(19) Europäisches Patentamt
European Patent Office
Office européen des brevets

(11) EP 2 017 830 B9

(12) CORRECTED EUROPEAN PATENT SPECIFICATION

(15) Correction information:
Corrected version no 1 (W1 B1)
Corrections, see
Claims EN

(48) Corrigendum issued on:
23.02.2011 Bulletin 2011/08

(45) Date of publication and mention of the grant of the patent:
31.03.2010 Bulletin 2010/13

(21) Application number: **07743017.1**

(22) Date of filing: **09.05.2007**

(51) Int Cl.:
G10L 21/02 (2006.01) ***G10L 19/02*** (2006.01)
H03M 7/30 (2006.01)

(86) International application number:
PCT/JP2007/059582

(87) International publication number:
WO 2007/129728 (15.11.2007 Gazette 2007/46)

(54) ENCODING DEVICE AND ENCODING METHOD

CODIERUNGSEINRICHTUNG UND CODIERUNGSVERFAHREN

DISPOSITIF DE CODAGE ET PROCEDE DE CODAGE

(84) Designated Contracting States:
AT BE BG CH CY CZ DE DK EE ES FI FR GB GR HU IE IS IT LI LT LU LV MC MT NL PL PT RO SE SI SK TR

(30) Priority: **10.05.2006 JP 2006131852**
27.02.2007 JP 2007047931

(43) Date of publication of application:
21.01.2009 Bulletin 2009/04

(60) Divisional application:
10003491.7 / 2 200 026

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- **RAMPRASHAD S A: "A two stage hybrid embedded speech/audio coding structure" ACOUSTICS, SPEECH AND SIGNAL PROCESSING, 1998. PROCEEDINGS OF THE 1998 IEEE INTERNATIONAL CONFERENCE ON SEATTLE, WA, USA 12-15 MAY 1998, NEW YORK, NY, USA,IEEE, US, vol. 1, 12 May 1998 (1998-05-12), pages 337-340, XP010279163 ISBN: 978-0-7803-4428-0**
- **GRILL B: "A bit rate scalable perceptual coder for MPEG-4 audio" AUDIO ENGINEERING SOCIETY. CONVENTION PREPRINT, XX, XX, 26 September 1997 (1997-09-26), XP002302435**

Description

Technical Field

[0001] The present invention relates to a encoding apparatus and encoding method used in a communication system for encoding and transmitting signals.

Background Art

[0002] When speech/sound signals are transmitted in a packet communication system represented by Internet communication, mobile communication system and so on, compression/coding techniques are often used to improve the transmission efficiency of speech/sound signals. Furthermore, in the recent years, while speech/sound signals are being encoded simply at low bit rates, there is a growing demand for techniques for encoding speech/sound signals of wider band.

[0003] To meet this demand, studies are underway to develop various techniques for encoding wideband speech/sound signals without drastically increasing the amount of encoded information. For example, patent document 1 discloses a technique of generating features of the high frequency band region in the spectral data obtained by converting an input acoustic signal of a certain period, as side information, and outputting this information together with encoded information of the low band region. To be more specific, the spectral data of the high frequency band region is divided into a plurality of groups, and, in each group, regards the spectrum of the low band region that is the most similar to the spectrum of the group, as the side information mentioned above.

[0004] Furthermore, patent document 2 discloses a technique of dividing the high band signal into a plurality of subbands, deciding, per subband, the degree of similarity between the signal of each subband and the low band signal, and changing the configurations of side information (i.e. the amplitude parameter of the subband, position parameter of a similar low band signal, residual signal parameter between the high band the and the low band) according to the decision result.

[0005] Document WO 2005111568 (A1) describes an encoding device capable of appropriately adjusting the dynamic range of spectrum inserted according to the technique for replacing a spectrum of a certain band with a spectrum of another band. The device includes a spectrum deformation unit which deforms a first spectrum S1(k) of the band 0 <= k < FL in various ways to change the dynamic range so that a way of deformation for obtaining an appropriate dynamic range is checked. The information concerning the deformation is encoded and given to a multiplexing unit. By using a second spectrum S2(k) having a valid signal band 0 <= k <FH as a reference signal, an extended band spectrum encoding unit estimates a spectrum (spectrum of extended band) to be contained in the higher range (FL <= k < FH) of the first spectrum S1(k) according to the first spectrum S1'(k) after the deformation, encodes the information concerning the estimation spectrum, and gives it to the multiplexing unit.

[0006] A two stage hybrid embedded speech/audio coding structure that aims at providing the minimal bit rate and an acceptable performance on speech inputs by using a speech coder as a core is described by RAMPRASHAD S A in "A two stage hybrid embedded speech/audio coding structure" ACOUSTICS, SPEECH AND SIGNAL PROCESSING, 1998. PROCEEDINGS OF THE 1998 IEEE INTERNATIONAL CONFERENCE ON SEATTLE , WA, USA 12-15 MAY 1998, NEW YORK, NY, USA, IEEE, US, vol. 1, 12 May 1998 (1998-05-12), pages 337-340, Xp010279163 ISBN: 978-0-7803-4428-0.

[0007] A bit rate scalable perceptual coder for MPEG-4 Audio for allowing to decode useful subsets of the bitstreams is described in GRILL B: "A bit rate scalable perceptual code for MPEG-4 audio" AUDIO ENGINEERING SOCIETY. CONVENTION PREPRINT, XX, XX 26 September 1997 (1997-09-26), XP002302435.

Patent Document 1: Japanese Patent Application Laid-Open No.2003-140692

Patent Document 2: Japanese Patent Application Laid-Open No.2004-004530

Disclosure of Invention

Problems to be Solved by the Invention

[0008] However, although the techniques disclosed in above-described patent document 1 and patent document 2 decide a low band signal that correlates with or that is similar to a high band region to generate a high band signal (i.e. spectral data of a high band region), this is performed per subband (group) of the high band signal, and, as a result, the amount of processing of calculations becomes enormous. Furthermore, since the above-described processing is carried out on a per band basis, not only the amount of calculation, but also the amount of information required to encode side information increases.

[0009] Furthermore, the techniques disclosed in above-described patent document 1 and patent document 2 decide the degree of similarity of spectral data of the high band region of an input signal in the same way as spectral data of the low band region of the input signal, and, given that spectral data of the low band region is not taken into account if it is distorted by quantization, a severe sound quality degradation is anticipated when spectral data of the low band region is distorted by quantization.

[0010] It is therefore an object of the present invention to provide a encoding apparatus and encoding method that make it possible to encoding spectral data of the high band region of a wideband signal based on spectral data of the low band region of the signal with a very little amount of information and calculation processing and furthermore obtain a decoded signal of high quality even when a severe quantization distortion occurs in the spectral data of the low band region.

Means for Solving the Problem

[0011] This object is solved by the present invention as claimed in the independent claims. Advantageous embodiments of the invention are defined by the dependent claims.

Advantageous Effect of the Invention

[0012] In accordance with the present invention, it is possible to encode spectral data of the high band region of a wideband signal based on spectral data of the low band region of the wideband signal with a very little amount of information and calculation processing and furthermore obtain a decoded signal of high quality even when a severe quantization distortion occurs in the spectral data of the low band region.

Brief Description of Drawings

[0013]

FIG.1 is a block diagram showing a configuration of a communication system provided with a encoding apparatus and decoding apparatus according to Embodiments 1 and 2 of the present invention;
FIG.2 is a block diagram showing a configuration of the encoding apparatus shown in FIG.1;
FIG.3 is a block diagram showing an internal configuration of the low band encoding section shown in FIG.2;
FIG.4 is a block diagram showing an internal configuration of the low band decoding section shown in FIG.2;
FIG.5 is a block diagram showing an internal configuration of the high band encoding section shown in FIG.2;
FIG.6 shows, conceptually, a similar-part search by the a similar-part search section shown in FIG.5;
FIG.7 shows, conceptually, the processing in the amplitude ratio adjusting section shown in FIG.5;
FIG.8 is a block diagram showing a configuration of the decoding apparatus shown in FIG.1; and
FIG.9 is a block diagram showing an internal configuration of the high band decoding section shown in FIG.8.

Best Mode for Carrying Out the Invention

[0014] Embodiments of the present invention will be explained below in detail with reference to the accompanying drawings.

(Embodiment 1)

[0015] FIG.1 is a block diagram showing a configuration of a communication system with a encoding apparatus and decoding apparatus according to Embodiment 1 of the present invention. In FIG.1, the communication system is provided with a encoding apparatus and decoding apparatus, which are able to communicate with each other via a channel. The channel may be wireless or wired or may be both wireless and wired.

[0016] Encoding apparatus 101 divides an input signal every N samples (N is a natural number), regards N samples one frame, and performs encoding per frame. Here, suppose the input signal to be encoded is expressed as "x_n" (n=0, ..., N-1). n indicates the (n+1)-th signal element of the input signal divided every N samples. The encoded input information (i.e. encoded information) is transmitted to decoding apparatus 103 via channel 102.

[0017] Decoding apparatus 103 receives the encoded information transmitted from encoding apparatus 101 via channel 102, decodes the signal and obtains an output signal.

[0018] FIG.2 is a block diagram showing an internal configuration of encoding apparatus 101 shown in FIG.1. When the sampling frequency of the input signal is SR_{input}, down-sampling processing section 201 down-samples the sampling frequency of the input signal from SR_{input} to SR_{base} ($SR_{base}<SR_{input}$), and outputs the down-sampled input signal to low band encoding section 202 as the down-sampled input signal.

[0019] Low band encoding section 202 encodes the down-sampled input signal outputted from down-sampling processing section 201 using a CELP type speech encoding method, to generate a low band component encoded information, and outputs the low band component encoded information generated, to low band decoding section 203 and encoded information integration section 207. The details of low band encoding section 202 will be described later.

[0020] Low band decoding section 203 decodes the low band component encoded information outputted from low band encoding section 202 using a CELP type speech decoding method, to generate a low band component decoded signal, and outputs the low band component decoded signal generated, to up-sampling processing section 204. The details of low band decoding section 203 will be described later.

[0021] Up-sampling processing section 204 up-samples the sampling frequency of the low band component decoded signal outputted from low band decoding section 203 from SR_{base} to SR_{input}, and outputs the up-sampled low band component decoded signal to orthogonal transform processing section 205 as the up-sampled low band component decoded signal.

[0022] Orthogonal transform processing section 205 contains buffers buf 1_n and buf 2_n (n=0, ..., N-1) in association with the aforementioned signal elements, and initializes the buffers using 0 as the initial value according to equation 1 and equation 2, respectively.

[1]

$$buf1_n = 0 \quad (n = 0, \cdots, N-1) \quad \ldots \text{ (Equation 1)}$$

[2]

$$buf2_n = 0 \quad (n = 0, \cdots, N-1) \quad \ldots \text{ (Equation 2)}$$

[0023] Next, as for the orthogonal transform processing in orthogonal transform processing section 205, the calculation procedures and data output to the internal buffers will be explained.

[0024] Orthogonal transform processing section 205 applies the modified discrete cosine transform ("MDCT") to input signal x_n and up-sampled low band component decoded signal y_n outputted from up-sampling processing section 204 and calculates MDCT coefficients X_k of the input signal and MDCT coefficients Y of up-sampled low band component decoded signal y_n according to equation 3 and equation 4.

