610 and response signal generator 20 620 are provided.

As an example, the transfer characteristic $Q(z)$ of the non-recursive filter in the reproduced signal generator 600 shown in Fig. 5 is given by the following equation (20).

25

$$Q(z) = \frac{1 - \sum_{i=1}^{P1} \alpha_{1i} z^{-i}}{1 - \sum_{i=1}^{P2} \alpha'_{2i} z^{-i}} \quad (20)$$

Here, the filter using the first coefficients α_{1i} is recursive-type. The weighting signal generator 610 and the response signal generator 620 likewise use the first coefficients α_{1i} , and thus use
5 non-recursive filters of the same construction.

With this speech coder, in which the signal reproduction section uses non-recursive filter using the first coefficient, it is possible to increase the robustness with respect errors on the
10 transmission line.

While in the excitation quantizer 350 in the above embodiments of the speech coder the pulse amplitude was expressed in terms of instantaneous polarities, it is also possible to collectively
15 store amplitudes of a plurality of pulses in an amplitude codebook and permit selection of an optimum amplitude codevector from this codebook. As a further alternative, it is possible to use, in place of the amplitude codebook, a polarity
20 codebook, in which pulse polarity combinations are prepared in a number corresponding to the number of the pulses.

As has been described in the foregoing, in the speech quantizer according to the present invention
25 first coefficients representing a spectral characteristic of past reproduced speech signal is derived, a predicted residue signal is obtained by predicting speech signal in the pertinent frame with

the derived first coefficients, second coefficients representing a spectral characteristic of the predicted residue signal is obtained, a quantized coefficient signal is obtained by quantizing the second coefficients, and an excitation signal is provided from the first coefficient signal, quantized coefficient signal and speech signal. Thus, it is possible to permit prediction in the sum of the degrees of the first and second coefficients, while sending out the sole second coefficient signal. Also, with an arrangement that the predicted gain is calculated from the first coefficient and that the second coefficients are used in combination with the first coefficients when and only when the predicted gain exceeds a predetermined predicted gain, changes in speech signal characteristics with time may be increased to prevent deterioration of the overall sound quality even in a section, in which the prediction based on the first coefficients is deteriorated. Thus, when an error occurs on the transmission line, the occurrence frequency of reproduced speech difference between the transmitting and receiving sides is reduced. Furthermore, with an arrangement that one of a plurality of modes is discriminated by extracting a feature quantity of speech signal and that the second coefficients are calculated from the predicted residue signal in a predetermined mode

after deriving the first coefficient, it is possible to use the first and second coefficients in combination. Thus, without need of the predicted gain judge it is possible to prevent deterioration
5 of the overall sound quality due the first coefficients, thereby reducing the occurrence frequency of reduced speech difference between the transmitting and receiving sides in the event of transmission line error generation. Moreover, by
10 replacing the reflexive filters in the speech reproducing section with non-recursive filters, the robustness with respect to transmission line errors can be improved, so that further sound quality improvement can be obtained with relatively less
15 computational effort.

Changes in construction will occur to those skilled in the art and various apparently different modifications and embodiments may be made without departing from the scope of the present invention.
20 The matter set forth in the foregoing description and accompanying drawings is offered by way of illustration only. It is therefore intended that the foregoing description be regarded as illustrative rather than limiting.

THE EMBODIMENTS OF THE INVENTION IN WHICH AN EXCLUSIVE PROPERTY OR PRIVILEGE IS CLAIMED ARE DEFINED AS FOLLOWS:

1. A speech coder comprising:

a divider operable to divide an input speech signal into a plurality of frames having a predetermined time length;

a first coefficient analyzing unit operable to derive first coefficients representing a spectral characteristic of a past speech reproduction signal and provide the first coefficients as a first coefficient signal;

a residue generating unit operable to derive a predicted residue signal from the input speech signal by using the first coefficient signal;

a second coefficient analyzing unit operable to derive second coefficients representing a spectral characteristic of the predicted residue signal and provide the second coefficients as a second coefficient signal;

a coefficient quantizing unit operable to quantize the second coefficients represented by the second coefficient signal and provide the quantized second coefficients as a quantized coefficient signal;

an excitation signal generating unit operable to derive an excitation signal in accordance with the input speech signal in a particular frame, the first coefficient signal, the second coefficient signal and the quantized coefficient signal, the excitation signal generating unit including a quantizer operable to quantize the excitation signal and provide the quantized signal as a quantized excitation signal; and

a speech reproducing unit operable to reproduce speech of the particular frame by using the first coefficient signal, the quantized coefficient signal and the quantized excitation signal to produce a speech reproduction signal; the past speech reproduction signal being derived from the speech reproduction signal.

2. The speech coder according to claim 1, wherein the speech reproducing unit uses a non-reflexive filter for filtering the first coefficient signal.

