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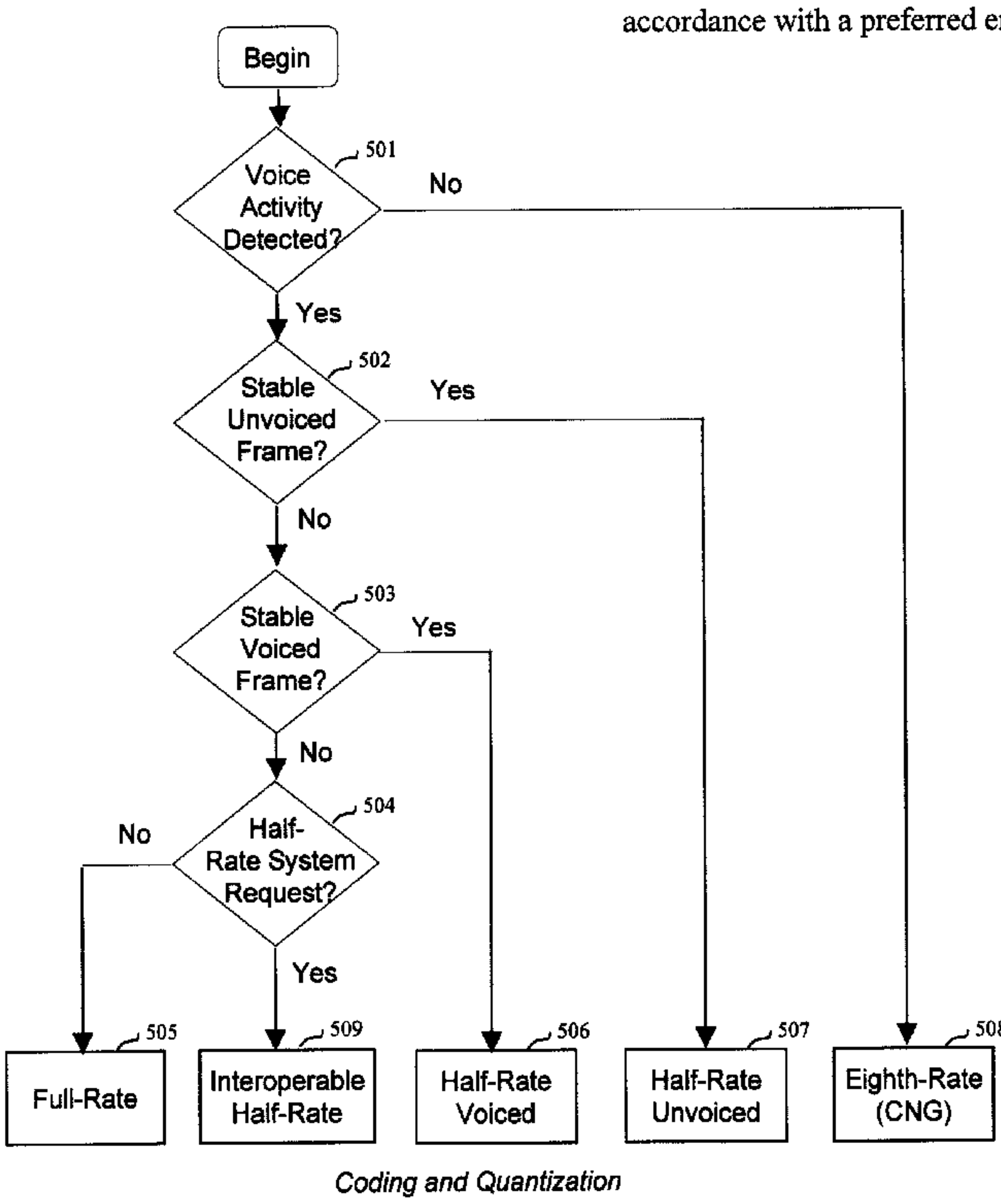
(54) Titre : METHODE ET DISPOSITIF DE SIGNALISATION ATTENUATION-RAFALE DE RESEAU INTELLIGENT EFFICACE ET EXPLOITATION MAXIMALE A DEMI-DEBIT DANS LE CODAGE DE LA PAROLE A LARGE BANDE A DEBIT BINAIRE VARIABLE POUR SYSTEMES AMRC SANS FIL

(54) Title: A METHOD AND DEVICE FOR EFFICIENT IN-BASED DIM-AND-BURST SIGNALING AND HALF-RATE MAX OPERATION IN VARIABLE BIT-RATE WIDEBAND SPEECH CODING FOR CDMA WIRELESS SYSTEMS

Functional block diagram of Figure 2 with including the new interoperable half-rate and its use within the rate determination logic in accordance with a preferred embodiment of the present invention.



A METHOD AND DEVICE FOR EFFICIENT IN-BAND DIM-AND-BURST SIGNALING AND HALF-RATE MAX OPERATION IN VARIABLE BIT-RATE WIDEBAND SPEECH CODING FOR CDMA WIRELESS SYSTEMS

BACKGROUND OF THE INVENTION

1. Field of the Invention

The present invention relates to an improved technique for digitally encoding a sound signal, in particular but not exclusively a speech signal, in view of transmitting and synthesizing this sound signal in a wireless CDMA system. In particular, the present invention relates to the design of variable bit-rate CELP-based coding capable of operating efficiently within the CDMA2000 system requirements such as in-band dim-and-burst signalling and half-rate max operation. Further, the present invention relates to the design of variable bit-rate CELP-based coding capable of operating efficiently across other systems such as IP-based or W-CDMA systems in a tandem-free operation setup.

2. Brief Description of the Prior Art

Demand for efficient digital narrowband and wideband speech coding techniques with a good trade-off between the subjective quality and bit rate is increasing in various application areas such as teleconferencing, multimedia, and wireless communications. Until recently, telephone bandwidth constrained into a range of 200–3400 Hz has mainly been used in speech coding applications. However, wideband speech applications provide increased intelligibility and naturalness in communication compared to the conventional telephone bandwidth. A bandwidth in the range 50–7000 Hz has been found sufficient for delivering a good quality giving an impression of face-to-face communication. For general audio signals, this bandwidth gives an acceptable subjective quality, but is still lower than the quality of FM radio or CD that operate on ranges of 20–16000 Hz and 20–20000 Hz, respectively.

A speech encoder converts a speech signal into a digital bitstream which is transmitted over a communication channel or stored in a storage medium. The speech signal is digitized, that is, sampled and quantized with usually 16-bits per sample. The speech encoder has the role of representing these digital samples with a smaller number of bits while maintaining a good subjective speech quality. The speech decoder or synthesizer operates on the transmitted or stored bit stream and converts it back to a sound signal.

Code-Excited Linear Prediction (CELP) coding is one of the best prior art techniques for achieving a good compromise between the subjective quality and bit rate. This coding technique is a basis of several speech coding standards both in wireless and wireline applications. In CELP coding, the sampled speech signal is processed in successive blocks of N samples usually called *frames*, where N is a predetermined number corresponding typically to 10–30 ms. A linear prediction (LP) filter is computed and transmitted every frame. The computation of the LP filter typically needs a *lookahead*, a 5–15 ms speech segment from the subsequent frame. The N-sample frame is divided into smaller blocks called *subframes*. Usually the number of subframes is three or four resulting in 4–10 ms subframes. In each subframe, an excitation signal is usually obtained from two components, the past excitation and the innovative, fixed-codebook excitation. The component formed from the past excitation is often referred to as the adaptive codebook or pitch excitation. The parameters characterizing the excitation signal are coded and transmitted to the decoder, where the reconstructed excitation signal is used as the input of the LP filter.

In wireless systems using code division multiple access (CDMA) technology, the use of source-controlled variable bit rate (VBR) speech coding significantly improves the system capacity. In source-controlled VBR coding, the codec operates at several bit rates, and a rate selection module is used to determine the bit rate used for encoding each speech frame based on the nature of the speech frame (e.g. voiced, unvoiced, transient, background noise). The goal is to attain the best speech quality at a given average bit rate, also referred to as average data rate (ADR). The codec can operate at different modes by tuning the rate selection module to attain different ADRs at the different modes where the codec performance is improved at increased ADRs. This enables the codec with a mechanism of trade-off between speech quality and system capacity. In CDMA systems (e.g. CDMA-one and CDMA2000), typically 4 bit rates are used and they are referred to as full-rate (FR), half-rate (HR), quarter-rate (QR), and eighth-rate (ER). In this system two rate sets are supported referred to as Rate Set I and Rate Set II. In Rate Set II, a variable-rate codec with rate selection mechanism operates at source-coding bit rates of 13.3 (FR), 6.2 (HR), 2.7 (QR), and 1.0 (ER) kbit/s, corresponding of gross bit rates of 14.4, 7.2, 3.6, and 1.8 kbit/s (with some bits added for error detection).