[3]

$$X_k = \frac{2}{N} \sum_{n=0}^{2N-1} x'_n \cos\left[\frac{(2n+1+N)(2k+1)\pi}{4N}\right] \quad (k = 0, \cdots, N-1) \quad \ldots \text{ (Equation 3)}$$

[4]

$$Y_k = \frac{2}{N} \sum_{n=0}^{2N-1} y'_n \cos\left[\frac{(2n+1+N)(2k+1)\pi}{4N}\right] \quad (k = 0, \cdots, N-1) \quad \ldots \text{ (Equation 4)}$$

[0025] Here, k is the index of each sample in a frame. Orthogonal transform processing section 205 calculates x_n', which is a vector combining input signal x_n and buffer buf 1_n, according to following equation 5. Furthermore, orthogonal transform processing section 205 calculates y_n', which is a vector combining up-sampled low band component decoded signal y_n and buffer buf 2_n, according to following equation 6.

[5]

$$x'_n = \begin{cases} buf1_n & (n = 0, \cdots N-1) \\ x_{n-N} & (n = N, \cdots 2N-1) \end{cases} \quad \ldots \text{(Equation 5)}$$

[6]

$$y'_n = \begin{cases} buf2_n & (n = 0, \cdots N-1) \\ y_{n-N} & (n = N, \cdots 2N-1) \end{cases} \quad \ldots \text{(Equation 6)}$$

[0026] Next, orthogonal transform processing section 205 updates buffers buf 1_n and buf 2_n according to equation 7 and equation 8.

[7]

$$buf1_n = x_n \quad (n = 0, \cdots N-1) \quad \ldots \text{(Equation 7)}$$

[8]

$$buf2_n = y_n \quad (n = 0, \cdots N-1) \quad \ldots \text{(Equation 8)}$$

[0027] Orthogonal transform processing section 205 outputs the MDCT coefficients X_k of the input signal and MDCT coefficients Y_k of the up-sampled low band component decoded signal, to high band encoding section 206.

[0028] High band encoding section 206 generates a high band component encoded information from the values of MDCT coefficients X_k of the input signal outputted from orthogonal transform processing section 205 and MDCT coefficients Y_k of the up-sampled low band component decoded signal, and outputs the high band component encoded information generated, to encoded information integration section 207. The details of high band encoding section 206 will be described later.

[0029] Encoded information integration section 207 integrates the low band component encoded information outputted from low band encoding section 202 with the high band component encoded information outputted from high band encoding section 206, adds, if necessary, a transmission error code and so on, to the integrated encoded information, and outputs the resulting code to channel 102 as encoded information.

[0030] Next, the internal configuration of low band encoding section 202 shown in FIG.2 will be explained using FIG. 3. Here, a case where low band encoding section 202 performs CELP type speech encoding, will be explained.

[0031] Pre-processing section 301 performs high pass filter processing of removing the DC component, waveform shaping processing or pre-emphasis processing, with the input signal, to improve the performance of subsequent encoding processing, and outputs the signal (Xin) subjected to such processing to LPC analysis section 302 and addition section 305.

[0032] LPC analysis section 302 performs a linear predictive analysis using Xin outputted from pre-processing section 301, and outputs the analysis result (linear predictive analysis coefficient) to LPC quantization section 303.

[0033] LPC quantization section 303 performs quantization processing of the linear predictive coefficient (LPC) outputted from LPC analysis section 302, outputs the quantized LPC to synthesis filter 304 and also outputs a code (L) representing the quantized LPC, to multiplexing section 314.

[0034] Synthesis filter 304 performs a filter synthesis on an excitation outputted from addition section 311 (described later) using a filter coefficient based on the quantized LPC outputted from LPC quantization section 303, generates a synthesized signal and outputs the synthesized signal to addition section 305.

[0035] Addition section 305 inverts the polarity of the synthesized signal outputted from synthesis filter 304, adds the synthesized signal with an inverse polarity to Xin outputted from pre-processing section 301, thereby calculating an error signal, and outputs the error signal to perceptual weighting section 312.

[0036] Adaptive excitation codebook 306 stores excitation outputted in the past from addition section 311 in a buffer, extracts one frame of samples from the past excitation specified by the signal outputted from parameter determining section 313 (described later) as an adaptive excitation vector, and outputs this vector to multiplication section 309.

[0037] Quantization gain generation section 307 outputs a quantization adaptive excitation gain and quantization fixed excitation gain specified by the signal outputted from parameter determining section 313, to multiplication section 309 and multiplication section 310, respectively.

[0038] Fixed excitation codebook 308 outputs a pulse excitation vector having a shape specified by a signal outputted from parameter determining section 313, to multiplication section 310 as a fixed excitation vector. A vector produced by multiplying the pulse excitation vector by a spreading vector may also be outputted to multiplication section 310 as a fixed excitation vector.

[0039] Multiplication section 309 multiplies the adaptive excitation vector outputted from adaptive excitation codebook 306 by the quantization adaptive excitation gain outputted from quantization gain generation section 307, and outputs the multiplication result to addition section 311. Furthermore, multiplication section 310 multiplies the fixed excitation vector outputted from fixed excitation codebook 308 by the quantization fixed excitation gain outputted from quantization gain generation section 307, and outputs the multiplication result to addition section 311.

[0040] Addition section 311 adds up the adaptive excitation vector multiplied by the gain outputted from multiplication section 309 and the fixed excitation vector multiplied by the gain outputted from multiplication section 310, and outputs an excitation, which is the addition result, to synthesis filter 304 and adaptive excitation codebook 306. The excitation outputted to adaptive excitation codebook 306 is stored in the buffer of adaptive excitation codebook 306.

[0041] Perceptual weighting section 312 assigns perceptual a weight to the error signal outputted from addition section 305, and outputs the resulting error signal to parameter determining section 313 as the coding distortion.

[0042] Parameter determining section 313 selects the adaptive excitation vector, fixed excitation vector and quantization gain that minimize the coding distortion outputted from perceptual weighting section 312 from adaptive excitation codebook 306, fixed excitation codebook 308 and quantization gain generation section 307, respectively, and outputs an adaptive excitation vector code (A), fixed excitation vector code (F) and quantization gain code (G) showing the selection results, to multiplexing section 314.

[0043] Multiplexing section 314 multiplexes the code (L) showing the quantized LPC outputted from LPC quantization section 303, the adaptive excitation vector code (A), fixed excitation vector code (F) and quantization gain code (G) outputted from parameter determining section 313 and outputs the multiplexed code to low band decoding section 203 and encoded information integration section 207 as a low band component encoded information.

[0044] Next, an internal configuration of low band decoding section 203 shown in FIG.2 will be explained using FIG. 4. Here, a case where low band decoding section 203 performs CELP type speech decoding will be explained.

[0045] Demultiplexing section 401 divides the low band component encoded information outputted from low band encoding section 202 into individual codes (L), (A), (G) and (F). The divided LPC code (L) is outputted to LPC decoding section 402, the divided adaptive excitation vector code (A) is outputted to adaptive excitation codebook 403, the divided quantization gain code (G) is outputted to quantization gain generation section 404 and the divided fixed excitation vector code (F) is outputted to fixed excitation codebook 405.

[0046] LPC decoding section 402 decodes the quantized LPC from the code (L) outputted from demultiplexing section 401, and outputs the decoded quantized LPC to synthesis filter 409.

[0047] Adaptive excitation codebook 403 extracts one frame of samples from the past excitation specified by the adaptive excitation vector code (A) outputted from demultiplexing section 401 as an adaptive excitation vector and outputs the adaptive excitation vector to multiplication section 406.

[0048] Quantization gain generation section 404 decodes the quantization adaptive excitation gain and quantization fixed excitation gain specified by the quantization gain code (G) outputted from demultiplexing section 401, outputs the quantization adaptive excitation gain to multiplication section 406 and outputs the quantization fixed excitation gain to multiplication section 407.

[0049] Fixed excitation codebook 405 generates a fixed excitation vector specified by the fixed excitation vector code (F) outputted from demultiplexing section 401, and outputs the fixed excitation vector to multiplication section 407.

[0050] Multiplication section 406 multiplies the adaptive excitation vector outputted from adaptive excitation codebook 403 by the quantization adaptive excitation gain outputted from quantization gain generation section 404, and outputs the multiplication result to addition section 408. Furthermore, multiplication section 407 multiplies the fixed excitation vector outputted from fixed excitation codebook 405 by the quantization fixed excitation gain outputted from quantization gain generation section 404, and outputs the multiplication result to addition section 408.

[0051] Addition section 408 adds up the adaptive excitation vector multiplied by the gain outputted from multiplication section 406 and the fixed excitation vector multiplied by the gain outputted from multiplication section 407 to generate an excitation, and outputs the excitation to synthesis filter 409 and adaptive excitation codebook 403.

[0052] Synthesis filter 409 performs a filter synthesis of the excitation outputted from addition section 408 using the filter coefficient decoded by LPC decoding section 402, and outputs the synthesized signal to post-processing section 410.

[0053] Post-processing section 410 applies processing for improving the subjective quality of speech such as formant emphasis and pitch emphasis and processing for improving the subjective quality of stationary noise, to the signal outputted from synthesis filter 409, and outputs the resulting signal to up-sampling processing section 204 as a low band component decoded signal.

[0054] Next, an internal configuration of high band encoding section 206 shown in FIG.2 will be explained using FIG. 5. A similar-part search section 501 calculates the search result position t_{MIN} ($t=t_{MIN}$) of when the error D between MDCT coefficients Y_k of the up-sampled low band component decoded signal outputted from orthogonal transform processing section 205 and M samples from the beginning of MDCT coefficients X_k of the input signal outputted from orthogonal transform processing section 205,
becomes a minimum, and gain β at that moment. The error D and gain β can be calculated from equation 9 and equation 10, respectively.

[9]

$$D=\sum_{i=0}^{M-1} X^i \cdot X^i - \frac{\left(\sum_{i=0}^{M-1} X^i \cdot Y_t^i\right)^2}{\sum_{i=0}^{M-1} Y_t^i \cdot Y_t^i} \quad \ldots \text{ (Equation 9)}$$

[10]

$$\beta=\frac{\sum_{i=0}^{M-1} X^i \cdot Y_{t_{MIN}}^i}{\sum_{i=0}^{M-1} Y_{t_{MIN}}^i \cdot Y_{t_{MIN}}^i} \quad \ldots \text{ (Equation 10)}$$

[0055] Here, FIG.6A and FIG.6B conceptually show a similar-part search by a similar-part search section 501. FIG. 6A shows an input signal spectrum, and shows the beginning part of the high band region (3.5 kHz to 7.0 kHz) of the input signal in a frame. FIG.6B shows a situation in which a spectrum similar to the spectrum inside the frame shown in FIG.6A is searched for sequentially from the beginning of the low band region of a decoded signal.

[0056] A similar-part search section 501 outputs MDCT coefficients X_k of the input signal, MDCT coefficients Y_k of the up-sampled low band component decoded signal, and calculated search result position t_{MIN} and gain β, to amplitude ratio adjusting section 502.

[0057] Amplitude ratio adjusting section 502 extracts the part from search result position t_{MIN} to $SR_{base}/SR_{input} \times (N-1)$ (if X_k becomes zero in the middle, the part up the position before X_k becomes zero), from MDCT coefficients Y_k of an up-sampled low band component decoded signal, and multiplies this pa-rt by gain β and designates the resulting value as copy source spectral data $Z1_k$, expressed by equation 11.