3. A speech coder comprising:

a divider operable to divide an input speech signal into a plurality of frames having a predetermined time length;

a first coefficient analyzing unit operable to derive first coefficients representing a spectral characteristic of a past speech reproduction signal and provide the first coefficients as a first coefficient signal;

a residue generating unit operable to derive a predicted residue from the input speech signal by using the first coefficients and provide a predicted gain signal representing a predicted gain calculated from the predicted residue;

a judging unit operable to determine whether the predicted gain represented by the predicted gain signal is above a predetermined threshold and provide a judge signal representing the result of the determination;

a second coefficient analyzing unit operative, when the judge signal represents a predetermined value, to derive second coefficients representing a spectral characteristic of the predicted gain from the predicted gain signal and provide the second coefficients as a second coefficient signal;

a coefficient quantizing unit operable to quantize the second coefficients represented by the second coefficient signal and provide the quantized second coefficients as a quantized coefficient signal;

an excitation signal generating unit operable to produce a quantized excitation signal in accordance with the input speech signal by quantizing the speech signal, the second coefficient signal and the quantized coefficient signal, the excitation signal generating unit using the second coefficients to produce the

quantized excitation signal depending on the value of the judge signal; and

a speech reproducing unit operable to produce a speech reproduction signal of a pertinent frame by using the second coefficients, the quantized coefficient signal and the quantized excitation signal, the speech reproducing unit using the first coefficients to produce the speech reproduction signal depending on the value of the judge signal;

the past speech reproduction signal being derived from the speech reproduction signal.

4. The speech coder according to claim 3, wherein the speech reproducing unit uses a non-reflexive filter for filtering the first coefficient signal.

5. A speech coder comprising:

a divider for dividing an input speech signal into a plurality of frames having a predetermined time length;

a mode judging unit for selecting one of a plurality of different modes by extracting a feature quantity from the input speech signal and providing a mode signal representing the selected mode;

a first coefficient analyzing unit operative, when a predetermined one of the modes exists as represented by the mode signal, to derive first coefficients representing a spectral characteristic of a past speech reproduction signal and providing the first coefficients as a first coefficient signal;

a residue generating unit for deriving a predicted residue signal for each frame of the input speech signal by using the first coefficient signal;

a second coefficient analyzing unit for deriving second coefficients representing a spectral characteristic of the predicted residue signal and providing the second coefficients as a second coefficient signal;

a coefficient quantizing unit or quantizing the second coefficients represented by the second coefficient signal and providing the quantized second coefficients as a quantized coefficient signal;

an excitation signal generating unit for deriving an excitation signal in accordance with the input speech signal, the first coefficient signal and the quantized coefficient signal; and

a speech reproducing unit for producing a speech reproduction signal by using the first coefficient signal, the quantized coefficient signal and the quantized excitation signal;

the past speech reproduction signal being derived from the speech reproduction signal.

6. The speech coder according to claim 5, wherein the speech reproducing unit uses a non-reflexive filter for filtering the first coefficient signal.

7. A speech coding method comprising the steps of:

dividing an input speech signal into a plurality of frames having a predetermined time length;

deriving first coefficients representing a spectral characteristic of a past speech reproduction signal and providing the first coefficients as a first coefficient signal;

deriving a predicted residue signal from the input speech signal by using the first coefficient signal;

deriving second coefficients representing a spectral characteristic of the predicted residue signal and providing the second coefficients as a second coefficient signal;

quantizing the second coefficients represented by the second coefficient signal and providing the quantized coefficients as a quantized coefficient signal;

deriving an excitation signal in accordance with the input speech signal in a particular frame, the first coefficient signal, the second coefficient signal and the quantized coefficient signal, quantizing the excitation signal, and providing the quantized signal as a quantized excitation signal;

reproducing speech of the particular frame by using the first coefficient signal, the quantized coefficient signal and the quantized excitation signal to produce a speech reproduction signal; and

the past speech reproduction signal being derived from the speech reproduction signal.

8. A speech coding method comprising the steps of:

dividing an input speech signal into a plurality of frames having a predetermined time length;

deriving first coefficients representing a spectral characteristic of a past speech reproduction signal and providing the first coefficients as a first coefficient signal;

deriving a predicted residue from the input speech signal by using the first coefficients and providing a predicted gain signal representing a predicted gain calculated from the predicted residue;

determining whether the predicted gain represented by the predicted gain signal is above a predetermined threshold and providing a judge signal representing the result of the determination;

deriving second coefficients representing a spectral characteristic of the predicted gain from the predicted gain signal and providing the second coefficients as a second coefficient signal, the deriving and providing steps operative when the judge signal represents a predetermined value;

quantizing the second coefficients represented by the second coefficient signal and providing the quantized second coefficients as a quantized coefficient signal;

producing a quantized excitation signal according to the input speech signal by quantizing the speech signal, the second coefficient signal and the quantized coefficient signal, wherein the second coefficients are used to produce the quantized excitation signal depending on the value of the judge signal; and

making a speech reproduction signal of the particular frame by using the second coefficients, the quantized coefficient signal and the quantized excitation signal, wherein the first coefficients are used to produce the speech reproduction signal depending on the value of the judge signal.