In CDMA systems, the system can impose the use of the half-rate instead of full-rate in some speech frames in order to send in-band signaling information (called dim-and-burst signaling). The use of half-rate as a maximum bit rate can be also imposed by the system during bad channel conditions (such as near the cell boundaries) in order to improve the codec robustness. This is referred to as half-rate max. Typically, in VBR coding, the half rate is used when the frame is stationary voiced or stationary unvoiced. Two codec structures are used for each type of signal (in unvoiced case a CELP model without the pitch codebook is used and in voiced case signal modification is used to enhance the periodicity and reduce the number of bits for the pitch indices). Full-rate is used for onsets, transient frames, and mixed voiced frames (a typical CELP model is usually used). When the rate-selection module chooses the frame to be encoded as a full-rate frame and the system imposes the half-rate frame the speech performance is degraded since the half-rate modes are not capable of efficiently encoding onsets and transient signals.

A wideband codec known as adaptive multi-rate wideband (AMR-WB) speech codec was recently selected by the ITU-T (International Telecommunications Union – Telecommunication Standardization Sector) for several wideband speech telephony and services and by 3GPP (third generation partnership project) for GSM and W-CDMA third generation wireless systems. AMR-WB codec consists of nine bit rates in the range from 6.6 to 23.85 kbit/s. Designing an AMR-WB-based source controlled VBR codec for CDMA2000 system has the advantage of enabling the interoperation between CDMA2000 and other systems using the AMR-WB codec. The AMR-WB bit rate of 12.65 kbit/s is the closest rate that can fit in the 13.3 kbit/s full-rate of Rate Set II. This rate can be used as the common rate between a CDMA2000 wideband VBR codec and AMR-WB which will enable the interoperability without the need for transcoding (which degrades the speech quality). A half-rate at 6.2 kbit/s has to be added to the CDMA2000 VBR wideband solution to enable the efficient operation in the Rate Set II framework. The codec then can operate in few CDMA2000-specific modes but it will have a mode that enables interoperability with systems using the AMR-WB codec. However, in a cross-system tandem free operation call between CDMA2000 and another system using AMR-WB, a case will arise where the CDAM2000 system with force the use of the half-rate as explained earlier (such as in dim-and-burst signaling). Since the AMR-WB codec doesn't recognize the 6.2 kbit/s half-rate of the CDMA2000 wideband codec, then forced half-rate frames will be interpreted as erased frames. This will adversely affect the performance of the connection.

OBJECTIVE OF THE INVENTION

An objective of the present invention is therefore to provide novel techniques to improve the performance of variable bit rate speech codecs operating in CDMA wireless systems in situations where the half-rate is imposed by the system. Another objective is to improve the performance in case of a cross-system tandem free operation between CDMA2000 and other systems using AMR-WB codec when the CDMA2000 system forces the use of the half-rate.

BRIEF DESCRIPTION OF THE DRAWINGS

Figure 1 is a schematic block diagram of a speech communication system illustrating the use of speech encoding and decoding devices in accordance with the present invention;

Figure 2 is a functional block diagram of a variable bit rate codec with rate determination logic in accordance with a preferred embodiment of the present invention;

Figure 3 is a functional block diagram of Figure 2 with including the new interoperable half-rate and its use within the rate determination logic in accordance with a preferred embodiment of the present invention;

Figure 4 is a functional block diagram similar to Figure 3 showing an alternative implementation of the interoperable half-rate in accordance with a preferred embodiment of the present invention; and

Figure 5 is An example configuration for the proposed dim and burst signaling method in the interoperable mode of VBR-WB when involved in a 3GPP $\leftrightarrow$ CDMA2000 mobile to mobile call.

DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENT

Figure 1 illustrates a speech communication system depicting the use of speech encoding and decoding in accordance with the present invention. The speech communication system supports transmission and reproduction of a speech signal across a communication channel 905. Although it may comprise for example a wire, optical or fiber link, the communication channel 905 typically comprises at least in part a radio frequency link. The radio frequency link often supports multiple, simultaneous speech communications requiring shared bandwidth resources such as may be found with cellular telephony embodiments. Although not shown, the communication channel may be replaced by a storage device in a single device embodiment of the communication system that records and stores the encoded speech signal for later playback.

A microphone 901 produces an analog speech signal that is conducted to an analog to digital (A/D) converter 902 for converting it into a digital form. A speech encoder 903 encodes the digitized speech signal producing a set of parameters that are coded into a binary form and delivered to a channel encoder 904. The optional channel encoder adds redundancy to the binary representation of the coding parameters before transmitting them over the communication channel 905. In the receiver side, a channel decoder 906 utilizes the said redundant information in the received bitstream to detect and correct channel errors occurred in the transmission. A speech decoder 907 converts the bitstream received from the channel decoder back to a set of coding parameters for creating a synthesized speech signal. The synthesized speech signal reconstructed at the speech decoder is converted to an analog form in a digital to analog (D/A) converter 908 and played back in a loudspeaker unit 909.

Source-controlled Variable Bit Rate Speech Coding

Figure 2 depicts a preferred embodiment of a variable bit rate coding configuration including a rate determination logic that controls four coding bit rates. In this particular embodiment, the bit rate set comprises a dedicated codec type for non-active speech frames (block 508), unvoiced speech frames (block 507), stable voiced frames (block 506), and other types of frames (block 505).

The rate determination logic is based on signal classification done in three steps in logic blocks 501, 502, and 503, whose operation is well known to the experts on prior art. First, a voice activity detector (VAD), block 501, discriminates between active and inactive speech frames. If an inactive speech frame is detected (background noise signal) then the classification chain ends and the frame is encoded in module 508 as an eighth-rate frame with comfort noise generation (CNG) at the decoder (1.0 kbit/s according to CDMA2000 Rate Set II). If an active speech frame is detected, the frame is subjected to a second classifier 502 dedicated to making a voicing decision. If the classifier 502 classifies the frame as unvoiced speech signal, the classification chain ends, and the frame is encoded in module 507 with a half rate optimized for unvoiced signals (6.2 kbit/s according to CDMA2000 Rate Set II). Otherwise, the speech frame is passed through to the "stable voiced" classification module 503. If the frame is classified as stable voiced frame, then the frame is encoded in module 506 with a half rate optimized for stable voiced signals (6.2 kbit/s according to CDMA2000 Rate Set II). Otherwise, the frame is likely to contain a nonstationary speech segment such as a voiced onset or rapidly evolving voiced speech signal. These frames typically require a high bit rate for sustaining good subjective quality. Thus, in this case, the speech frame is encoded in module 505 as a full-rate frame (13.3 kbit/s according to CDMA2000 Rate Set II).

The classification modules 501, 502, and 503 are well-known to people skilled in the art and will not be detailed in this invention. According to a preferred embodiment of the present invention, the coding modules at different bit rates in modules 505, 506, and 507 are based on code-excited linear prediction (CELP) coding techniques well known in prior art. In this preferred embodiment, the bit rates are set according of Rate Set II of the CDMA2000 system described above.

In this preferred embodiment, the disclosed invention is explained based on a wideband speech codec that has been standardized by the International Telecommunications Union (ITU) as Recommendation G.722.2 and known as the AMR-WB codec (Adaptive Multi-Rate Wideband codec) [1]. This codec has also been selected by the third generation partnership project (3GPP) for wideband telephony in third generation wireless systems [2]. AMR-WB can operate at 9 bit rates from 6.6 to 23.85 kbit/s. Here, the bit rate at 12.65 kbit/s is used as the full-rate to illustrate the present invention.

In full-rate, the AMR-WB standard codec at 12.65 kbit/s is used with the bit allocation given in Table 1. The use of the 12.65 kbit/s rate of the AMR-WB codec enables the design of a variable bit rate codec for the CDMA2000 system capable of interoperating with other systems using the AMR-WB codec standard. Extra 13 bits are added to fit in the 13.3 kbit/s full-rate of CDMA2000 Rate Set II. These bits are used to improve the codec robustness in case of erased frames. More details about the AMR-WB codec can be found in reference [1]. The codec is based on the algebraic code-excited linear prediction (ACELP) model optimized for wideband signals. It operates on 20 ms speech frames with a sampling frequency of 16 kHz. The LP filter parameters are encoded once per frame using 46 bits. Then the frame is divided into four subframes where adaptive and fixed codebook indices and gains are encoded once per frame. The fixed codebook is constructed using an algebraic codebook structure where the 64 positions in a subframe are divided into 4 tracks of interleaved positions and where 2 signed pulses are placed in each track. The two pulses per track are encoded using 9 bits giving a total of 36 bits per subframe.