[11]

$$Z1_k = Y_k \cdot \beta \quad \left(k = t_{MIN}, \ldots, SR_{base}/SR_{input} \cdot N - 1\right) \quad \ldots \text{ (Equation 11)}$$

[0058] Next, amplitude ratio adjusting section 502 generates temporary spectral data $Z2_k$ from copy source spectral data $Z1_k$. To be more specific, amplitude ratio adjusting section 502 divides the length $((1-SR_{base}/SR_{input}) \times N)$ of the spectral data of the high band component by the length $(SR_{base}/SR_{input} \times N-1-t_{MIN})$ of copy source spectral data $Z1_k$, repeats copying the source spectral data $Z1_k$ a number of times equaling the quotient such that source spectral data $Z1_k$ continues from the part of $k=SR_{base}/SR_{input} \times N-1$ of temporary spectral data $Z2_k$, and then copies copy source spectral data $Z1_k$ for a number of samples equaling the samples of the remainder after dividing the length $((1-SR_{base}/SR_{input}) \times N)$ of the spectral data of the high band component by the length $(SR_{base}/SR_{input} \times N-1-t_{MIN})$ of copy source spectral data $Z1_k$, from the beginning of copy source spectral data $Z1_k$, to the tail end of temporary spectral data $Z2_k$.

[0059] Furthermore, suppose, when X_k becomes zero in the middle, amplitude ratio adjusting section 502 adds the length of the part where X_k is zero to the length $((1-SR_{base}/SR_{input})\times N)$ of the spectral data of the aforementioned high band component, and starts copying copy source spectral data $Z1_k$ to temporary spectral data $Z2_k$ from the part where X_k is zero in the middle.

[0060] Next, amplitude ratio adjusting section 502 adjusts the amplitude ratio of temporary spectral data $Z2_k$. To be more specific, amplitude ratio adjusting section 502 divides MDCT coefficients X_k of the input signal and the high band component ($k=SR_{base}/SR_{input}\times N$, ..., N-1) of temporary spectral data $Z2_k$ into a plurality of bands first.

[0061] Here, a case where temporary spectral data $Z2_k$ is copied from the part of $k=SR_{base}/SR_{input}\times N$ in the aforementionedprocessing, will be explained. Amplitude ratio adjusting section 502 calculates amplitude ratio α_j for each band as expressed by equation 12 for MDCT coefficients X_k of the input signal and the high band component of temporary spectral data $Z2_k$. In equation 12, suppose "NUM BAND" is the number of bands and "band index(j)" is the minimum sample index out of the indexes making up band j.

[12]

$$\alpha_j = \sqrt{\frac{\sum_{k=band\_index(j)}^{band\_index(j+1)-1} X_k}{\sum_{k=band\_index(j)}^{band\_index(j+1)-1} Z2_k}} \quad (j = 0, \dots, NUM\_BAND - 1)$$

... (Equation 12)

[0062] FIG.7 shows, conceptually, the processing in amplitude ratio adjusting section 502. FIG.7 shows a situation in which the spectrum of the high band region is generated based on the similar-part searched from the low band region in FIG.6(b) (when NUM BAND=5).

[0063] Amplitude ratio adjusting section 502 outputs amplitude ratio α_j for each band obtained from equation 12, search result position t_{MIN} and gain β to quantization section 503.

[0064] Quantization section 503 quantizes amplitude ratio α_j for each band, search result position t_{MIN} and gain β outputted from amplitude ratio adjusting section 502 using codebooks provided in advance and outputs the index of each codebook, to encoded information integration section 207 as a high band component encoded information.

[0065] Here, suppose amplitude ratio α_j for each band, search result position t_{MIN} and gain β are quantized all separately and the selected codebook indexes are code_A, code_T and code_B, respectively. Furthermore, a quantization method is employed here whereby the code vector (or code) having the minimum distance (i.e. square error) to the quantization target is selected from the codebooks. However, this quantization method is in the public domain and will not be described in detail.

[0066] FIG.8 is a block diagram showing an internal configuration of decoding apparatus 103 shown in FIG.1. Encoded information division section 601 divides the low band component encoded information and the high band component encoded information from the inputted encoded information, outputs the divided low band component encoded information to low band decoding section 602, and outputs the divided high band component encoded information to high band decoding section 605.

[0067] Low band decoding section 602 decodes the low band component encoded information outputted from encoded information division section 601 using a CELP type speech decoding method, to generate a low band component decoded signal and outputs the low band component decoded signal generated to up-sampling processing section 603. Since the configuration of low band decoding section 602 is the same as that of aforementioned low band decoding section 203, its detailed explanations will be omitted.

[0068] Up-sampling processing section 603 up-samples the sampling frequency of the low band component decoded signal outputted from low band decoding section 602 from SR_{base} to SR_{input}, and outputs the up-sampled low band component decoded signal to orthogonal transform processing section 604 as the up-sampled low band component decoded signal.

[0069] Orthogonal transform processing section 604 applies orthogonal transform processing (MDCT) to the up-sampled low band component decoded signal outputted from up-sampling processing section 603, calculates MDCT coefficients Y'_k of the up-sampled low band component decoded signal and outputs this MDCT coefficients Y'_k to high band decoding section 605. The configuration of orthogonal transform processing section 604 is the same as that of a forementioned orthogonal transform processing section 205, and therefore detailed explanations thereof will be omitted.

[0070] High band decoding section 605 generates a signal including the high band component from MDCT coefficients Y'_k of the up-sampled low band component decoded signal outputted from orthogonal transform processing section 604 and the high band component encoded information outputted from encoded information division section 601, and makes this the output signal.

[0071] Next, an internal configuration of high band decoding section 605 shown in FIG.8 will be explained using FIG. 9. Dequantization section 701 dequantizes the high band component encoded information (i.e. code_A, code_T and code_B) outputted from encoded information division section 601 for the codebooks provided in advance, and outputs amplitude ratio α_j for each band produced, search result position t_{MIN} and gain β, to similar-part generation section 702. To be more specific, the vectors and values indicated by the high band component encoded information (i.e. code_A, code_T and code_B) from each codebook are outputted to similar-part generation section 702 as amplitude ratio α_j for each band, search result position t_{MIN} and gain β, respectively. Here, suppose amplitude ratio α_j for each band, search result position t_{MIN} and gain β are dequantized using different codebooks as in the case of quantization section 503.

[0072] Similar-part generation section 702 generates a high band component ($k=SR_{base}/SR_{input}\times N, ..., N-1$) of MDCT coefficients Y' from MDCT coefficients Y'_k of the up-sampled low band component outputted from orthogonal transform processing section 604 and search position result t_{MIN} outputted from dequantization section 701 and gain β. To be more specific, copy source spectral data $Z1'_k$ is generated according to equation 13.

[13]

$$Z1'_k = Y'_k \cdot \beta \quad \left(k = t_{MIN}, \ldots, SR_{base} / SR_{input} \cdot N - 1\right) \quad \ldots \text{ (Equation 13)}$$

[0073] Furthermore, suppose, when Y'_k is zero in the middle, copy source spectral data $Z1'_k$ covers the part from the position where k is t_{MIN} up to the position before Y'_k becomes zero, according to equation 13.

[0074] Next, similar-part generation section 702 generates temporary spectral data $Z2'_k$ from copy source spectral data $Z1'_k$ calculated according to equation 13. To be more specific, similar-part generation section 702 divides the length $((1-SR_{base}/SR_{input})\times N)$ of the spectral data of the high band component by the length $(SR_{base}/SR_{input}\times N-1-t_{MIN})$ of copy source spectral data $Z1'_k$, repeats copying copy source spectral data $Z1'_k$ a number of time equaling the quotient such that copy source spectral data $Z1'_k$ continues from the part of $k=SR_{base}/SR_{input}\times N-1$ of temporary spectral data $Z2'_k$, and then copies copy source spectral data $Z1'_k$ for a number of samples equaling the samples of the remainder after dividing the length $((1-SR_{base}/SR_{input})$ xN) of the spectral data of the high band component by the length $(SR_{base}/SR_{input}\times N-1-t_{MIN})$ of copy source spectral data $Z1'_k$ from the beginning of copy source spectral data $Z1'_k$ to the tail end of temporary spectral data $Z2'_k$.

[0075] Furthermore, suppose, when Y'_k becomes zero in the middle, similar-part generation section 702 adds the length of the part where Y'_k is zero, to the length $((1-SR_{base}/SR_{input})\times N)$ of the spectral data of the aforementioned high band component, and starts copying copy source spectral data $Z1'_k$ to temporary spectral data $Z2'_k$ from the part where Y'_k is zero in the middle.

[0076] Next, similar-part generation section 702 copies the value of the low band component of Y'_k to the low band component of temporary spectral data $Z2'_k$, expressed by equation 14. Here, a case where the temporary spectral data $Z2'_k$ is copied from the part of $k=SR_{base}/SR_{input}\times N$ in the aforementioned processing, will be explained.

[14]

$$Z2'_k = Y'_k \quad \left(k = 0, \ldots, SR_{base} / SR_{input} \cdot N - 1\right) \quad \ldots \text{ (Equation 14)}$$

[0077] Similar-part generation section 702 outputs the calculated temporary spectral data $Z2'_k$ and amplitude ratio α_j per band, to amplitude ratio adjusting section 703.

[0078] Amplitude ratio adjusting section 703 calculates temporary spectral data $Z3'_k$ from temporary spectral data $Z2'_k$ and amplitude ratio α_j for each band outputted from similar-part generation section 702, expressed by equation 15. Here, α_j in equation 15 is the amplitude ratio of each band and band_index (j) is the minimum sample index in the indexes making up band j.

[15]

$$Z3'_k = \begin{cases} Z2'_k & (k = 0, \ldots, SR_{base} / SR_{input} \cdot N - 1) \\ Z2'_k \cdot \alpha_j & (k = SR_{base} / SR_{input} \cdot N, \ldots, N-1 : band\_index(j) \le k < band\_index(j+1)) \\ & (j = 0, \ldots, NUM\_BAND - 1) \end{cases}$$

... (Equation 15)

[0079] Amplitude ratio adjusting section 703 outputs temporary spectral data $Z3'_k$ calculated according to equation 15 to orthogonal transform processing section 704.

[0080] Orthogonal transform processing section 704 contains buffer buf'_k and is initialized according to equation 16.

[16]

$$buf'_k = 0 \quad (k = 0, \cdots, N-1)$$ ... (Equation 16)

[0081] Orthogonal transform processing section 704 calculates decoded signal Y''_n using temporary spectral data Z3'k outputted from amplitude ratio adjusting section 703, according to equation 17.

[17]

$$Y''_n = \frac{2}{N} \sum_{n=0}^{2N-1} Z3''_k \cos\left[\frac{(2n+1+N)(2k+1)\pi}{4N}\right] \quad (n = 0, \cdots, N-1)$$

... (Equation 17)

[0082] Here, $Z3''_k$ is a vector combining temporary spectral data $Z3'_k$ and buffer buf'_k and is calculated according to equation 18.

[18]

$$Z3''_k = \begin{cases} buf'_k & (k = 0, \cdots N-1) \\ Z3'_k & (k = N, \cdots 2N-1) \end{cases}$$ ... (Equation 18)

[0083] Next, orthogonal transform processing section 704 updates buffer buf'_k according to equation 19.

[19]

$$buf'_k = Z3'_k \quad (k = 0, \cdots N-1)$$ ... (Equation 19)

[0084] Orthogonal transform processing section 704 obtains decoded signal Y''_n as an output signal.