9. A speech coding method comprising the steps of:

dividing an input speech signal into a plurality of frames having a predetermined time length;

selecting one of a plurality of different modes by extracting a feature quantity from the input speech signal and providing a mode signal representing the selected mode;

deriving first coefficients representing a spectral characteristic of a past speech reproduction signal and providing the first coefficients as a first coefficient signal;

deriving a predicted residue signal for each frame of the input speech signal by using the first coefficient signal when a predetermined mode is represented by the mode signal;

deriving second coefficients representing a spectral characteristic of the predicted residue signal and providing the second coefficients as a second coefficient signal;

quantizing the second coefficients represented by the second coefficient signal and providing the quantized second coefficients as a quantized coefficient signal;

deriving an excitation signal in accordance with the input speech signal, the first coefficient signal and the quantized coefficient signal;

producing a speech reproduction signal by using the first coefficient signal, the quantized coefficient signal and the quantized excitation signal;

the past speech reproduction signal being derived from the speech reproduction signal.

10. A coder for producing an output speech signal from an input speech signal, comprising:

a frame divider adapted to divide the input speech signal into time frames of a predetermined length;

a first signal generator having a linear prediction analyzer to produce first linear prediction coefficients (FLPCs) from a predetermined number of samples of an output speech feedback signal, the FLPCs being of a predetermined degree;

a residue signal generator adapted to produce a predictive residue signal as a function of inverse filtering a predetermined number of samples of the input speech signal and the FLPCs;

a second signal generator having a linear prediction analyzer to produce second linear prediction coefficients (SLPCs) from a predetermined number of samples of the predictive residue signal, the SLPCs being of a predetermined degree, the second signal generator having a linear spectrum pair (LSP) analyzer to produce LSP parameters from the SLPCs;

a quantizer adapted to produce a quantized signal obtained by quantizing the LSP parameters;

an excitation unit having an excitation quantizer, the excitation unit being adapted to produce a quantized excitation signal based on the input speech signal, the FLPCs, the SLPCs, and the quantized signal; and

a speech reproducing unit adapted to produce a speech reproduction signal for each frame and the output speech feedback signal using the FLPCs, the quantized signal and the quantized excitation signal.

11. The coder of claim 10, wherein the linear prediction analyzer employs linear prediction coding analysis to produce the FLPCs.

12. The coder of claim 10, wherein the linear prediction analyzer employs Burg analysis to produce the FLPCs.

13. The coder of claim 10, wherein the predictive residue signal substantially adheres to the following equation:

$$e(n) = x(n) - \sum_{i=1}^{P1} \alpha_{1i} x(n-i)$$

where $e(n)$ is the predictive residue signal, $x(n)$ is the input speech signal, α_{1i} are the FLPCs, and $P1$ is the predetermined degree of the FLPCs.

14. The coder of claim 10, wherein the quantizer comprises a codebook unit having a plurality of sets of data including: an i -th input LSP value ($LSP(i)$), a j -th quantized LSP value ($QLSP(i)$), an i -th weighing value ($W(i)$), and an i -th indexed codevector ($D(j)$) representing the quantized signal, wherein the quantized signal substantially adheres to the following equation:

$$D_j = \sum_{i=1}^{P2} W(i) [LSP(i) - (QLSP(i))_j]^2$$

where $P2$ is the degree of the codevector $D(j)$.

15. The coder of claim 10, wherein the excitation unit comprises:

an acoustical weighing circuit having a linear prediction analyzer adapted to produce third linear prediction coefficients (TLPCs) from the input speech signal, the acoustical weighing circuit also having a filter adapted to receive the TLPCs to produce a weighted speech signal;

an impulse generator adapted to produce an impulse response from a z-transform circuit;

a response signal generator adapted to produce a response signal representing the input speech signal at zero value from the FLPCs, SLPCs, quantized signal and stored memory values;

a subtractor adapted to produce a subtraction signal by subtracting the response signal from the weighted speech signal;

an adaptive codebook unit adapted to determine a pitch prediction signal as a function of a delay T, the subtraction signal, the impulse response from the impulse generator, and a past sample of the excitation signal, the adaptive codebook unit also being adapted to produce a pitch prediction residue signal as a function of a gain value, the delay T, the subtraction signal, the impulse response from the impulse generator, and a past sample of the excitation signal; and

an excitation quantizer adapted to produce a quantized excitation signal from the impulse response from the impulse generator, pitch prediction signal, and the pitch prediction signal.

16. The coder of claim 15, wherein the filter of the acoustical weighing circuit has a transfer function which substantially adheres to the following equation:

$$H(z) = \frac{1 - \sum_{i=1}^P \beta_i \gamma_1^i z^{-i}}{1 - \sum_{i=1}^P \beta_i \gamma_2^i z^{-i}}$$

where β_i , are the TLPCs having a predetermined degree P , and γ_1 and γ_2 are acoustical weighing factor control constants selected such that $0 < \gamma_2 < \gamma_1 \leq 1.0$.

17. The coder of claim 15, wherein the z -transform of circuit which produces the calculated impulse response substantially adheres to the following equation:

$$H(z) = \frac{1 - \sum_{i=1}^P \beta_i \gamma_1^i z^{-i}}{1 - \sum_{i=1}^P \beta_i \gamma_2^i z^{-i}} \frac{1}{1 - \sum_{i=1}^{P1} \alpha_{1i} z^{-i}} \frac{1}{1 - \sum_{i=1}^{P2} \alpha'_{2i} z^{-i}}$$

where β_i are the TLPCs having a predetermined degree P , γ_1 and γ_2 are acoustical weighing factor control constants selected such that $0 < \gamma_2 < \gamma_1 \leq 1.0$, α_{1i} are the FLPCs having predetermined degree $P1$, and α'_{2i} represents the quantized signal having predetermined degree $P2$.