Table 1. Bit allocation of the 13.3 kbit/s full-rate in accordance with the AMR-WB standard at 12.65 kbit/s (20 ms frames comprising four subframes).

VAD flag	1
LP Parameters	46
Pitch Delay	30 = 9 + 6 + 9 + 6
Pitch Filtering	4 = 1 + 1 + 1 + 1
Gains	28 = 7 + 7 + 7 + 7
Algebraic Codebook	144 = 36 + 36 + 36 + 36
Extra bits	13

In case of stable voiced frames, the half-rate voiced coding module 506 is used. The half-rate voiced bit allocation is given in Table 2. Since the frames to be coded in this mode are characteristically very periodic, a substantially lower bit rate suffices for sustaining good subjective quality compared for instance to transition frames. Signal modification is used which allows efficient coding of the delay information using only nine bits per 20-ms frame saving a considerable proportion of the bit budget for other parameters. In signal modification, the signal is forced to follow a certain pitch contour that can be transmitted with 9 bits per frame. Good performance of long term prediction allows to use only 13 bits per 5-ms subframe for the fixed-codebook excitation without sacrificing the subjective speech quality. The fixed-codebook is an algebraic codebook comprises one track with two pulses, both having 64 possible positions. One bit is used to indicate that the frame is half rate voiced.

Table 2. Bit allocation of the half-rate voiced at 6.2 kbit/s for a 20-ms frame comprising four subframes.

LP Parameters	34
Pitch Delay	9
Pitch Filtering	4 = 1 + 1 + 1 + 1
Gains	24 = 6 + 6 + 6 + 6
Algebraic Codebook	52 = 13 + 13 + 13 + 13
Mode Bit	1

In case of unvoiced frames, the adaptive codebook (or pitch codebook) is not used. A 13-bit Gaussian codebook is used in each subframe where the codebook gain is encoded with 6 bits per subframe. 2 bits are used for the half-rate mode: the first bit to indicate that the half rate is not stable voiced and the second bit to indicate it is stable unvoiced and not interoperable half rate (the interoperable half rate will be explained in the next section

Table 3. Bit allocation of the half-rate unvoiced at 6.2 kbit/s
for a 20-ms frame comprising four subframes.

[redacted]	
LP Parameters	46
Gains	24 = 6 + 6 + 6 + 6
Gaussian Codebook	52 = 13 + 13 + 13 + 13
Mode Bit	2
[redacted]	

The eighth-rate is used to encode inactive speech frames (silence or background noise). In this case only the LP filter parameters are encoded with 14 bits per frame and a gain is encoded with 6 bits per frame. These parameters are used for comfort noise generation (CNG) at the decoder.

Table 4. Bit allocation of the eighth-rate at 1.0 kbit/s
for a 20-ms frame.

[redacted]	
LP Parameters	14
Gain	6
[redacted]	

System-imposed half-rate operation

In CDMA systems, the system can impose the use of the half-rate instead of full-rate in some speech frames in order to send in-band signaling information. This referred to as dim-and-burst signaling. The use of half-rate as a maximum bit rate can be also imposed by the system during bad channel conditions (such as near the cell boundaries) in order to improve the codec robustness. This is referred to as half-rate max. In the VBR coding configuration described above, the half rate is used when the frame is stationary voiced or stationary unvoiced. Full-rate is used for onset, transient frames, and mixed voiced When the rate-selection module chooses the frame to be encoded as a full-rate frame and the system imposes the half-rate frame the speech performance is degraded since the half-rate modes are not capable of efficiently encoding onsets and transient signals.

Further, in a cross-system tandem free operation call between CDMA2000 using the VBR Rate Set II solution based on AMR-WB and another system using the standard AMR-WB, a case will arise where the CDMA2000 system will force the use of the half-rate as explained earlier (such as in dim-and-burst signaling). Since the AMR-WB codec doesn't recognize the 6.2 kbit/s half-rate of the CDMA2000 wideband codec, then forced half-rate frames will be interpreted as erased frames. This will affect the performance of the connection.

In this invention, a novel technique is disclosed which improves the performance of variable bit rate speech codecs operating in CDMA wireless systems in situations where the half-rate is imposed by the system. Futher, the disclosed technique improves the performance in case of a cross-system tandem free operation between CDMA2000 and other systems using AMR-WB codec when the CDMA2000 system forces the use of the half-rate.

In dim-and-burst signaling or half-rate max operation, when the system requests the use of half-rate while a full-rate has been used by the classification mechanism, this indicates that the frame is not unvoiced nor stable voiced and the frame is likely to contain a nonstationary speech segment such as a voiced onset or rapidly evolving voiced speech signal. Thus the use of half-rate optimized for unvoiced or stable voiced signals will degrade the speech performance. A new half-rate mode is needed in this case, however, there are not enough bits to maintain good quality in case of such nonstationary signals. Thus designing a half-rate mode for these signals will not guarantee good performance and it will likely increase the memory requirements. In this invention, we disclose the use of a half-rate mode directly derived from the full rate mode by dropping the fixed codebook indices after the frame has been encoded as a full rate frame. At the decoder side, the fixed codebook indices can be randomly generated and the decoder will operate as if it is in full-rate. This half-rate mode is referred to as interoperable half-rate since both encoding and decoding are performed in full-rate. The bit allocation of the interoperable half-rate mode in accordance to a preferred embodiment of the present invention is given in Table 5. In this preferred embodiment, the full-rate is based on the AMR-WB standard at 12.65 kbit/s, and the half-rate is derived by dropping the 144 bits needed for the indices of the algebraic fixed codebook. 2 bits are added for the half-rate mode: the first bit to indicate that the half rate is not stable voiced and the second bit to indicate it is interoperable half rate and not unvoiced.

Table 5. Bit allocation of the interoperable half-rate at 6.2 kbit/s compared to the full-rate (20 ms frames comprising four subframes).

Half-rate mode	0	2
VAD flag	1	1
LP Parameters	46	46
Pitch Delay	30 = 9 + 6 + 9 + 6	30 = 9 + 6 + 9 + 6
Pitch Filtering	4 = 1 + 1 + 1 + 1	4 = 1 + 1 + 1 + 1
Gains	28 = 7 + 7 + 7 + 7	28 = 7 + 7 + 7 + 7
Algebraic Codebook	144 = 36 + 36 + 36 + 36	0
Extra bits	13	13

Figure 3 depicts the functional block diagram of Figure 2 by adding the new interoperable half-rate mode and the it shows its use withing the rate determination logic in accordance with a preferred embodiment of the present invention. At the end of the rate determination chain, module 504 verifies if a half-rate system request is present. If the rate determination logic indicates that the frame is active speech frame, and it is not unvoiced nor stable voiced, but the system requests a half-rate operation, then the interoperable half-rate mode is used and the frame is encoded in module 509 as a full-rate frame then the indices of the fixed codebook are dropped in order to obtain a half-rate frame (6.2 kbit/s according to CDMA2000 Rate Set II). Otherwise (no half-rate system request is present) the speech frame is encoded in module 505 as a full-rate frame (13.3 kbit/s according to CDMA2000 Rate Set II).

Figure 4 shows an alternative approach to implement the interoperable half-rate operation. Here, the rate determination logic and variable rate coding is initially the same as in Figure 2. However, after a full-rate frame has been encoded, a test is performed to verify if the system requests a half-rate operation. If this is the case then the fixed codebook indices are dropped in order to obtain an interoperable half-rate frame. Note that in this preferred embodiment, two bits are used for the half-rate mode (stable voiced, unvoiced, or interoperable). Thus, the two bits indicating a half-rate interoperable mode are added after the fixed codebook indices are dropped.

In this preferred embodiment, in interoperable half-rate operation at the encoder side, the encoder operates as a full rate encoder. The fixed codebook search is performed as usual and the determined fixed codebook excitation is used in updating the adaptive codebook content and filter memories for next frames according to AMR-WB standard at 12.65 kbit/s [1], [2]. Therefore, no random codebook indices are used within the encoder operation. This is evident in the implementation of Figure 4 where the half-rate system request is verified after the frame has been encoded in normal full-rate operation.