[0085] In this way, in accordance with Embodiment 1, to generate spectral data of the high band region of a signal to be encoded based on spectral data of the low band region of the signal, a similar-part search is performed for a part (e.g. beginning part) in the spectral data of the high band region, in the quantized low band region, and spectral data of the high band region is generated based on the search result, so that it is possible to encode spectral data of the high band region of a wideband signal based on spectral data of the low band region with an extremely small amount of information and amount of calculation processing, and, furthermore, obtain a decoded signal of high quality even when a significant quantization distortion occurs in the spectral data of the low band region.

(Embodiment 2)

[0086] Embodiment 1 has explained a method of performing a similar-part search with respect to MDCT coefficients of up-sampled low band component decoded signal, and the beginning part of high band components of MDCT coefficients of an input signal, and calculating parameters for generating MDCT coefficients for the high band component at the time of decoding. Now, with embodiment 2, a weighted similar-part search method will be described, whereby, in high band components of the MDCT coefficients of an input signal, lower band components are regarded more important.

[0087] Since the communication system according to Embodiment 2 is similar to the configuration of Embodiment 1 shown in FIG.1, FIG.1 will be used, and furthermore, since the encoding apparatus according to Embodiment 2 of the present invention is similar to the configuration of Embodiment 1 shown in FIG.2, FIG.2 will be used and overlapping explanations will be omitted. However, in the configuration shown in FIG.2, high band encoding section 206 has a function different from that in Embodiment 1, and therefore high band encoding section 206 will be explained using FIG.5.

[0088] Similar-part search section 501 calculates a search result position t_{MIN} ($t=t_{MIN}$) when error D2 between MDCT coefficients Y_k of an up-sampled low band component decoded signal outputted from orthogonal transform processing section 205 and M (M is an integer equal to or greater than 2) samples from the beginning of MDCT coefficients X_k of the input signal outputted from orthogonal transform processing section 205 becomes a minimum, and gain β2 at that moment. Error D2 and β2 are calculated according to equation 20 and equation 21, respectively.

[20]

$$D2 = \left(\sum_{i=0}^{M-1} X^i \cdot X^i - \frac{\left(\sum_{i=0}^{M-1} X^i \cdot Y_t^i \right)^2}{\sum_{i=0}^{M-1} Y_t^i \cdot Y_t^i} \right) \cdot W_i \quad \ldots \text{ (Equation 20)}$$

[21]

$$\beta 2 = \frac{\sum_{i=0}^{M-1} X^i \cdot Y_{t_{MIN}}^i}{\sum_{i=0}^{M-1} Y_{t_{MIN}}^i \cdot Y_{t_{MIN}}^i} \quad \ldots \text{ (Equation 21)}$$

[0089] Here, W_i in equation 20 is a weight having a value of about 0.0 to 1.0, and is multiplied when error D2 (i.e. distance) is calculated. To be more specific, a smaller error sample index (that is, an MDCT coefficients of a lower band region), is assigned a greater weight. An example of W_i is shown in equation 22.

[22]

$$W_i = -\frac{0.5}{M-1} i + 1.0 \quad (i = 0, \ldots, M-1,\ M \geq 2) \quad \ldots \text{ (Equation 22)}$$

[0090] In this way, by calculating the distance using a greater weight for MDCT coefficients of lower band, it is possible to realize a search placing the emphasis on the distortion in the part connecting the low band component and the high band component.

[0091] The configurations of amplitude ratio adjusting section 502 and quantization section 503 are the same as those for the processing explained in Embodiment 1, and therefore detailed explanations thereof will be omitted.

[0092] Encoding apparatus 101 has been explained so far. The configuration of decoding apparatus 103 is the same as explained in Embodiment 1, and therefore detailed explanations thereof will be omitted.

[0093] In this way, in accordance with Embodiment 2, to generate spectral data of the high band region of a signal to be encoded based on spectral data of the low band region of the signal, the distance is calculated by assigning greater weights to smaller error sample indexes, a similar-part search for part (i.e. beginning part) of spectral data of the high band region is performed in spectral data of the quantized low band region and spectral data of the high band region is generated based on the result of the search, so that it is possible to encode spectral data of the high band region of a wideband signal in high perceptual quality based on spectral data of the low band region of the signal, with a very little amount of information and calculation processing and furthermore obtain a decoded signal of high quality even when a significant quantization distortion occurs in the spectral data of the low band region.

[0094] The present embodiment has explained a case where, to generate spectral data of the high band region of a signal to be encoded based on spectral data of the low band region of the signal, a similar-part search for a part (i.e. beginning part) of the spectral data of the high band region is performed in the spectral data of the quantized low band region, so that the present invention is not limited to this and it is equally possible to adopt the above-described weighting in distance calculation for the entire part of the spectral data of the high band region.

[0095] Furthermore, although the present embodiment has explained a method of generating spectral data of the high band region of a signal to be encoded is generated based on spectral data of the low band region of the signal, by calculating the distance by assigning greater weights to smaller error sample indexes, performing a similar-part search for a part (i.e. beginning part) of the spectral data of the high band region in spectral data of the quantized low band region, and generating spectral data of the high band region based on the result of the search, but the present invention is by no means limited to this and may likewise adopt a method of introducing the length of copy source spectral data as an evaluation measure during a search. To be more specific, by making a search result that increases the length of the copy source spectral data, that is, by making an entry of a search position of a low band more likely to be selected, it is possible to further improve the quality of an output signal by reducing the number of discontinuous parts caused when the spectral data of the high band region is copied a plurality of times and placing the discontinuous parts in high frequency bands.

[0096] The above-described embodiments have explained that the index of the MDCT coefficients of the spectral data of the high band region generated starts from $SR_{base}/SR_{input} \times (N-1)$, but the present invention is not limited to this, and the present invention is also applicable to cases where spectral data of the high band region is generated likewise from a part where low band spectral data becomes zero, irrespective of sampling frequencies. Furthermore, the present invention is also applicable to a case where spectral data of the high band region is generated from an index specified from the user and system side.

[0097] The above-described embodiments have explained the CELP type speech encoding scheme in the low band encoding section as an example, but the present invention is not limited to this and is also applicable to cases where a down-sampled input signal is coded according to a speech/sound encoding scheme other than CELP type. The same applies to the low band decoding-section.

[0098] The present invention is further applicable to a case where a signal processing program is recorded or written into a mechanically readable recording medium such as a memory, disk, tape, CD, DVD and operated, and operations and effects similar to those of the present embodiment can be obtained.

[0099] Each function block employed in the description of each of the aforementioned embodiments may typically be implemented as an LSI constituted by an integrated circuit. These may be individual chips or partially or totally contained on a single chip.

[0100] "LSI" is adopted here but this may also be referred to as "IC", "system LSI", "super LSI", or "ultra LSI" depending on differing extents of integration.

[0101] Further, the method of circuit integration is not limited to LSI's, and implementation using dedicated circuitry or general purpose processors is also possible. After LSI manufacture, utilization of an FPGA (Field Programmable Gate Array) or a reconfigurable processor where connections and settings of circuit cells within an LSI can be reconfigured is also possible.

[0102] Further, if integrated circuit technology comes out to replace LSI's as a result of the advancement of semiconductor technology or a derivative other technology, it is naturally also possible to carry out function block integration using this technology. Application of biotechnology is also possible.

Industrial Applicability

[0103] The encoding apparatus and encoding method according to the present invention make it possible to encode spectral data of the high band region of a wideband signal based on spectral data of the low band region of the signal with a very little amount of information and calculation processing, and produce a decoded signal of high quality even when a significant quantization distortion occurs in the spectral data of the low band region, and are therefore applicable for use in , for example, a packet communication system and mobile communication system.

Claims

1. An encoding apparatus comprising:

   a first encoding section (202) adapted to encode a speech/sound input signal to generate first low band encoded information;
   a decoding section (203) adapted to decode the first encoded low band information to generate a decoded signal;
   an orthogonal transform section (205) adapted to orthogonal-transform the input signal and the decoded signal to generate orthogonal transform coefficients for the signals;
   a second encoding section (206) adapted to generate second encoded information representing a high band part in the orthogonal transform coefficients of the decoded signal, based on the orthogonal transform coefficients of the input signal and the orthogonal transform coefficients of the decoded signal; and
   an integration section (207) adapted to integrate the first encoded information and the second encoded information;
   characterized in that
   the second encoding section (206) is adapted to search for a part that is the most similar to a part of the orthogonal transform coefficient of the input signal, in the orthogonal transform coefficients of the decoded signal.

2. The encoding apparatus according to claim 1, wherein the second encoding section (206) searches for said part that is the most similar to the orthogonal transform coefficients of the input signal, in the orthogonal transform coefficients of the decoded signal, the second encoding section (206) performs a similar-part search.

3. The encoding apparatus according to claim 1, wherein the second encoding section (206) is adapted to calculate a first orthogonal transform coefficient using the search result and to adjust an amplitude of the first orthogonal transform coefficient so that the amplitude of the calculated first orthogonal transform coefficient is equal to an amplitude of the orthogonal transform coefficient of the input signal.

4. The encoding apparatus according to claim 1, wherein the first encoding section (202) is adapted to perform encoding using a CELP type encoding method.

5. The encoding apparatus according to claim 1, wherein the second encoding section (206) is adapted to multiply a difference between the orthogonal transform coefficients of the input signal and the orthogonal transform coefficients of the decoded signal by a greater weight for a low frequency region, and, using the multiplication result, to search for a part that is the most similar to the orthogonal transform coefficients of the input signal, in the orthogonal transform coefficient of the decoded signal.

6. The encoding apparatus according to claim 1, wherein the second encoding section (206) is adapted to multiply a difference between the orthogonal transform coefficients of the input signal and the orthogonal transform coefficients of the decoded signal by a weight that causes entries on a low frequency band to be selected as a search position, and, using the multiplication result, to search for a part that is the most similar to the orthogonal transform coefficients of the input signal, in the orthogonal transform coefficients of the decoded signal.

7. An encoding method comprising:

   a first encoding step of encoding a speech/sound input signal to generate first encoded low band information;
   a decoding step of decoding the first encoded low band information to generate a decoded signal;
   an orthogonal transform step of orthogonal-transforming the input signal and the decoded signal to generate orthogonal transform coefficients for the signals;
   a second encoding step of generating second encoded information representing a high band part of the orthog-

onal transform coefficients of the decoded signal based on the orthogonal transform coefficients of the input signal and the orthogonal transform coefficients of the decoded signal; and
an integration step of integrating the first encoded information and the second encoded information;
characterized in that
the second encoding step comprises searching for a part that is the most similar to a part of the orthogonal transform coefficients of the input signal, in the orthogonal transform coefficients of the decoded signal.

8. An encoding program adapted to execute on a computer the method according to claim 7.

Patentansprüche

1. Verschlüsselungsvorrichtung, umfassend:

einen ersten Verschlüsselungsabschnitt (202) zum Verschlüsseln eines Sprach-/Toneingabesignals zur Erzeugung von erster verschlüsselter Niederbandinformation;
einen Entschlüsselungsabschnitt (203) zum Entschlüsseln der ersten verschlüsselten Niederbandinformation zur Erzeugung eines entschlüsselten Signals;
einen Orthogonaltransformationsabschnitt (205) zum Orthogonaltransformieren des Eingabesignals und des entschlüsselten Signals zur Erzeugung von Orthogonaltransformationskoeffizienten für die Signale;
einen zweiten Verschlüsselungsabschnitt (206) zum Erzeugen von zweiter verschlüsselter Information, die eine Hochbandteil in den Orthogonaltransformationskoeffizienten des entschlüsselten Signals darstellt, auf Grundlage der Orthogonaltransformationskoeffizienten des Eingabesignals und der Orthogonaltransformationskoeffizienten des entschlüsselten Signals; und
einen Integrationsabschnitt (207) zum Integrieren der ersten verschlüsselten Information und der zweiten verschlüsselten Information;
dadurch gekennzeichnet, dass
der zweite Verschlüsselungsabschnitt (206) dem Suchen nach einem Teil mit der größten Ähnlichkeit zu einem Teil der Orthogonaltransformationskoeffizienten des Eingabesignals in den Orthogonaltransformationskoeffizienten des entschlüsselten Signals dient.