18. The coder of claim 15, wherein the response signal generator is adapted to produce a response signal which substantially adheres to the following equation:

$$x_2(n) = d(n) - \sum_{i=1}^P \beta_i \gamma_1^i d(n-i) + \sum_{i=1}^P \beta_i \gamma_2^i \gamma_1(n-i) + \sum_{i=1}^{P1} \alpha_{1i} \gamma_2(n-i) + \sum_{i=1}^{P2} \alpha'_{2i} x_2(n-i)$$

where $d(n)$ is the input speech signal at zero value, β_i are TLPCs having a predetermined degree P , γ_1 and γ_2 are acoustical weighing factor control constants selected such that $0 < \gamma_2 < \gamma_1 \leq 1.0$, α_{1i} are the FLPCs having predetermined degree $P1$, and α'_{2i} represents the quantized signal having predetermined degree $P2$.

19. The coder of claim 15, wherein the delay T is determined by minimizing a distortion equation which substantially adheres to the following equation:

$$D_T = \sum_{n=0}^{N-1} x_w'^2(n) - \left[\sum_{n=0}^{N-1} x_w'(n) y_w(n-T) \right]^2 / \sum_{n=0}^{N-1} y_w^2(n-T)$$

where $x_w'(n)$ is the subtraction signal and $y_w(n-T)$ is the pitch prediction signal, the pitch prediction signal being equal to $v(n-T) * h_w(n)$, where $v(n)$ is the past excitation signal and $h_w(n)$ is the impulse response.

20. The coder of claim 19, wherein the pitch prediction residue signal substantially adheres to the following equation: $z_w(n) = x_w'(n) - \eta v(n-T) * h_w(n)$, where $*$ denotes convolution and η substantially adheres to the following equation:

$$\eta = \sum_{n=0}^{N-1} x_w'(n) y_w(n-T) / \sum_{n=0}^{N-1} y_w^2(n-T)$$

FIG. 1

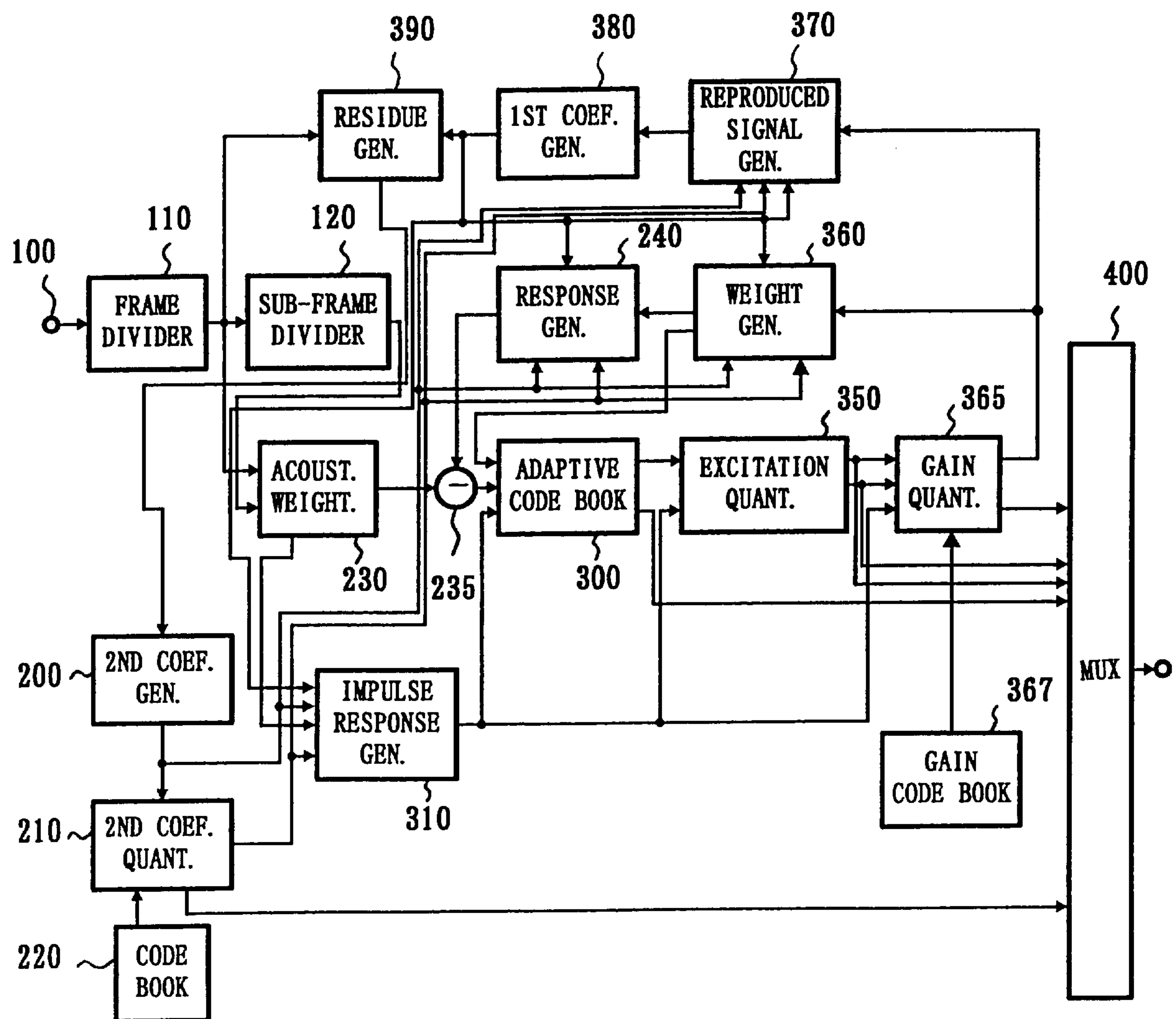


FIG. 2

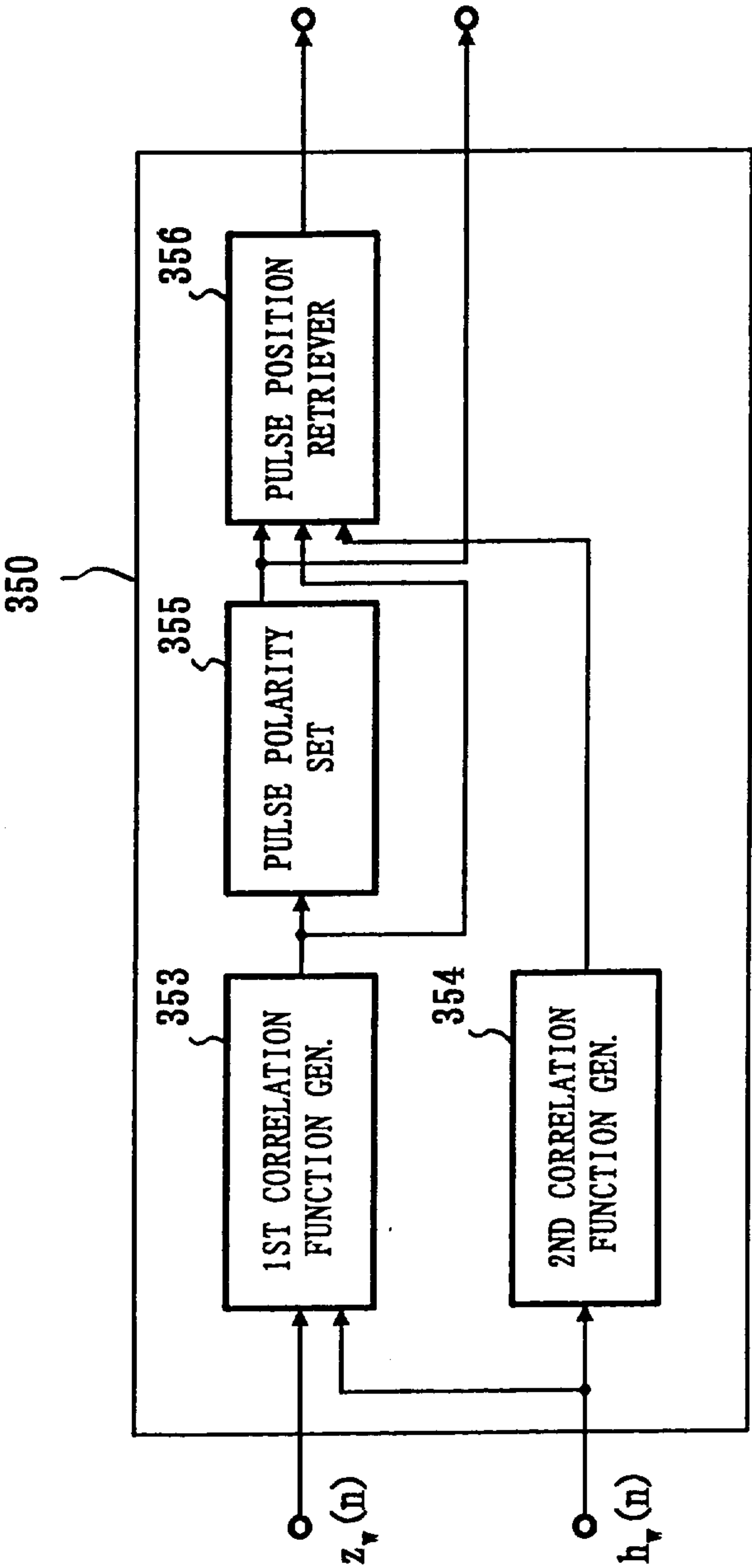


FIG. 3