In interoperable half-rate operation at the decoder side, the indices of the fixed codebook are randomly generated. The decoder then operates as in full-rate operation. Other methods for generating the missed indices can be used. For instance, the indices can be obtained by copying parts of the received bitstream. Note that a mismatch can happen between the memories at the encoder and decoder side, since the fixed codebook excitation is not the same. However, such mismatch didn't seem to impact the performance especially in case of dim-and-burst signaling where typical rates are around 2%. The encoder and decoder operation can be synchronized if needed by using the same indices generated at the decoder to update the memory at the encoder side. Note that the index generation mechanism should be the same at the encoder and decoder and this is only possible within a CDMA2000 call. This approach can be incorporated in the implementation of Figure 3.

The performance of the proposed approach in dim-and-burst operation is almost transparent compared to the case where there is no half-rate system request. In lots of cases, the rate determination logic already determines the frame to be encoded with either quarter rate, half-rate voiced, or half-rate unvoiced. In such a case, the system request is neglected since it is already accommodated by the encoder and the type of signal in the frame is suitable for encoding at a half-rate or a lower rate. The interoperable half-rate is used only when the rate determination logic chooses a full-rate frame and the system requests half-rate operation. With typical dim-and-burst signaling rates (less than 2%) the actual percentage of frames classified as full rate and forced to operate in half-rate is much lower. In half-rate max operation, the use of interoperable half-rate is more frequent, however, it is much better than using either half-rate voiced or half-rate unvoiced in case of nonstationary frames.

It should be noted that the classification logic is adaptive with a mode of operation. Therefore in order to improve the performance, in the half-rate-max mode and dim-and-burst signaling, the logic can be made more relaxed for using the specific half-rate codecs (the half-rate voiced and unvoiced are used relatively more often than in normal operation). This is a sort of extension to the multi-mode operation, where the logic is more relaxed modes with lower average data rates.

Tandem free operation between CDMA2000 system and other systems using the AMR-WB standard

As mentioned earlier, designing a variable bit rate wideband (VBR-WB) codec for the CDMA2000 system based on the AMR-WB codec has the advantage of enabling tandem free operation (TFO) between the CDMA2000 system and other systems using the AMR-WB standard (such as the mobile GSM system or W-CDMA third generation wireless system). However, in a cross-system tandem free operation call between CDMA2000 and another system using AMR-WB, a case will arise where the CDAM2000 system with force the use of the half-rate as explained earlier (such as in dim-and-burst signaling). Since the AMR-WB codec doesn't recognize the 6.2 kbit/s half-rate of the CDMA2000 wideband codec, then forced half-rate frames will be interpreted as erased frames. This will affect the performance of the connection. The use of the interoperable half-rate mode disclosed earlier will significantly improve the performance since this mode can interoperate with the 12.65 kbit/s rate of the AMR-WB standard.

As disclosed above, the interoperable half-rate is basically a pseudo full-rate, where the codec operates as if it is in the full-rate mode. The difference is that the algebraic codebook indices are dropped at the end and are not transmitted. At the decoder side, the indices are randomly generated and then the decoder operates as if it is in a full-rate mode.

Figure 5 illustrates a TFO configuration demonstrating the use of the interoperable half-rate mode during in-band transmission of signalling information (i.e., dim and burst condition) in CDMA2000 system side. In this figure, the other side is a system using the AMR-WB standard and a 3GPP wireless system is given as an example.

In the link with the direction from CDMA2000 to 3GPP, when the multiplex sub-layer indicates a request for half-rate mode, the VBR-WB codec will operate in the interoperable half rate (I-HR) described earlier. At the system interface, when an I-HR frame is received, randomly generated algebraic codebook indices are added to the bit stream to output a 12.65 kbit/s rate. The decoder at the 3GPP side will interpret it as an ordinary 12.65 kbit/s frame.

In the other direction, that is in a link from 3GPP to CDMA2000, if at the system interface a half-rate request is received, then the algebraic codebook indices are dropped and two bits indicating the I-HR frame type are added. The decoder at the CDMA2000 side will operate as an I-HR frame type, which is part of the VBR-WB solution.

This proposal requires a minimal logic at the system interface and it significantly improves the performance over forcing dim-and-burst frames as blank-and-burst frames (erased frames).

Of course, many other modifications and variations are possible to the disclosed invention. In view of the above detailed description of the present invention and associated drawings, such other modifications and variations will now become apparent to those skilled in the art. It should also be apparent that such other variations may be effected without departing from the spirit and scope of the present invention. As an example, the fixed codebook indices are dropped in order to obtain an interoperable half-rate frame, however, other bits with less bit error sensitivity can be dropped for this purpose.

REFERENCES

[1] ITU-T Recommendation G.722.2 "Wideband coding of speech at around 16 kbit/s using Adaptive Multi-Rate Wideband (AMR-WB)", Geneva, 2002.

[2] 3GPP TS 26.190, "AMR Wideband Speech Codec: Transcoding Functions," *3GPP Technical Specification.*

Appendix: Overview of the AMR-WB codec

Overview of AMR-WB encoder

The sampled speech signal is encoded on a block by block basis by the encoding device 100 of Figure 6 which is broken down into eleven modules numbered from 101 to 111.

The input speech is processed into the above mentioned L-sample blocks called frames.

Referring to Figure 6, the sampled input speech signal 114 is down-sampled in a down-sampling module 101. The signal is down-sampled from 16 kHz down to 12.8 kHz, using techniques well known to those of ordinary skill in the art. Down-sampling increases the coding efficiency, since a smaller frequency bandwidth is encoded. This also reduces the algorithmic complexity since the number of samples in a frame is decreased. After down-sampling, the 320-sample frame of 20 ms is reduced to 256-sample frame (down-sampling ratio of 4/5).

The input frame is then supplied to the optional pre-processing block 102. Pre-processing block 102 may consist of a high-pass filter with a 50 Hz cut-off frequency. High-pass filter 102 removes the unwanted sound components below 50 Hz.

The down-sampled pre-processed signal is denoted by $s_p(n)$, n=0, 1, 2, …,L-1, where L is the length of the frame (256 at a sampling frequency of 12.8 kHz). In a preferred embodiment of the preemphasis filter 103, the signal $s_p(n)$ is preemphasized using a filter having the following transfer function:

$$P(z) = 1 - \mu z^{-1}$$

where μ is a preemphasis factor with a value located between 0 and 1 (a typical value is μ = 0.7). The function of the preemphasis filter 103 is to enhance the high frequency contents of the input signal. It also reduces the dynamic range of the input speech signal, which renders it more suitable for fixed-point implementation. Preemphasis also plays an important role in achieving a proper overall perceptual weighting of the quantization error, which contributes to improved sound quality. This will be explained in more detail herein below.

The output of the preemphasis filter 103 is denoted $s(n)$. This signal is used for performing LP analysis in calculator module 104. LP analysis is a technique well known to those of ordinary skill in the art. In this preferred embodiment, the autocorrelation approach is used. In the autocorrelation approach, the signal $s(n)$ is first windowed using with typically a Hamming window having usually a length of the order of 30-40 ms. The autocorrelations are computed from the windowed signal, and Levinson-Durbin recursion is used to compute LP filter coefficients, a_i, where i=1,...,p, and where p is the LP order, which is typically 16 in wideband coding. The parameters a_i are the coefficients of the transfer function of the LP filter, which is given by the following relation:

$$A(z) = 1+\sum_{i=1}^{p} a_i\, z^{-1}$$

LP analysis is performed in calculator module 104, which also performs the quantization and interpolation of the LP filter coefficients. The LP filter coefficients are first transformed into another equivalent domain more suitable for quantization and interpolation purposes. The line spectral pair (LSP) and immitance spectral pair (ISP) domains are two domains in which quantization and interpolation can be efficiently performed. The 16 LP filter coefficients, a_i, can be quantized in the order of 30 to 50 bits using split or multi-stage quantization, or a combination thereof. The purpose of the interpolation is to enable updating the LP filter coefficients every subframe while transmitting them once every frame, which improves the encoder performance without increasing the bit rate. Quantization and interpolation of the LP filter coefficients is believed to be otherwise well known to those of ordinary skill in the art and, accordingly, will not be further described in the present specification.

The following paragraphs will describe the rest of the coding operations performed on a subframe basis. In this embodiment, the input frame is divided into 4 subframes of 5 ms (64 samples at 12.8 kHz sampling). In the following description, the filter $A(z)$ denotes the unquantized interpolated LP filter of the subframe, and the filter $\hat{A}(z)$ denotes the quantized interpolated LP filter of the subframe.