2. Verschlüsselungsvorrichtung nach Anspruch 1, wobei bei der Suche des zweiten Verschlüsselungsabschnittes (206) nach dem Teil mit der größten Ähnlichkeit zu den Orthogonaltransformationskoeffizienten des Eingabesignals in den Orthogonaltransformationskoeffizienten des entschlüsselten Signals der zweite Verschlüsselungsabschnitt (206) eine Ähnlicher-Teil-Suche durchführt.

3. Verschlüsselungsvorrichtung nach Anspruch 1, wobei der zweite Verschlüsselungsabschnitt (206) dem Berechnen eines ersten Orthogonaltransformationskoeffizienten unter Verwendung des Suchergebnisses und dem Anpassen einer Amplitude des ersten Orthogonaltransformationskoeffizienten derart, dass die Amplitude des berechneten ersten Orthogonaltransformationskoeffizienten gleich einer Amplitude des Orthogonaltransformationskoeffizienten des Eingabesignals ist, dient.

4. Verschlüsselungsvorrichtung nach Anspruch 1, wobei der erste Verschlüsselungsabschnitt (202) dem Durchführen einer Verschlüsselung unter Verwendung eines Verschlüsselungsverfahrens vom CELP-Typ dient.

5. Verschlüsselungsvorrichtung nach Anspruch 1, wobei der zweite Verschlüsselungsabschnitt (206) dem Multiplizieren einer Differenz zwischen den Orthogonaltransformationskoeffizienten des Eingabesignals und den Orthogonaltransformationskoeffizienten des entschlüsselten Signals mit einem größeren Gewicht für einen Niederfrequenzbereich und dem unter Verwendung des Multiplikationsergebnisses erfolgenden Suchen nach einem Teil mit der größten Ähnlichkeit zu den Orthogonaltransformationskoeffizienten des Eingabesignals in den Orthogonaltransformationskoeffizienten des entschlüsselten Signals dient.

6. Verschlüsselungsvorrichtung nach Anspruch 1, wobei der zweite Verschlüsselungsabschnitt (206) dem Multiplizieren einer Differenz zwischen den Orthogonaltransformationskoeffizienten des Eingabesignals und den Orthogonaltransformationskoeffizienten des entschlüsselten Signals mit einem Gewicht, das eine Auswahl von Einträgen in dem Niederfrequenzband als Suchposition bewirkt, und dem unter Verwendung des Multiplikationsergebnisses erfolgenden Suchen nach einem Teil mit der größten Ähnlichkeit zu den Orthogonaltransformationskoeffizienten des Eingabesignals in den Orthogonaltransformationskoeffizienten des entschlüsselten Signals dient.

7. Verschlüsselungsverfahren, umfassend:

einen ersten Verschlüsselungsschritt des Verschlüsselns eines Sprach-/Toneingabesignals zur Erzeugung von erster verschlüsselter Niederbandinformation;
einen Entschlüsselungsschritt des Entschlüsselns der ersten verschlüsselten Niederbandinformation zur Erzeugung eines entschlüsselten Signals;
einen Orthogonaltransformationsschritt des Orthogonaltransformierens des Eingabesignals und des entschlüsselten Signals zur Erzeugung von Orthogonaltransformationskoeffizienten für die Signale;
einen zweiten Verschlüsselungsschritt des Erzeugens von zweiter verschlüsselter Information, die eine Hochbandteil der Orthogonaltransformationskoeffizienten des entschlüsselten Signals darstellt, auf Grundlage der Orthogonaltransformationskoeffizienten des Eingabesignals und der Orthogonaltransformationskoeffizienten des entschlüsselten Signals; und
einen Integrationsschritt des Integrierens der ersten verschlüsselten Information und der zweiten verschlüsselten Information;
dadurch gekennzeichnet, dass
der zweite Verschlüsselungsschritt ein Suchen nach einem Teil mit der größten Ähnlichkeit zu einem Teil der Orthogonaltransformationskoeffizienten des Eingabesignals in den Orthogonaltransformationskoeffizienten des entschlüsselten Signals umfasst.

8. Verschlüsselungsprogramm zum auf einem Computer erfolgenden Ausführen des Verfahrens nach Anspruch 7.

Revendications

1. Appareil de codage comprenant :

une première section de codage (202) adaptée pour coder un signal vocal/sonore d'entrée afin de générer une première information codée à bande basse ;
une section de décodage (203) adaptée pour décoder la première information codée à bande basse afin de générer un signal décodé ;
une section de transformation orthogonale (205) adaptée pour transformer de manière orthogonale le signal d'entrée et le signal décodé afin de générer des coefficients de transformation orthogonale pour les signaux ;
une deuxième section de codage (206) adaptée pour générer une deuxième information codée représentant une partie de bande haute dans les coefficients de transformation orthogonale du signal décodé, sur la base des coefficients de transformation orthogonale du signal d'entrée et des coefficients de transformation orthogonale du signal décodé ; et
une section d'intégration (207) adaptée pour intégrer la première information codée et la deuxième information codée ;
caractérisé en ce que
la deuxième section de codage (206) est adaptée pour chercher une partie qui est la plus similaire à une partie des coefficients de transformation orthogonale du signal d'entrée, dans les coefficients de transformation orthogonale du signal décodé.

2. Appareil de codage selon la revendication 1, dans lequel, lorsque la deuxième section de codage (206) cherche ladite partie qui est la plus similaire aux coefficients de transformation orthogonale du signal d'entrée, dans les coefficients de transformation orthogonale du signal décodé, la deuxième section de codage (206) effectue une recherche de partie similaire.

3. Appareil de codage selon la revendication 1, dans lequel la deuxième section de codage (206) est adaptée pour calculer un premier coefficient de transformation orthogonale en utilisant le résultat de recherche et pour ajuster une amplitude du premier coefficient de transformation orthogonale de sorte que l'amplitude du premier coefficient de transformation orthogonale calculé soit égale à une amplitude du coefficient de transformation orthogonale du signal d'entrée.

4. Appareil de codage selon la revendication 1, dans lequel la première section de codage (202) est adaptée pour effectuer un codage en utilisant un procédé de codage du type CELP.

5. Appareil de codage selon la revendication 1, dans lequel la deuxième section de codage (206) est adaptée pour

multiplier une différence entre les coefficients de transformation orthogonale du signal d'entrée et les coefficients de transformation orthogonale du signal décodé par un plus grand poids pour une région basse fréquence, et, en utilisant le résultat de multiplication, pour chercher une partie qui est la plus similaire aux coefficients de transformation orthogonale du signal d'entrée, dans le coefficient de transformation orthogonale du signal décodé.

6. Appareil de codage selon la revendication 1, dans lequel la deuxième section de codage (206) est adaptée pour multiplier une différence entre les coefficients de transformation orthogonale du signal d'entrée et les coefficients de transformation orthogonale du signal décodé par un poids qui amène des entrées sur une bande basse fréquence à être choisies comme position de recherche, et, en utilisant le résultat de multiplication, pour chercher une partie qui est la plus similaire aux coefficients de transformation orthogonale du signal d'entrée, dans les coefficients de transformation orthogonale du signal décodé.

7. Procédé de codage comprenant :

une première étape de codage qui consiste à coder un signal vocal/sonore d'entrée afin de générer une première information codée à bande basse ;
une étape de décodage qui consiste à décoder la première information codée à bande basse afin de générer un signal décodé ;
une étape de transformation orthogonale qui consiste à transformer de manière orthogonale le signal d'entrée et le signal décodé afin de générer des coefficients de transformation orthogonale pour les signaux ;
une deuxième étape de codage qui consiste à générer une deuxième information codée représentant une partie à bande haute des coefficients de transformation orthogonale du signal décodé sur la base des coefficients de transformation orthogonale du signal d'entrée et des coefficients de transformation orthogonale du signal décodé ; et
une étape d'intégration qui consiste à intégrer la première information codée et la deuxième information codée ;
caractérisé en ce que
la deuxième étape de codage comprend la recherche d'une partie qui est la plus similaire à une partie des coefficients de transformation orthogonale du signal d'entrée, dans les coefficients de transformation orthogonale du signal décodé.

8. Programme de codage adapté pour exécuter le procédé selon la revendication 7 sur un ordinateur.

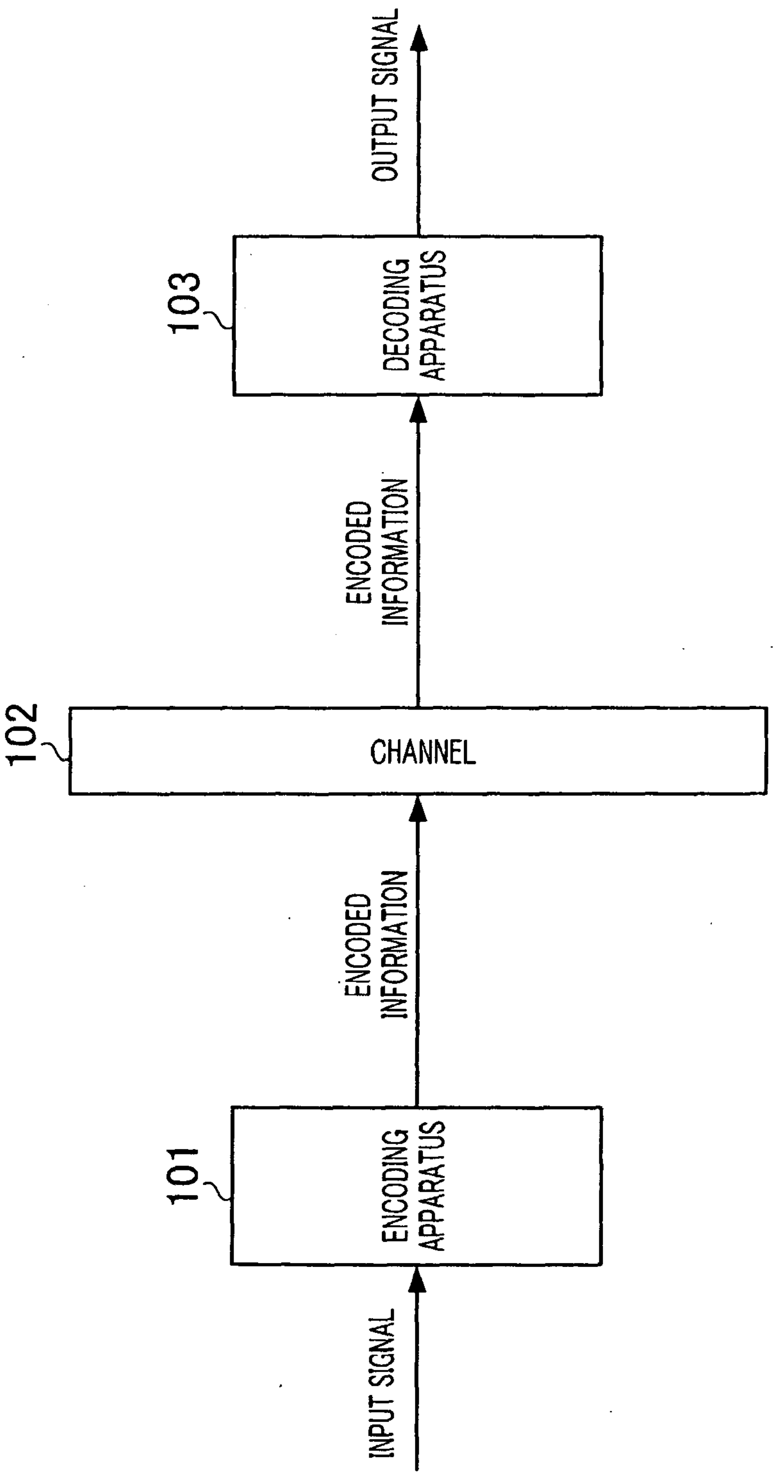
INPUT SIGNAL
101
ENCODING APPARATUS
ENCODED INFORMATION
102
CHANNEL
ENCODED INFORMATION
103
DECODING APPARATUS
OUTPUT SIGNAL