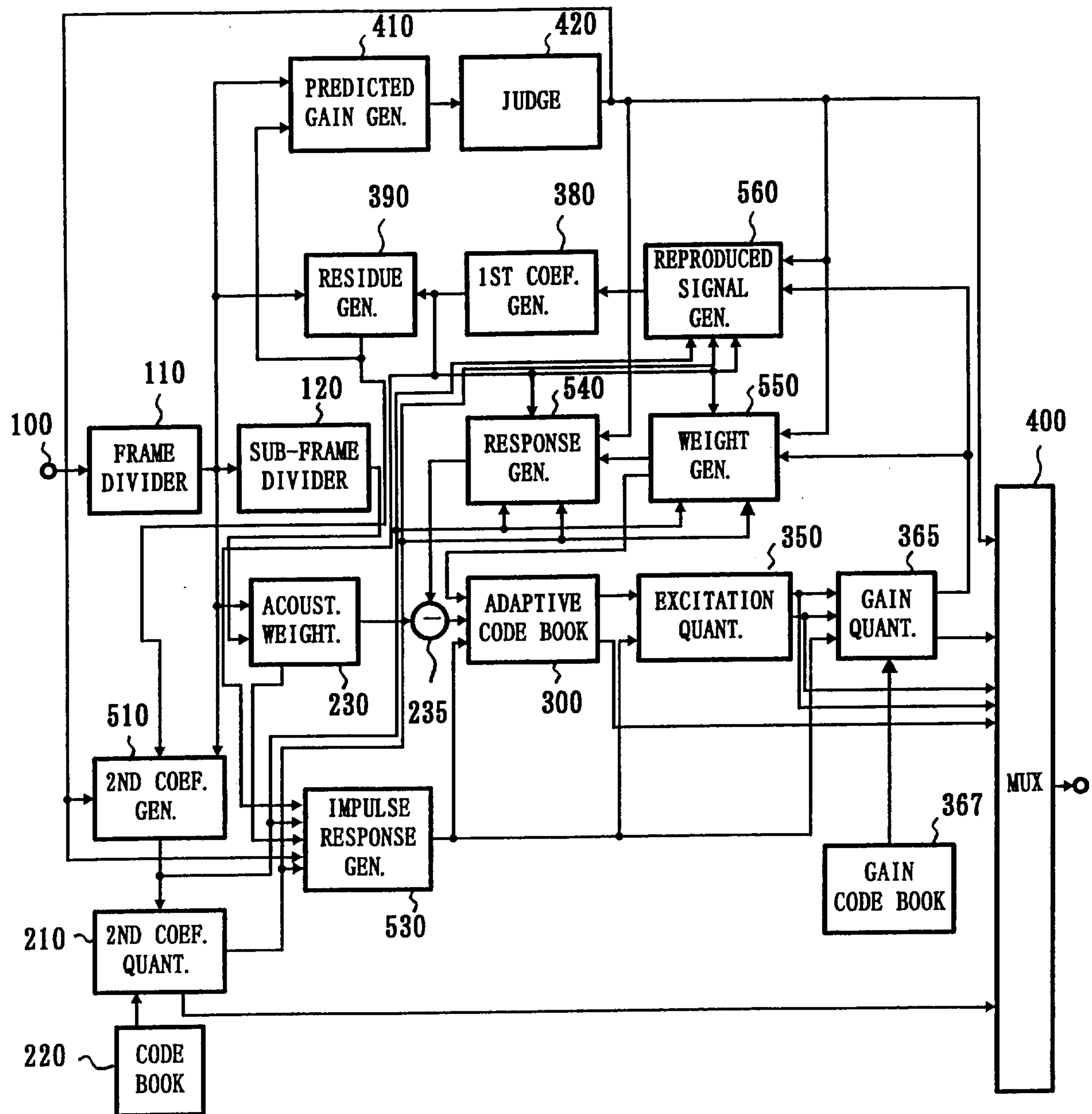


FIG. 4

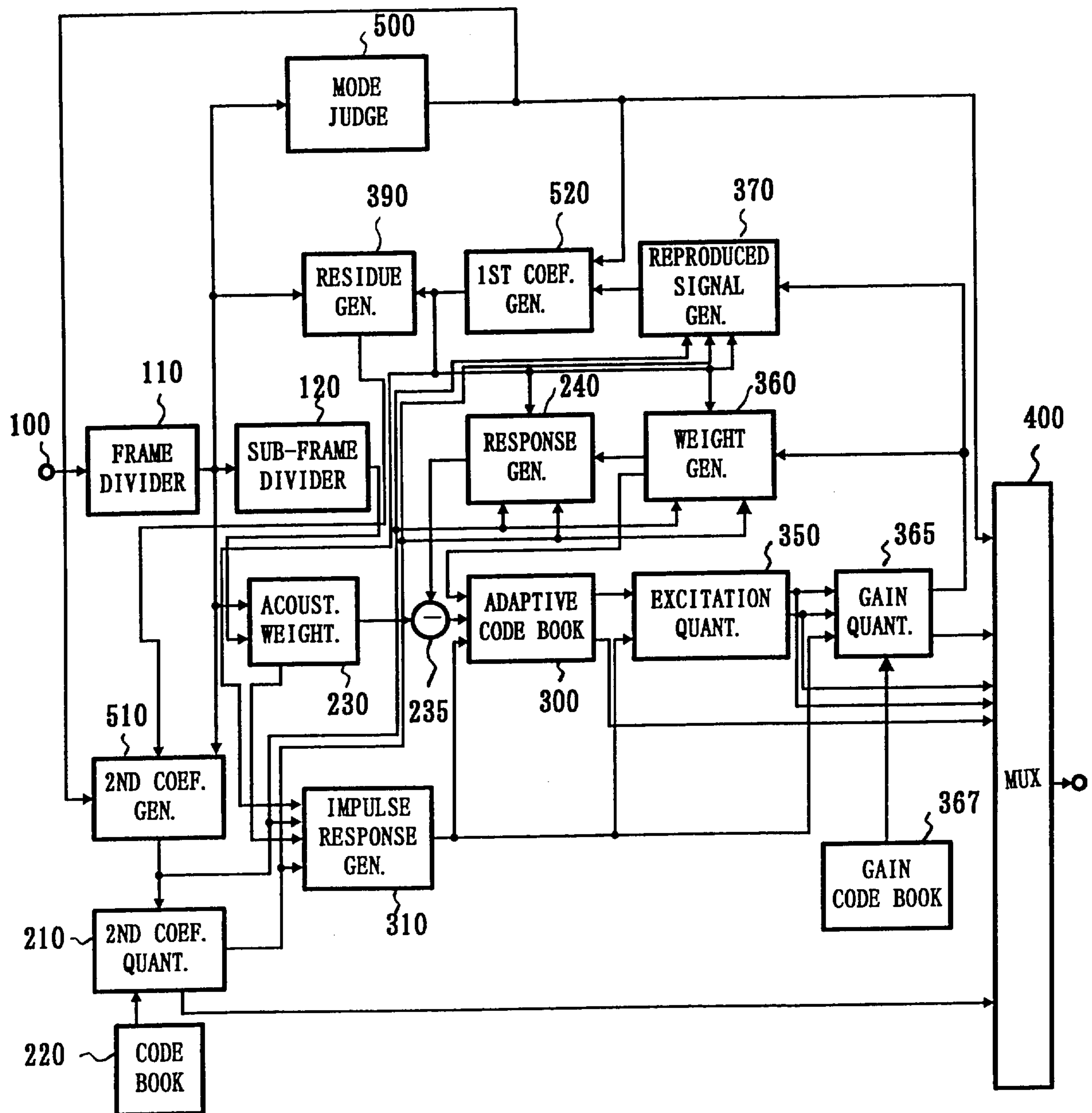


FIG.5