In analysis-by-synthesis encoders, the optimum pitch and innovation parameters are searched by minimizing the mean squared error between the input speech and synthesized speech in a perceptually weighted domain. The weighted signal $s_w(n)$ is computed in a perceptual weighting filter 105. A perceptual weighting filter 105 with fixed denominator, suited for wideband signals, is used. An example of transfer function for the perceptual weighting filter 104 is given by the following relation:

$$W(z) = A(z/\gamma_1)/(1-\gamma_2 z^{-1}) \qquad \text{where} \qquad 0 < \gamma_2 < \gamma_1 \leq 1$$

In order to simplify the pitch analysis, an open-loop pitch lag T_{OL} is first estimated in the open-loop pitch search module 106 using the weighted speech signal $s_w(n)$. Then the closed-loop pitch analysis, which is performed in closed-loop pitch search module 107 on a subframe basis, is restricted around the open-loop pitch lag T_{OL} which significantly reduces the search complexity of the LTP parameters T and b (pitch lag and pitch gain). Open-loop pitch analysis is usually performed in module 106 once every 10 ms (two subframes) using techniques well known to those of ordinary skill in the art.

The target vector $\boldsymbol{x}$ for LTP (Long Term Prediction) analysis is first computed. This is usually done by subtracting the zero-input response $\mathbf{s}_0$ of weighted synthesis filter $W(z)/\hat{A}(z)$ from the weighted speech signal $s_w(n)$. This zero-input response $\mathbf{s}_0$ is calculated by a zero-input response calculator 108. This operation is well known to those of ordinary skill in the art and, accordingly, will not be further described.

A N-dimensional impulse response vector $\mathbf{h}$ of the weighted synthesis filter $W(z)/\hat{A}(z)$ is computed in the impulse response generator 109 using the LP filter coefficients $A(z)$ and $\hat{A}(z)$ from module 104. Again, this operation is well known to those of ordinary skill in the art and, accordingly, will not be further described in the present specification.

The closed-loop pitch (or pitch codebook) parameters b, T and j are computed in the closed-loop pitch search module 107, which uses the target vector x, the impulse response vector **h** and the open-loop pitch lag T_{OL} as inputs.

The pitch search consists of finding the best pitch lag T and gain b that minimize the mean squared weighted error E between the target vector **x** and the scaled filtered past excitation.

In the preferred embodiment of the present invention, the pitch (pitch codebook) search is composed of three stages.

In the first stage, an open-loop pitch lag T_{OL} is estimated in open-loop pitch search module 106 in response to the weighted speech signal $s_w(n)$. As indicated in the foregoing description, this open-loop pitch analysis is usually performed once every 10 ms (two subframes) using techniques well known to those of ordinary skill in the art.

In the second stage, the search criterion C is searched in the closed-loop pitch search module 107 for integer pitch lags around the estimated open-loop pitch lag T_{OL} (usually ±5), which significantly simplifies the search procedure. A simple procedure is used for updating the filtered codevector $\mathbf{y}_T$ without the need to compute the convolution for every pitch lag.

Once an optimum integer pitch lag is found in the second stage, a third stage of the search (module 107) tests the fractions around that optimum integer pitch lag (AMR-WB standard uses ¼ and ½ subsample resolution).

In wideband signals, the harmonic structure exists only up to a certain frequency, depending on the speech segment. Thus, in order to achieve efficient representation of the pitch contribution in voiced segments of wideband speech, the pitch prediction filter needs to have the flexibility of varying the amount of periodicity over the wideband spectrum. This is achieved by adding a potential frequency shaping filters after the pitch predictor and select the filter that minimizes the mean-squared weighted error.

The pitch codebook index T is encoded and transmitted to multiplexer 112. The pitch gain b is quantized and transmitted to multiplexer 112. One extra bit is used to encode the index j of the selected frequency shaping filter in multiplexer 112.

Once the pitch, or LTP (Long Term Prediction) parameters b, T, and j are determined, the next step is to search for the optimum innovative excitation by means of search module 110 of Figure 6. First, the target vector $\mathbf{x}$ is updated by subtracting the LTP contribution:

$$\mathbf{x}_2 = \mathbf{x} - b\mathbf{y}_T$$

where b is the pitch gain and $\mathbf{y}_T$ is the filtered pitch codebook vector (the past excitation at delay T filtered with the selected low pass filter and convolved with the inpulse response $\mathbf{h}$).

The search procedure in CELP is performed by finding the optimum excitation codevector $\mathbf{c}_k$ and gain g which minimize the mean-squared error between the target vector and the scaled filtered codevector.

It is worth noting that the used innovation codebook is a dynamic codebook consisting of an algebraic codebook followed by an adaptive prefilter $F(z)$ which enhances special spectral components in order to improve the synthesis speech quality, according to US Patent 5,444,816. In the preferred embodiment of the present invention, the innovative codebook search is performed in module 110 by means of an algebraic codebook as described in US patents Nos: 5,444,816 (Adoul et al.) issued on August 22, 1995; 5,699,482 granted to Adoul et al., on December 17, 1997; 5,754,976 granted to Adoul et al., on May 19, 1998; and 5,701,392 (Adoul et al.) dated December 23, 1997.

Overview of AMR-WB Decoder

The speech decoding device 200 of Figure 7 illustrates the various steps carried out between the digital input 222 (input stream to the demultiplexer 217) and the output sampled speech 223 (output of the adder 221).

Demultiplexer 217 extracts the synthesis model parameters from the binary information received from a digital input channel. From each received binary frame, the extracted parameters are:

- the short-term prediction parameters (STP) $\hat{A}(z)$ (once per frame);

- the long-term prediction (LTP) parameters T, b, and j (for each subframe); and

- the innovation codebook index k and gain g (for each subframe).

The current speech signal is synthesized based on these parameters as will be explained hereinbelow.

The innovative codebook 218 is responsive to the index k to produce the innovation codevector $\mathbf{c}_k$, which is scaled by the decoded gain factor g through an amplifier 224. In the preferred embodiment, an innovative codebook 218 as described in the above mentioned US patent numbers 5,444,816;

5,699,482; 5,754,976; and 5,701,392 is used to represent the innovative codevector $\mathbf{c}_k$.

The generated scaled codevector at the output of the amplifier 224 is processed through a frequency-dependent pitch enhancer 205.

Enhancing the periodicity of the excitation signal $\mathbf{u}$ improves the quality in case of voiced segments. The periodicity enhancement is achieved by filtering the innovative codevector $\mathbf{c}_k$ from the innovative (fixed) codebook through an innovation filter 205 ($F(z)$) whose frequency response emphasizes the higher frequencies more than lower frequencies. The coefficients of $F(z)$ are related to the amount of periodicity in the excitation signal $\mathbf{u}$.

An efficient way to derive the filter $F(z)$ coefficients used in a preferred embodiment, is to relate them to the amount of pitch contribution in the total excitation signal $\mathbf{u}$. This results in a frequency response depending on the subframe periodicity, where higher frequencies are more strongly emphasized (stronger overall slope) for higher pitch gains. Innovation filter 205 has the effect of lowering the energy of the innovative codevector $\mathbf{c}_k$ at low frequencies when the excitation signal $\mathbf{u}$ is more periodic, which enhances the periodicity of the excitation signal $\mathbf{u}$ at lower frequencies more than higher frequencies. Suggested form for innovation filter 205 is

$$F(z)=-\alpha z+1-\alpha z^{-1}$$

where α is a periodicity factor derived from the level of periodicity of the excitation signal $\mathbf{u}$. The periodicity factor α is computed in the voicing factor generator 204. First, a voicing factor r_v is computed in voicing factor generator 204 by

$$r_v = (E_v - E_c) / (E_v + E_c)$$

where E_v is the energy of the scaled pitch codevector $b\mathbf{v}_T$ and E_c is the energy of the scaled innovative codevector $g\mathbf{c}_k$. That is

$$E_v = b^2 \mathbf{v}_T{}^t \mathbf{v}_T = b^2 \sum_{n=0}^{N-1} v_T^2(n)$$

and

$$E_c = g^2 \mathbf{c}_k{}^t \mathbf{c}_k = g^2 \sum_{n=0}^{N-1} c_k^2(n)$$

Note that the value of r_v lies between -1 and 1 (1 corresponds to purely voiced signals and -1 corresponds to purely unvoiced signals).

In this preferred embodiment, the factor α is then computed in voicing factor generator 204 by

$$\alpha = 0.125\ (1 + r_v)$$

which corresponds to a value of 0 for purely unvoiced signals and 0.25 for purely voiced signals.

The enhanced signal $\mathbf{c}_f$ is therefore computed by filtering the scaled innovative codevector gc_k through the innovation filter 205 ($F(z)$).