FIG.1

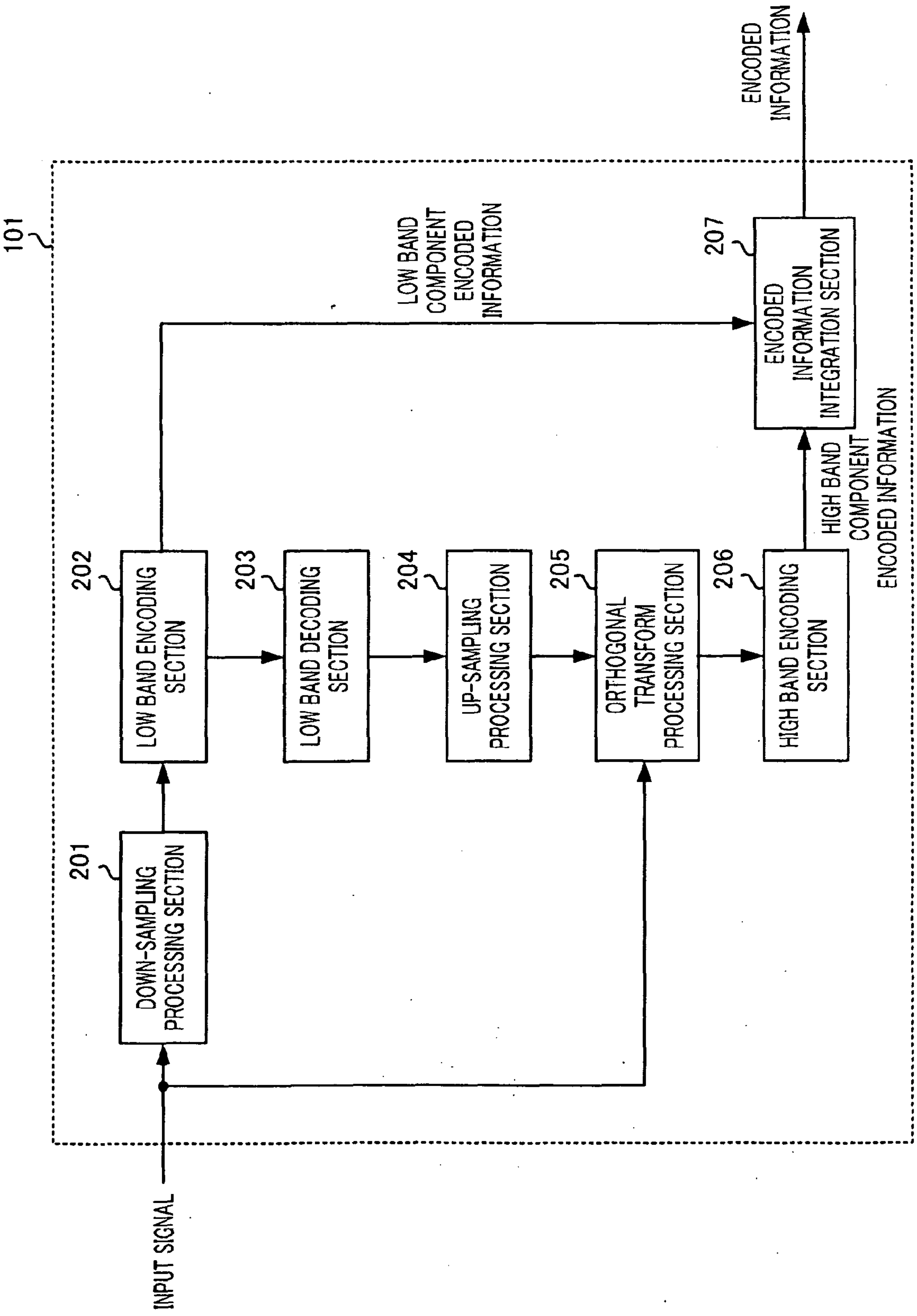
101
INPUT SIGNAL
201
DOWN-SAMPLING PROCESSING SECTION
202
LOW BAND ENCODING SECTION
203
LOW BAND DECODING SECTION
204
UP-SAMPLING PROCESSING SECTION
205
ORTHOGONAL TRANSFORM PROCESSING SECTION
206
HIGH BAND ENCODING SECTION
HIGH BAND COMPONENT ENCODED INFORMATION
LOW BAND COMPONENT ENCODED INFORMATION
207
ENCODED INFORMATION INTEGRATION SECTION
ENCODED INFORMATION

FIG.2

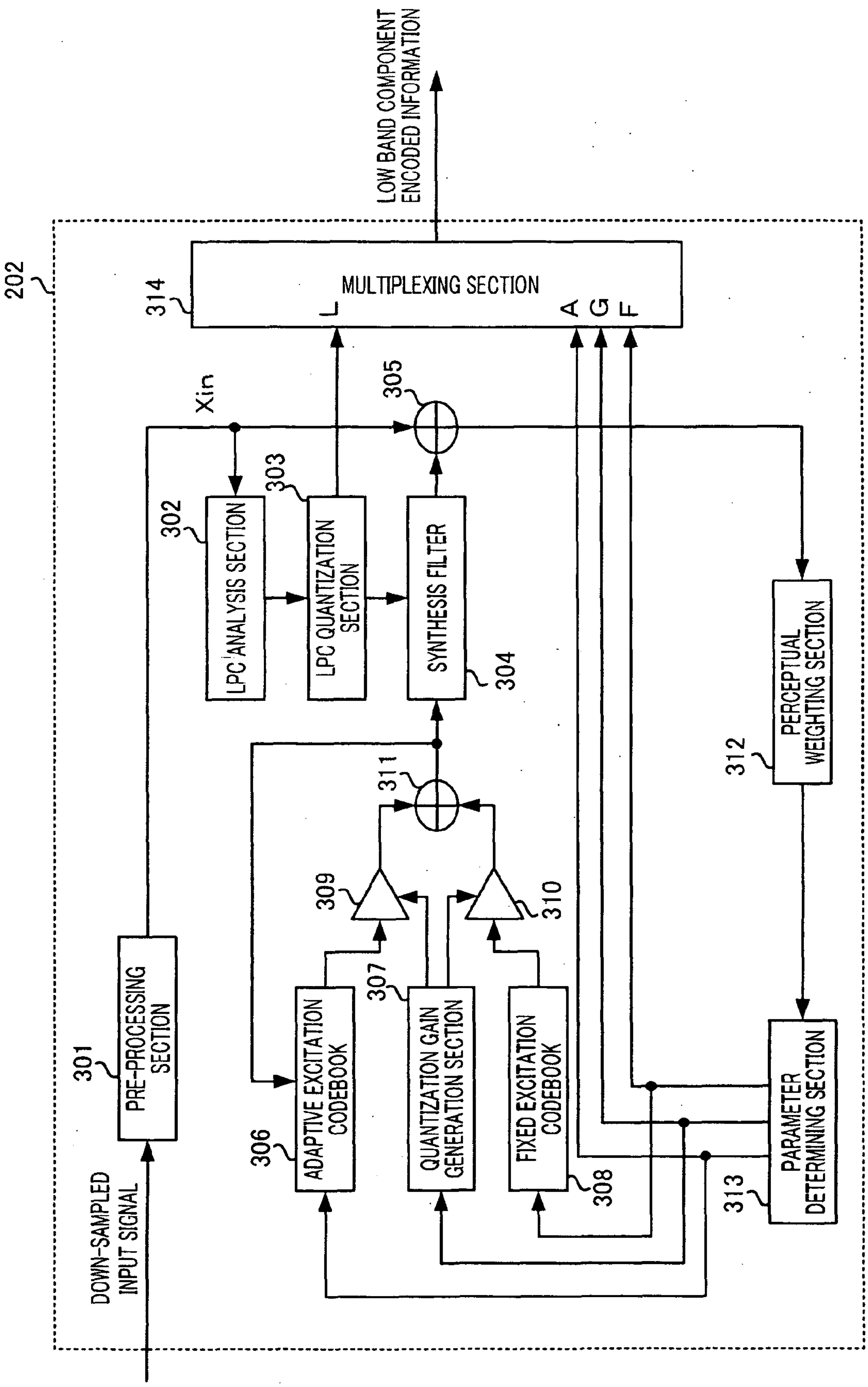
LOW BAND COMPONENT ENCODED INFORMATION
202
314
MULTIPLEXING SECTION
L
A
G
F
Xin
305
302
LPC ANALYSIS SECTION
303
LPC QUANTIZATION SECTION
SYNTHESIS FILTER
304
311
309
310
312
PERCEPTUAL WEIGHTING SECTION
301
PRE-PROCESSING SECTION
DOWN-SAMPLED INPUT SIGNAL
306
ADAPTIVE EXCITATION CODEBOOK
307
QUANTIZATION GAIN GENERATION SECTION
FIXED EXCITATION CODEBOOK
308
313
PARAMETER DETERMINING SECTION