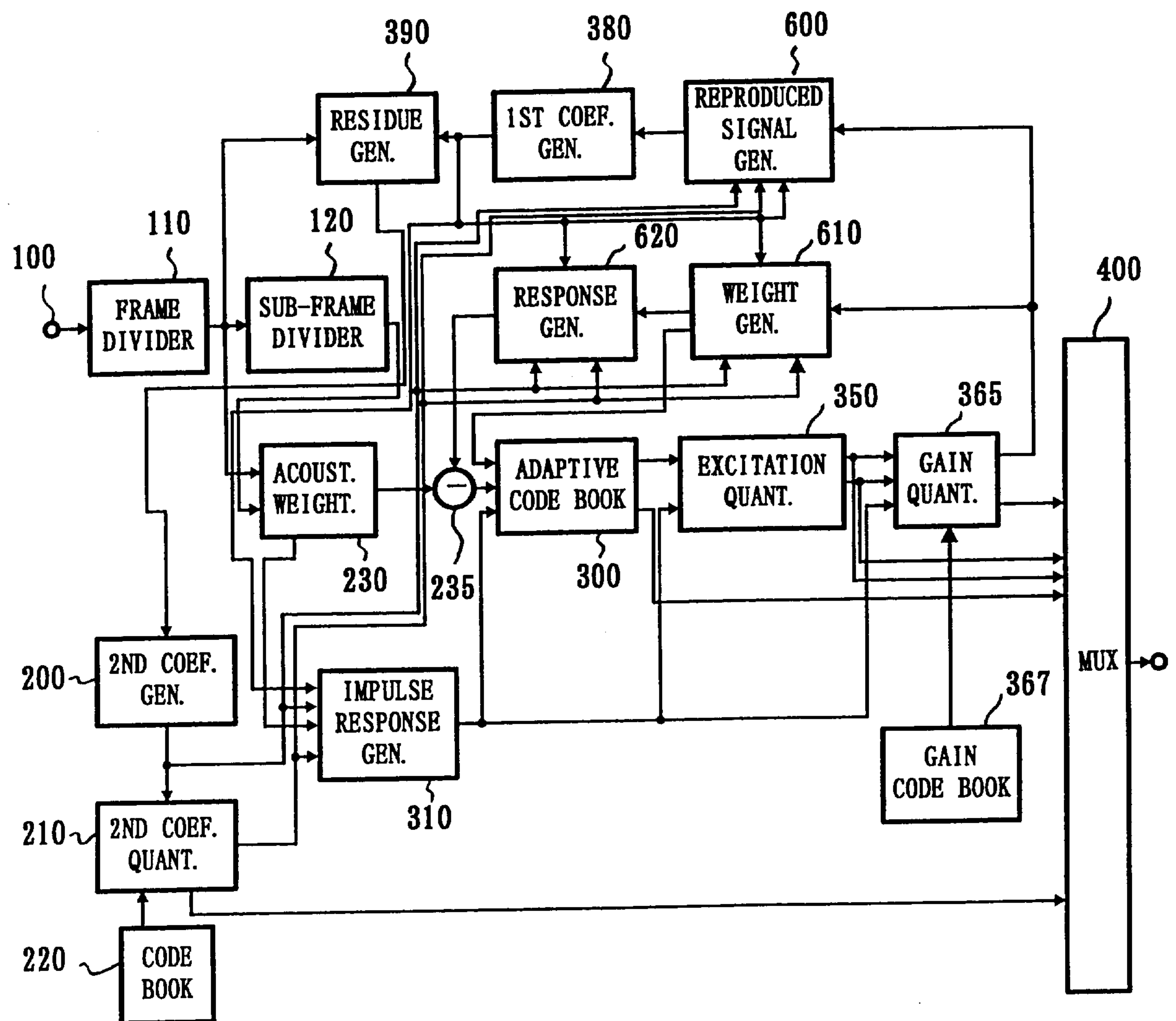


FIG. 6

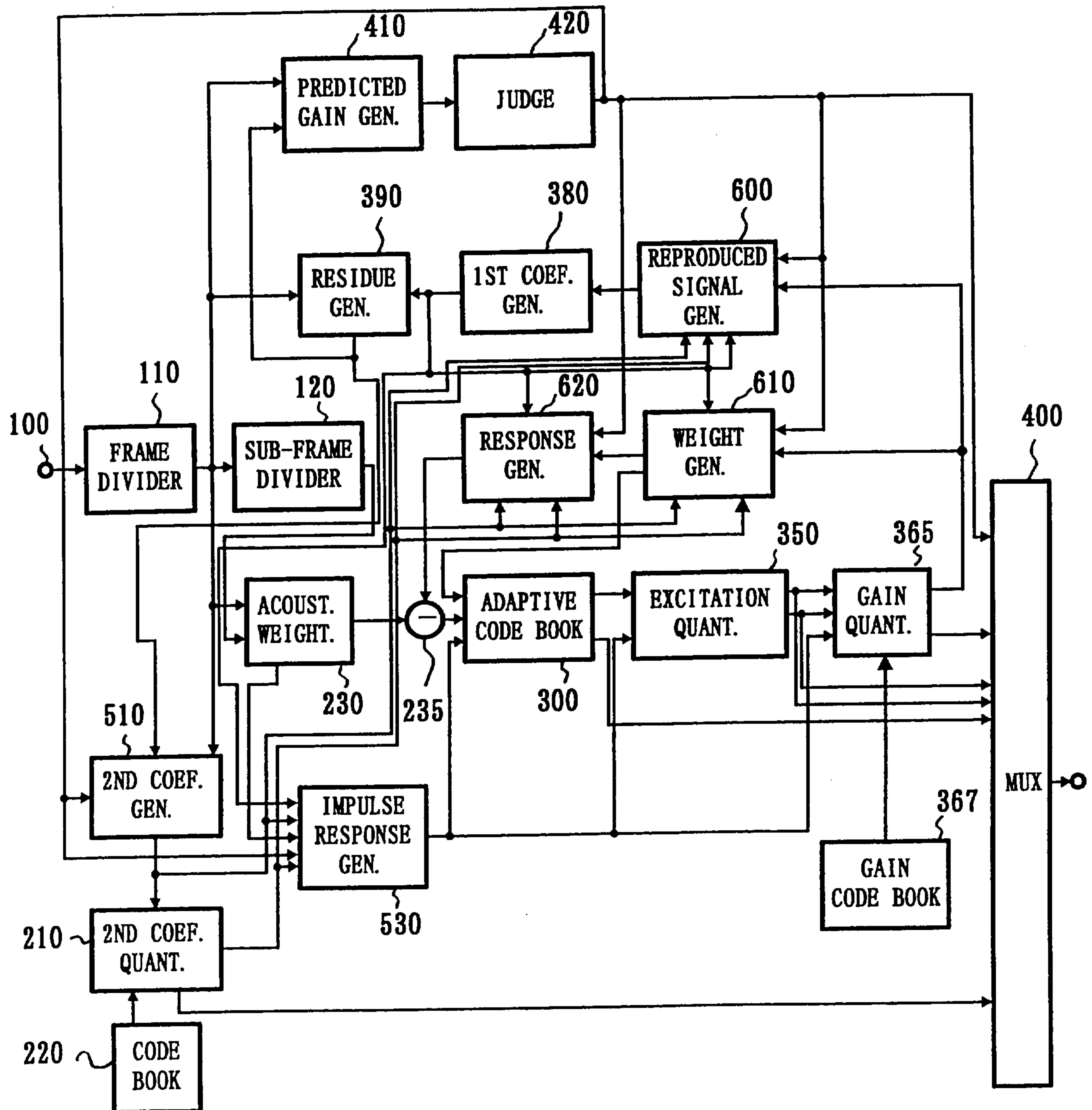


FIG. 7