The enhanced excitation signal $\mathbf{u}'$ is computed by the adder 220 as:

$$\mathbf{u}' = \mathbf{c}_f + b\mathbf{v}_T$$

Note that this process is not performed at the encoder 100. Thus, it is essential to update the content of the pitch codebook 201 using the excitation signal **u** without enhancement to keep synchronism between the encoder 100 and decoder 200. Therefore, the excitation signal **u** is used to update the memory 203 of the pitch codebook 201 and the enhanced excitation signal **u'** is used at the input of the LP synthesis filter 206.

The synthesized signal **s'** is computed by filtering the enhanced excitation signal **u'** through the LP synthesis filter 206 which has the form $1/\hat{A}(z)$, where $\hat{A}(z)$ is the interpolated LP filter in the current subframe. As can be seen in Figure 7, the quantized LP coefficients $\hat{A}(z)$ on line 225 from demultiplexer 217 are supplied to the LP synthesis filter 206 to adjust the parameters of the LP synthesis filter 206 accordingly. The deemphasis filter 207 is the inverse of the preemphasis filter 103 of Figure 6. The transfer function of the deemphasis filter 207 is given by

$$D(z) = 1/(1 - \mu z^{-1})$$

where μ is a preemphasis factor with a value located between 0 and 1 (a typical value is $\mu = 0.7$). A higher-order filter could also be used.

The vector **s'** is filtered through the deemphasis filter $D(z)$ (module 207) to obtain the vector $\mathbf{s}_d$, which is passed through the high-pass filter 208 to remove the unwanted frequencies below 50 Hz and further obtain $\mathbf{s}_h$.

The over-sampling module 209 conducts the inverse process of the down-sampling module 101 of Figure 6. In this preferred embodiment, oversampling converts from the 12.8 kHz sampling rate to the original 16 kHz sampling rate, using techniques well known to those of ordinary skill in the art. The oversampled synthesis signal is denoted $\hat{\mathbf{s}}$. Signal $\hat{\mathbf{s}}$ is also referred to as the synthesized wideband intermediate signal.

The oversampled synthesis signal $\hat{\mathbf{s}}$ does not contain the higher frequency components which were lost by the downsampling process (module 101 of Figure 6) at the encoder 100. This gives a low-pass perception to the synthesized speech signal. To restore the full band of the original signal, a high frequency generation procedure is perform in modules 210 and requires input from voicing factor generator 204 (Figure 7).

The resulting band-pass filtered noise sequence $\mathbf{z}$ is added in adder 221 to the oversampled synthesized speech signal $\hat{\mathbf{s}}$ to obtain the final reconstructed sound signal $\mathbf{s}_{out}$ on the output 223.

Application number / numéro de demande: 2392640

Figures: 5

Pages:

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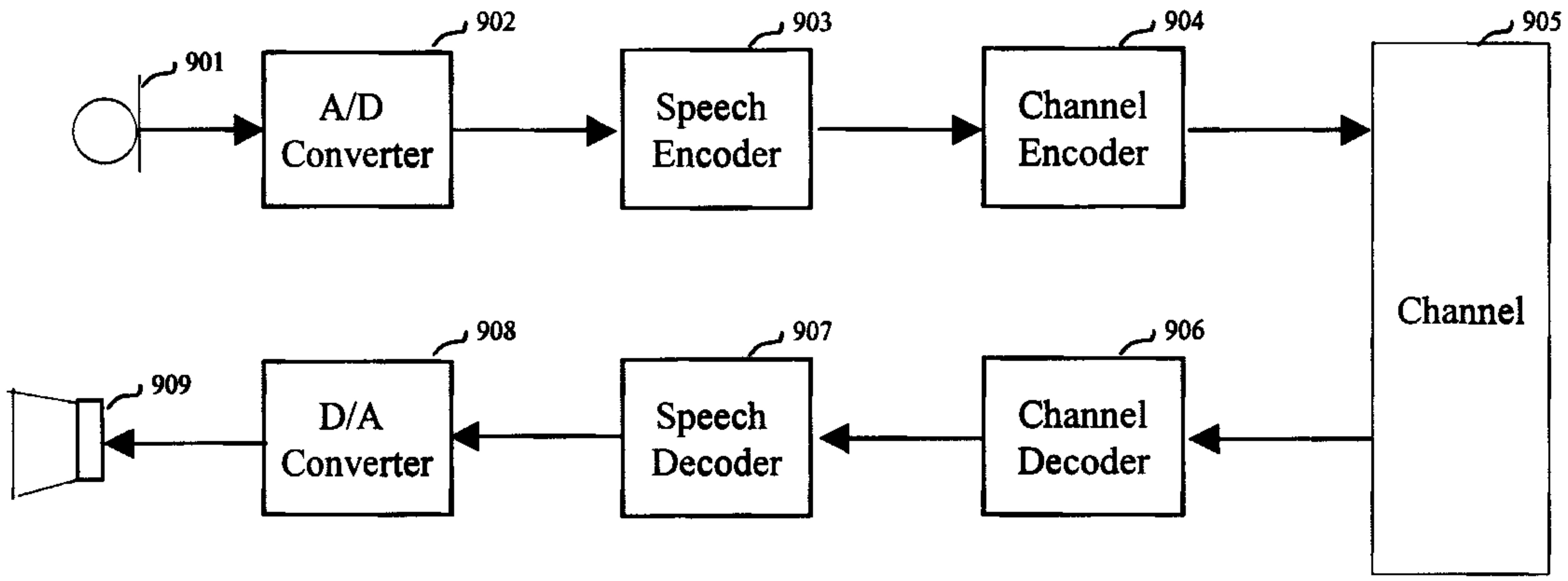


Figure 1: Schematic block diagram of a speech communication system illustrating the use of speech encoding and decoding devices in accordance with the present invention.

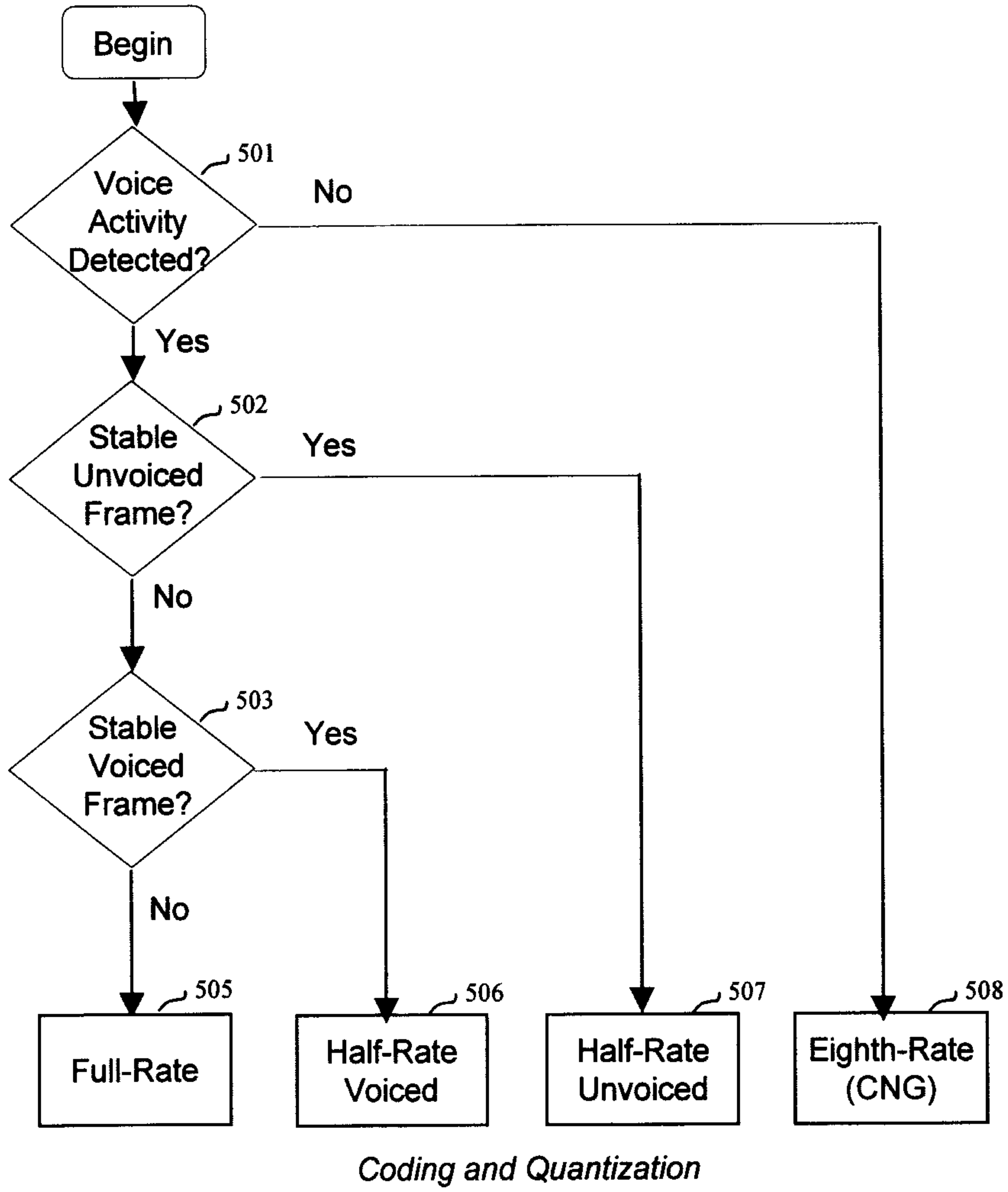


Figure 2: Functional block diagram of a variable bit rate codec with rate determination logic in accordance with a preferred embodiment of the present invention.