FIG.3

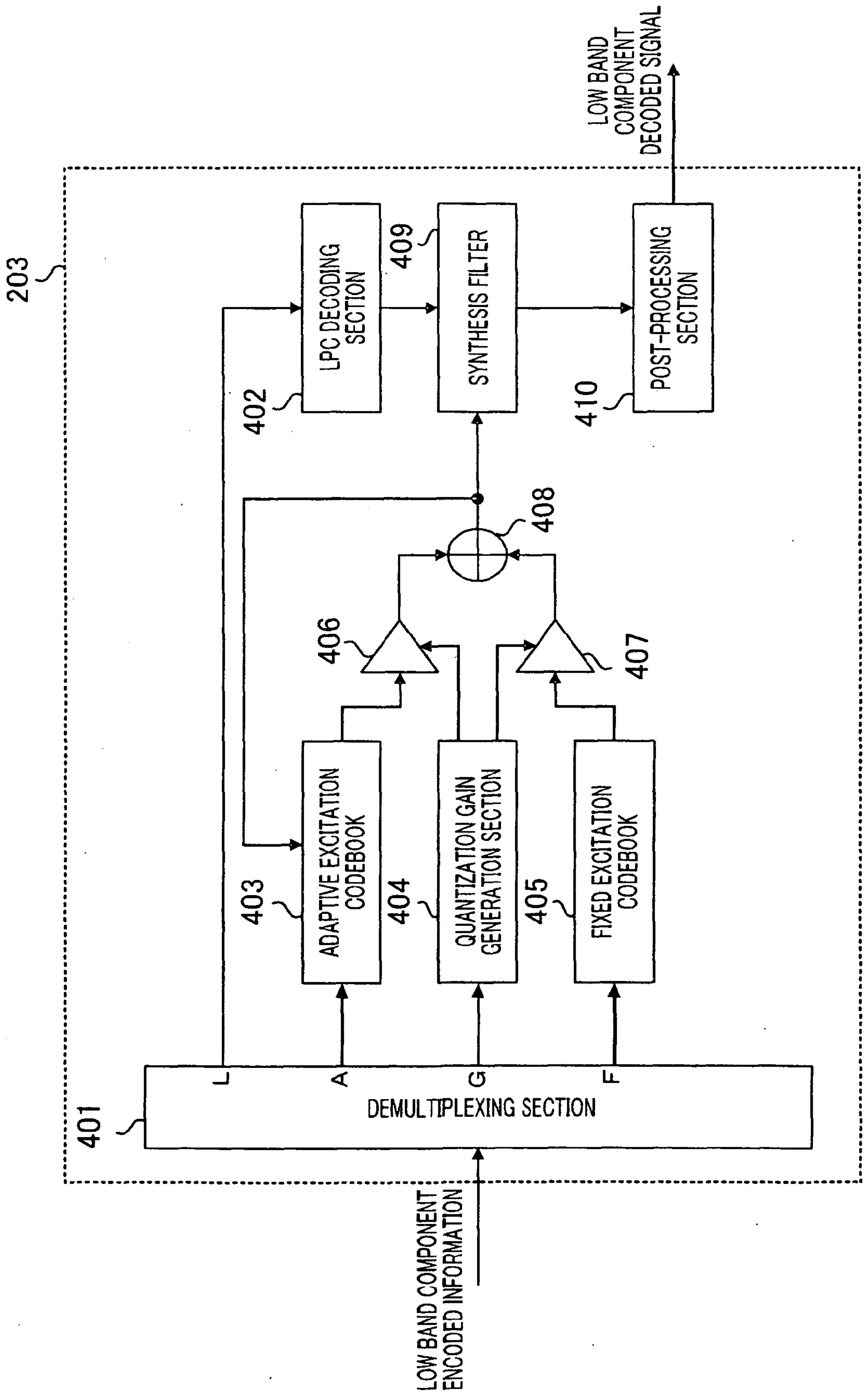
203
LOW BAND COMPONENT DECODED SIGNAL
402
LPC DECODING SECTION
409
SYNTHESIS FILTER
410
POST-PROCESSING SECTION
408
406
407
403
ADAPTIVE EXCITATION CODEBOOK
404
QUANTIZATION GAIN GENERATION SECTION
405
FIXED EXCITATION CODEBOOK
401
L
A
G
F
DEMULTIPLEXING SECTION
LOW BAND COMPONENT ENCODED INFORMATION

FIG.4

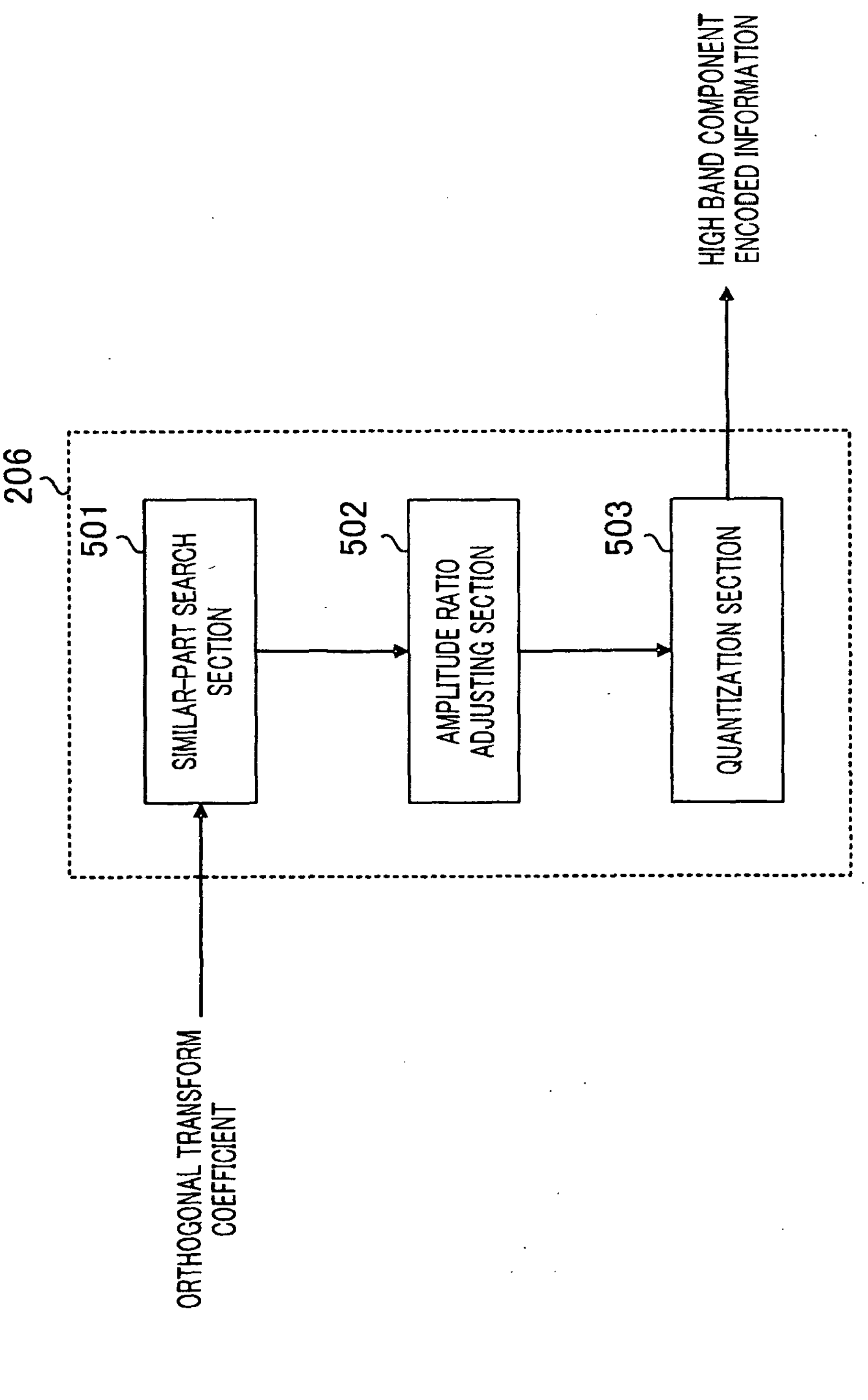
206
501
SIMILAR-PART SEARCH SECTION
502
AMPLITUDE RATIO ADJUSTING SECTION
503
QUANTIZATION SECTION
ORTHOGONAL TRANSFORM COEFFICIENT
HIGH BAND COMPONENT ENCODED INFORMATION

FIG.5

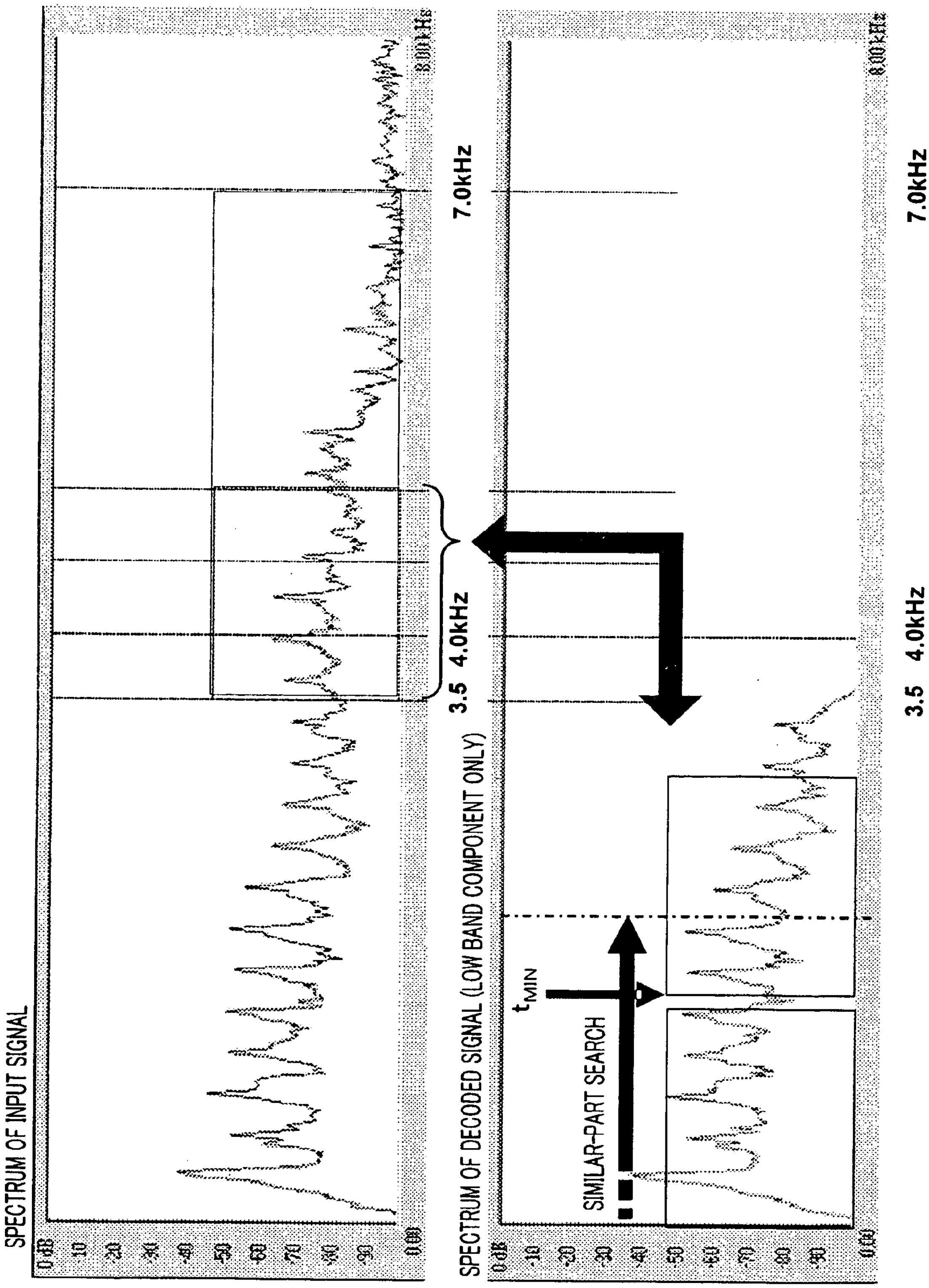
SPECTRUM OF INPUT SIGNAL
0 dB
-10
-20
-30
-40
-50
-60
-70
-80
-90
0.00
8.00kHz
3.5 4.0kHz
7.0kHz
SPECTRUM OF DECODED SIGNAL (LOW BAND COMPONENT ONLY)
SIMILAR-PART SEARCH
t_{MIN}
3.5 4.0kHz
7.0kHz

FIG.6A

FIG.6B

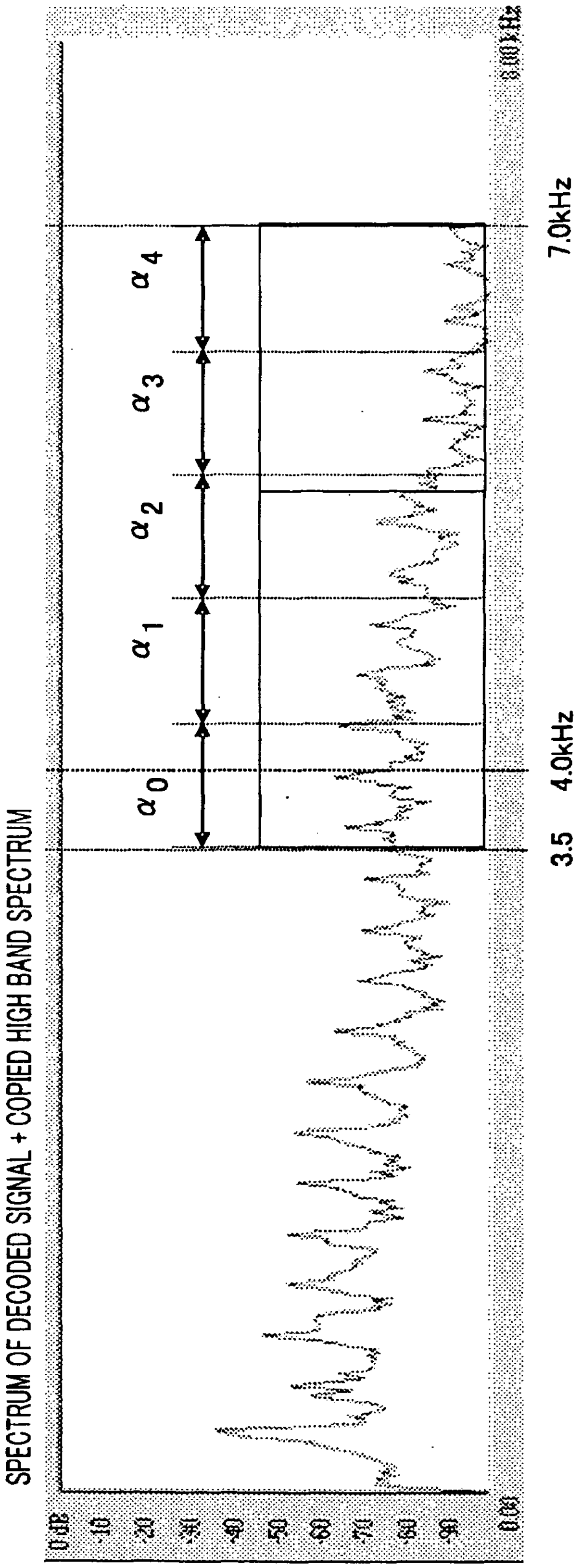
SPECTRUM OF DECODED SIGNAL + COPIED HIGH BAND SPECTRUM
α_0
α_1
α_2
α_3
α_4
3.5
4.0kHz
7.0kHz

FIG.7

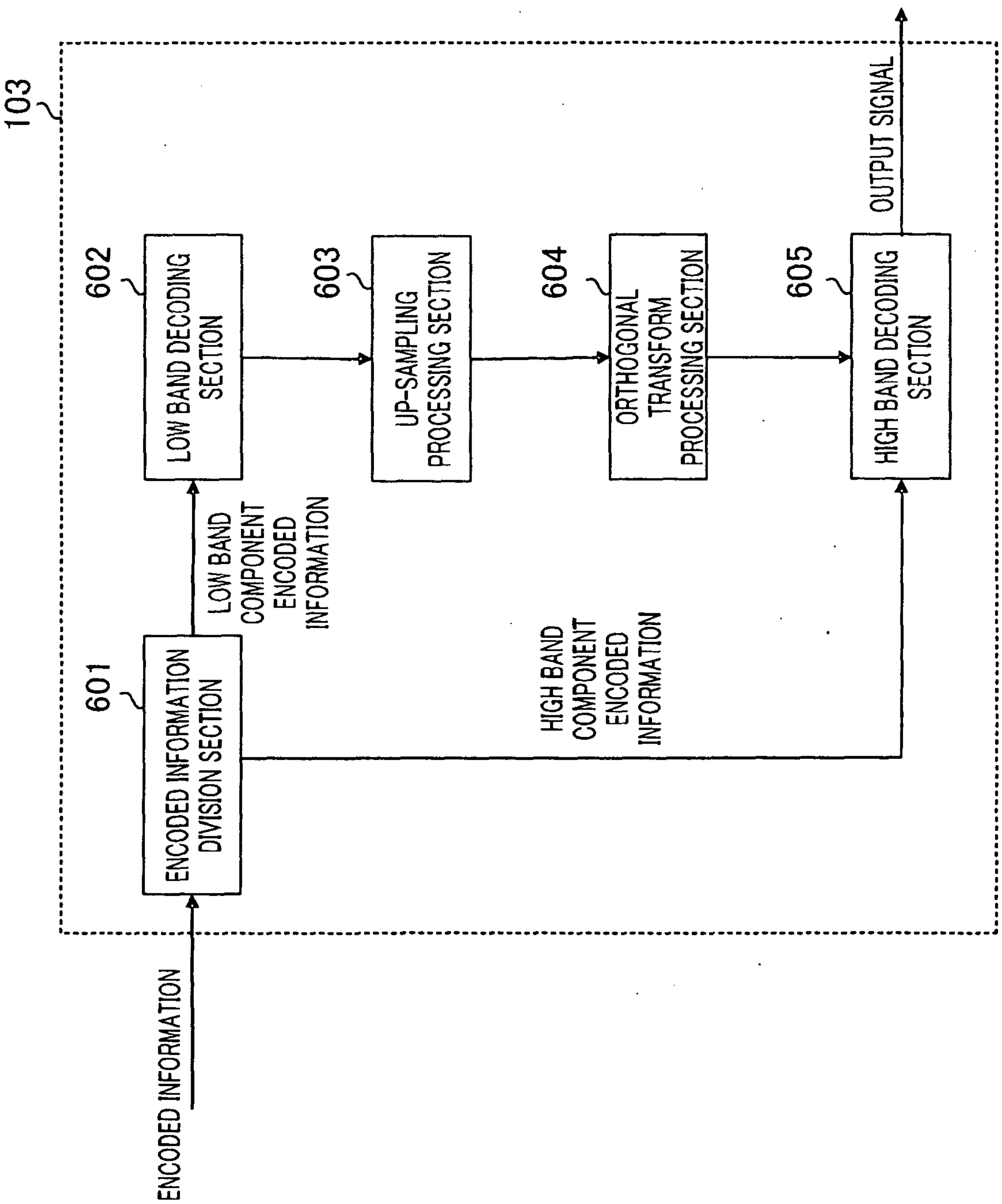
103
ENCODED INFORMATION
601
ENCODED INFORMATION DIVISION SECTION
LOW BAND COMPONENT ENCODED INFORMATION
602
LOW BAND DECODING SECTION
603
UP-SAMPLING PROCESSING SECTION
604
ORTHOGONAL TRANSFORM PROCESSING SECTION
HIGH BAND COMPONENT ENCODED INFORMATION
605
HIGH BAND DECODING SECTION
OUTPUT SIGNAL

FIG.8

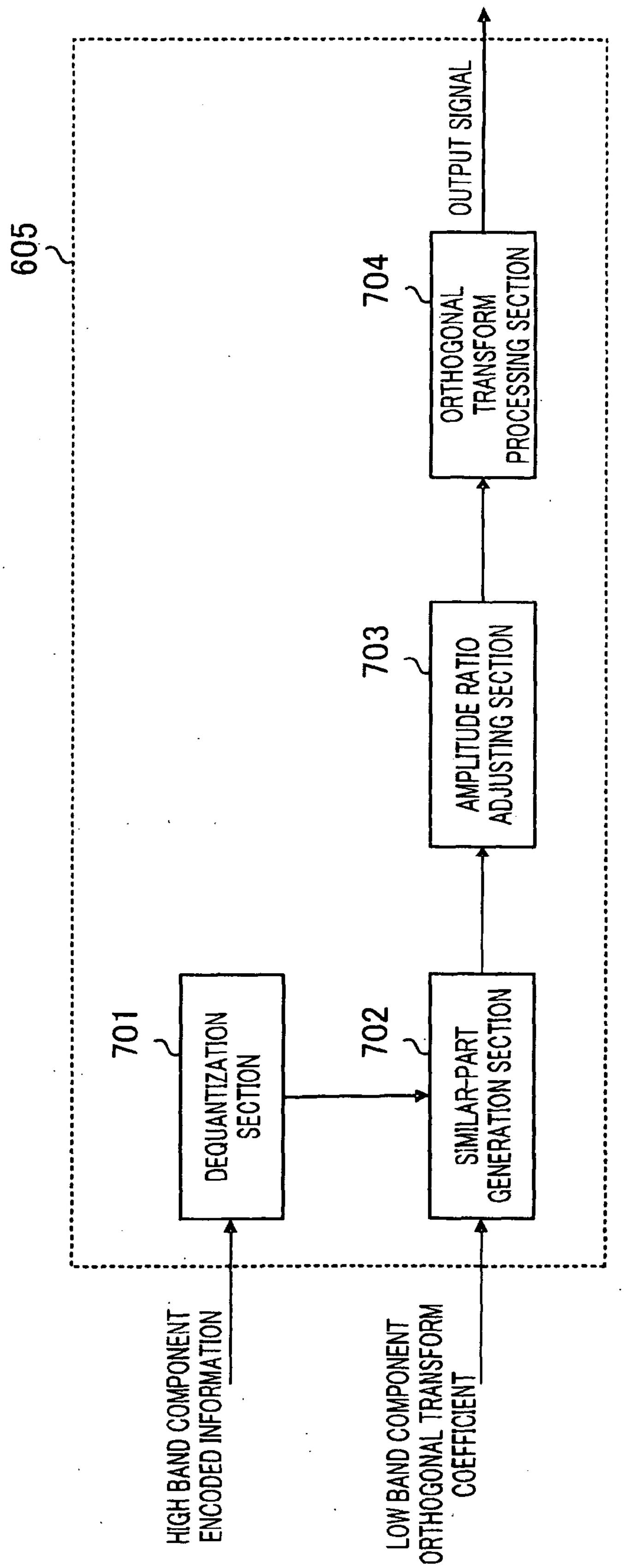
605
701
DEQUANTIZATION SECTION
702
SIMILAR-PART GENERATION SECTION
703
AMPLITUDE RATIO ADJUSTING SECTION
704
ORTHOGONAL TRANSFORM PROCESSING SECTION
OUTPUT SIGNAL
HIGH BAND COMPONENT ENCODED INFORMATION
LOW BAND COMPONENT ORTHOGONAL TRANSFORM COEFFICIENT

FIG.9

REFERENCES CITED IN THE DESCRIPTION

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- JP 2003140692 A **[0007]**
- JP 2004004530 A **[0007]**

Non-patent literature cited in the description

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- **GRILL B.** A bit rate scalable perceptual code for MPEG-4 audio. *AUDIO ENGINEERING SOCIETY. CONVENTION PREPRINT,* 26 September 1997, vol. XX, XX **[0007]**