Figure 3: Functional block diagram of Figure 2 with including the new interoperable half-rate and its use within the rate determination logic in accordance with a preferred embodiment of the present invention.

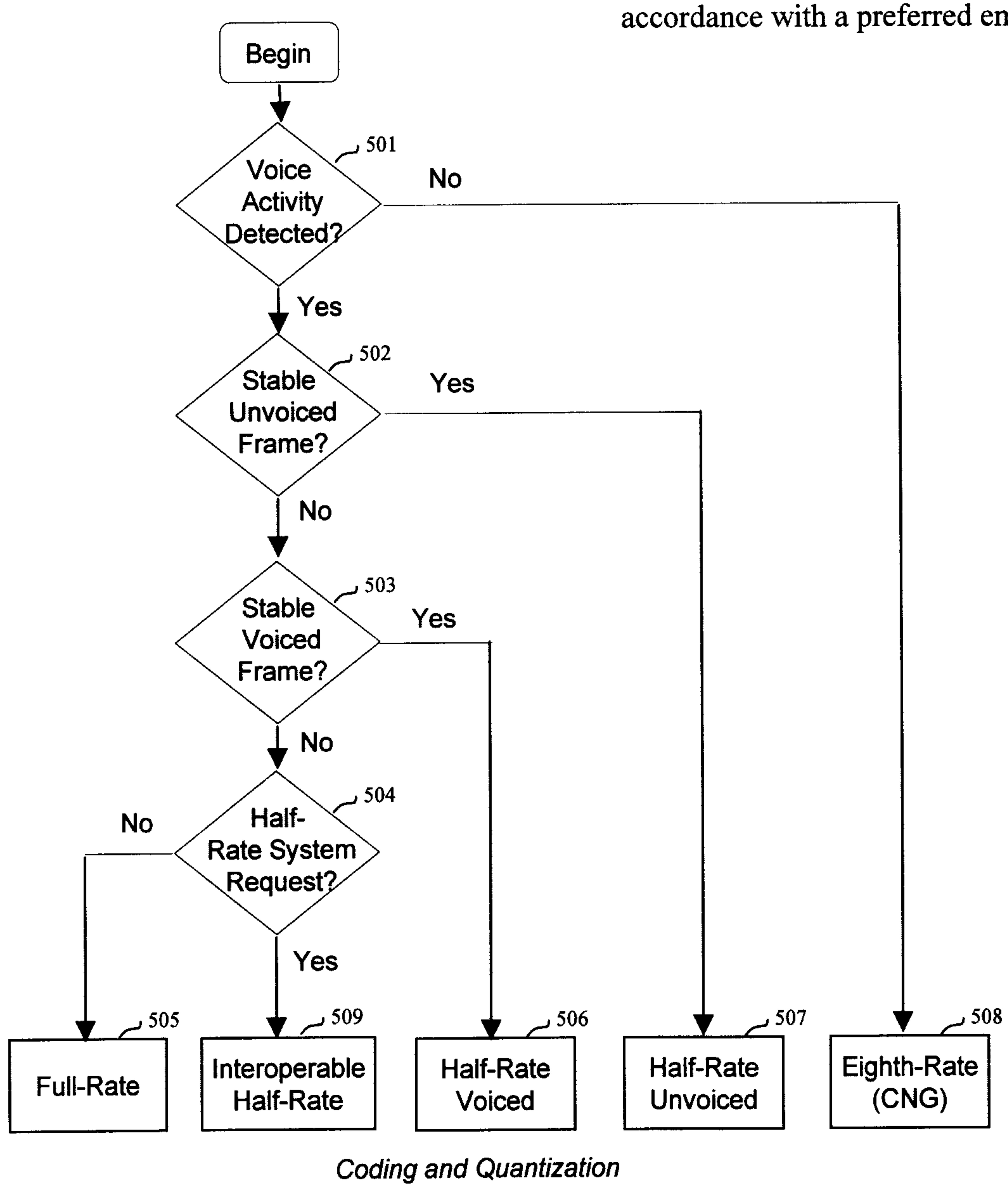
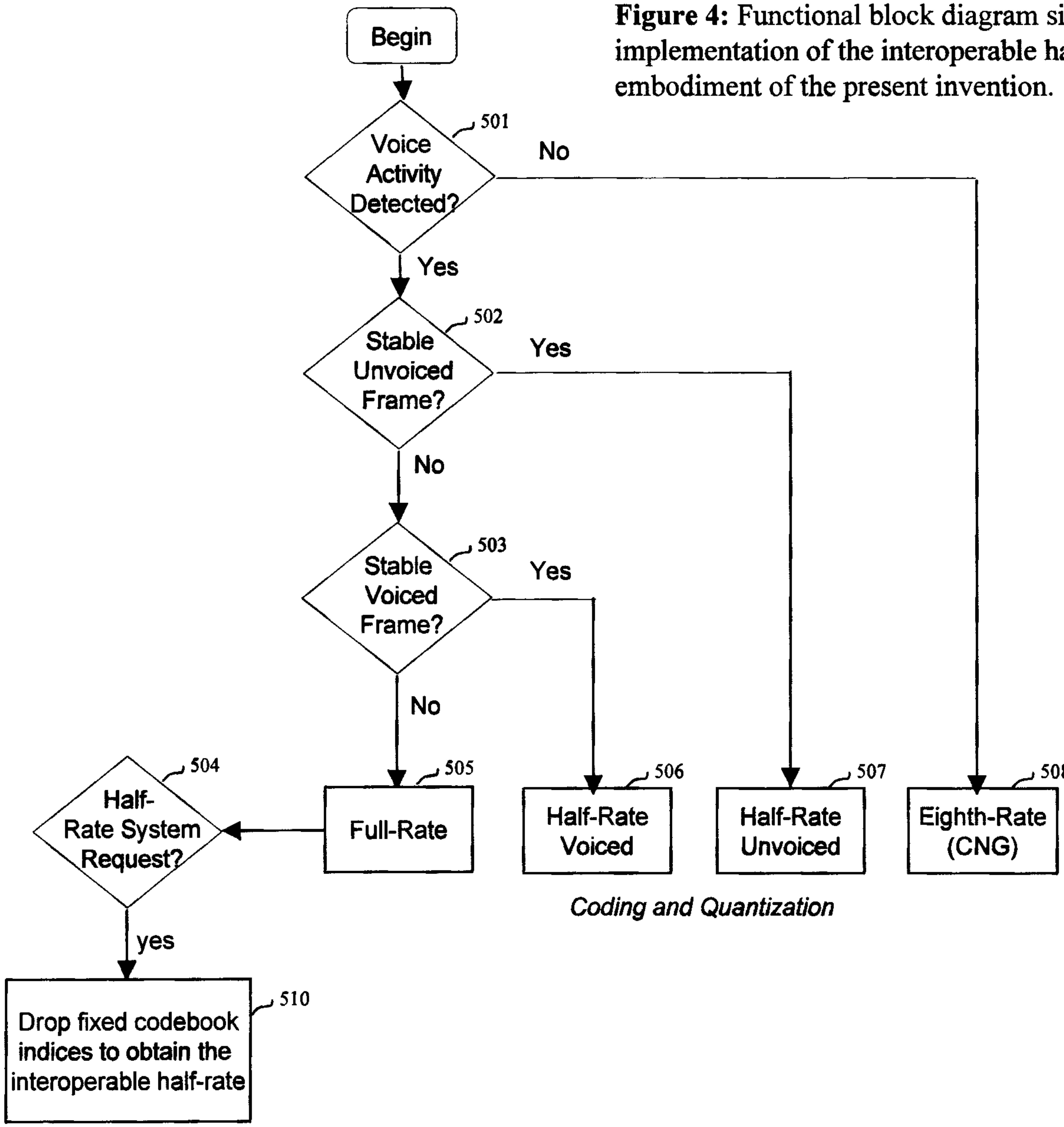


Figure 4: Functional block diagram similar to Figure 3 showing an alternative implementation of the interoperable half-rate in accordance with a preferred embodiment of the present invention.



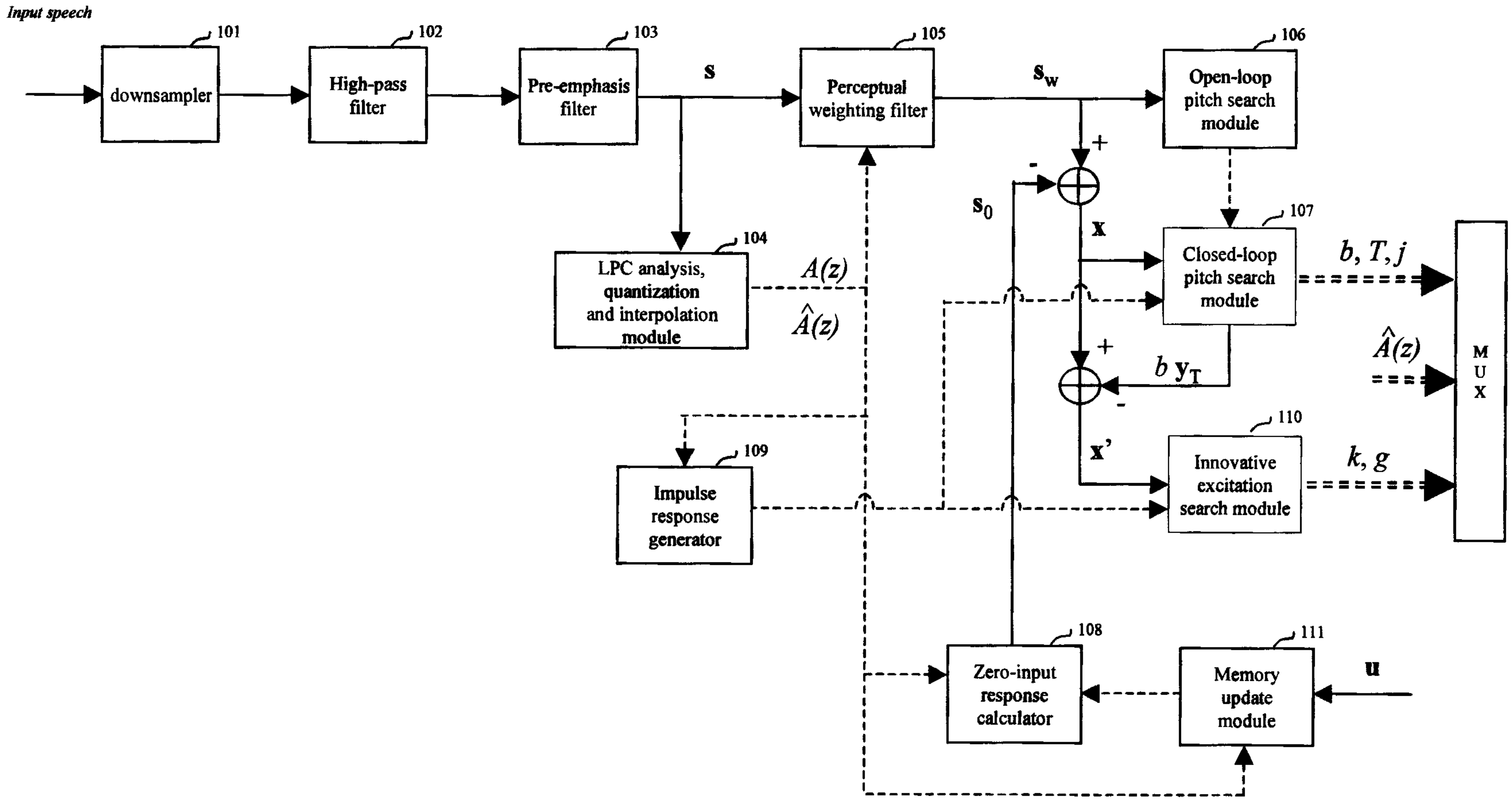


Figure 6: Schematic block diagram of a preferred embodiment of wideband encoding device (AMR-WB encoder).

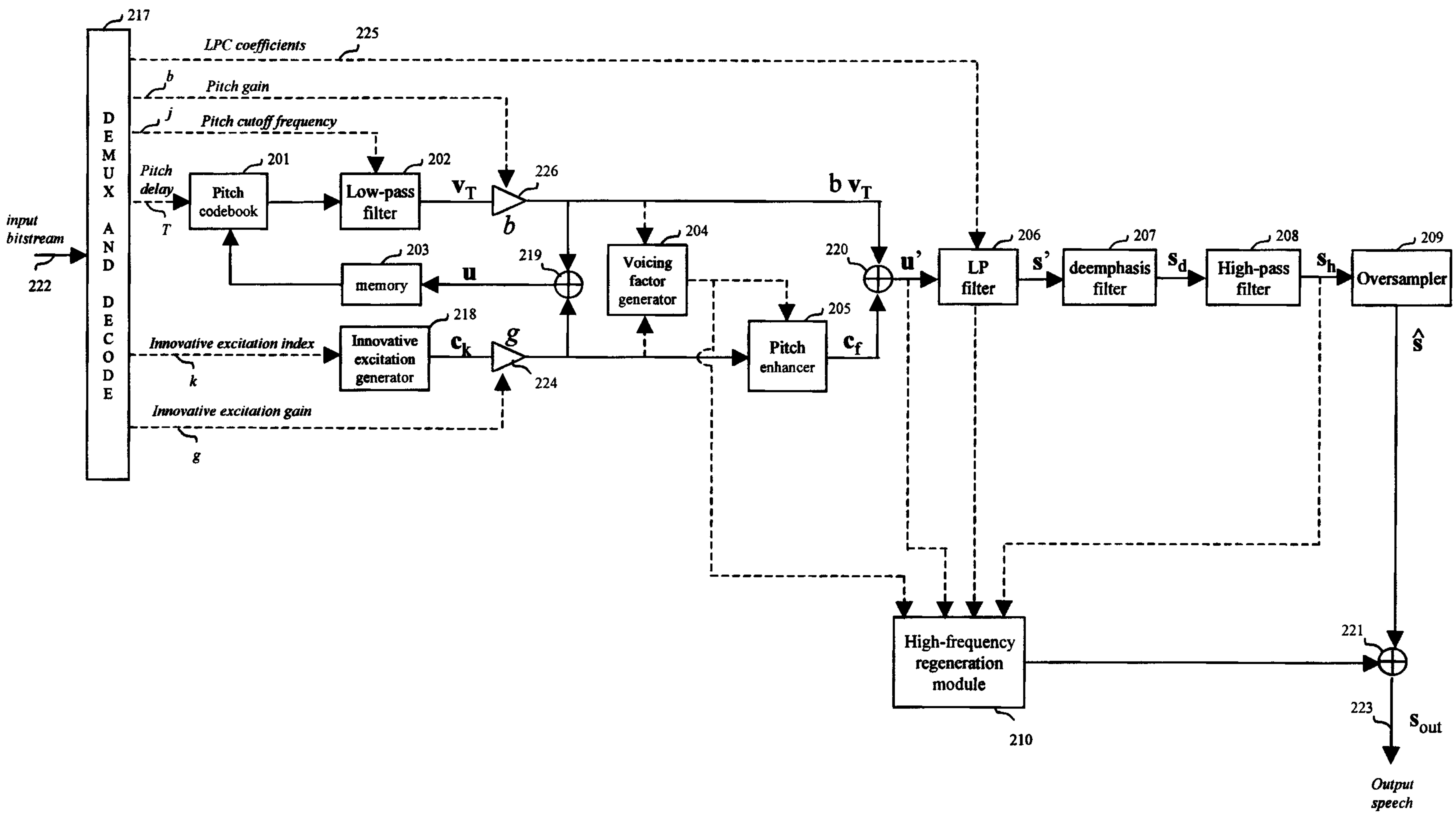


Figure 7: Schematic block diagram of a preferred embodiment of wideband decoding device (AMR-WB decoder).

Functional block diagram of Figure 2 with including the new interoperable half-rate and its use within the rate determination logic in accordance with a preferred embodiment of the present invention.