

**ARIB STD-T64-C.S0052-A v1.0**

**Source-Controlled Variable-Rate  
Multimode Wideband Speech  
Codec (VMR-WB), Service  
Option 62 and 63 for Spread  
Spectrum Systems**

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1 **Original Specification**

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2 This standard, ARIB STD-T64-C.S0052-A v1.0, was prepared by 3GPP2-WG of Association of  
3 Radio Industries and Businesses (ARIB) based upon the 3GPP2 specification, C.S0052-A v1.0.

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5 **Modification to the original specification**

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6 None.

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8 **Notes**

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9 None.

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**3GPP2 C.S0052-A**

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**3RD GENERATION  
PARTNERSHIP  
PROJECT 2  
"3GPP2"**

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**Source-Controlled Variable-Rate Multimode  
Wideband Speech Codec (VMR-WB),  
Service Options 62 and 63 for Spread Spectrum  
Systems**

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## Revision History

Revision	Description	Date
3GPP2 C.S0052-0 v1.0	VMR-WB algorithmic description (Service Option 62) corresponding to operating modes in Rate-Set II	July 2004
3GPP2 C.S0052-A v1.0	VMR-WB algorithmic description (Service Options 62 and 63) corresponding to operating modes in Rate-Set I and Rate-Set II  Note: The algorithmic description and the operation of Rate-Set II compliant modes of VMR-WB (i.e., VMR-WB mode 0, 1, 2, 2 with maximum half-rate, and 3) are identical to those described in 3GPP2 C.S0052-0 v1.0.	April 2005

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## FOREWORD

These technical requirements form a standard for Service Options 62 and 63, source-controlled variable-rate multimode two-way wideband speech Service Option (VMR-WB). VMR-WB has a number of operating modes where each mode corresponds to a certain quality and average data rate. VMR-WB modes 0, 1, and 2 associated with Service Option 62 are fully compliant with Rate-Set II of CDMA systems. The maximum speech-coding rate of the Service Option 62 is 13.3 kbps. VMR-WB mode 4 associated with Service Option 63 is fully compliant with Rate-Set I of CDMA systems. The maximum speech-coding rate of the Service Option 63 is 8.55 kbps.

VMR-WB standard is also interoperable with 3GPP/AMR-WB (ITU-T/G.722.2) standard at 12.65, 8.85, and 6.60 kbps. While support for AMR-WB codec modes 6.60 and 8.85 kbps is included in VMR-WB, due to insufficient quality of these modes, the use of these modes for voice services is not recommended. The VMR-WB acronym has been chosen to reflect the algorithmic similarities and interoperability between the two codecs. This document further describes the necessary interworking functions for establishing an interoperable interconnection between VMR-WB and AMR-WB in circuit-switched networks. The details of interoperable interconnection between VMR-WB and AMR-WB in packet-switched networks as well as transport of VMR-WB encoded speech data over the Internet Protocol are described in a separate specification developed by Internet Engineering Task Force (IETF) for VMR-WB [29]. The applications of the AMR-WB interoperable mode and methods for initiation and setup of interoperable calls are beyond the scope of this specification.

The VMR-WB standard, while a wideband speech codec by default, is capable of processing narrowband input speech signals and produce narrowband outputs in all modes of operation. Therefore, the procedures for initialization and call setup using Service Options 62 and 63 can utilize either or both narrowband and wideband speech processing capabilities of VMR-WB codec.

This standard does not address the quality or reliability of Service Options 62 and 63, nor does it cover equipment performance or measurement procedures.

**NOTES**

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1. The appropriate version of the Minimum Performance Standard for the variable-rate multimode wideband speech codec, Service Options 62 and 63 provides specifications and measurement methods.
2. “Base station” refers to the functions performed on the landline side, which are typically distributed among a cell, a sector of a cell, a mobile switching center, and a personal communications switching center.
3. This document uses the following verbal forms: “Shall” and “shall not” identify requirements to be followed strictly to conform to the standard and from which no deviation is permitted. “Should” and “should not” indicate that one of several possibilities is recommended as particularly suitable, without mentioning or excluding others; that a certain course of action is preferred but not necessarily required; or that (in the negative form) a certain possibility or course of action is discouraged but not prohibited. “May” and “need not” indicate a course of action permissible within the limits of the standard. “Can” and “cannot” are used for statements of possibility and capability, whether material, physical, or causal.
4. Footnotes appear at various points in this specification to elaborate and further clarify items discussed in the body of the specification.
5. Unless indicated otherwise, this document presents numbers in decimal form.  
Binary numbers are distinguished in the text by the use of single quotation marks. In some tables, binary values may appear without single quotation marks if table notation clearly specifies that values are binary. The character ‘x’ is used to represent a binary bit of unspecified value. For example ‘xxx00010’ represents any 8-bit binary value such that the least significant five bits equal ‘00010’.  
Hexadecimal numbers (base 16) are distinguished in the text by use of the form 0x...h, where h...h represents a string of hexadecimal digits. For example, 0x2FA1 represents a number whose binary value is ‘10111110100001’ and whose decimal value is 12913.

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**NOTES**

2 6. The following conventions apply to mathematical expressions in this standard:

- 3 •  $\lfloor x \rfloor$  indicates the largest integer less than or equal to  $x$ :  $\lfloor 1.1 \rfloor = 1$ ,  $\lfloor 1.0 \rfloor = 1$ , and  $\lfloor -1.1 \rfloor = -2$ .
- 4 •  $\lceil x \rceil$  indicates the smallest integer greater than or equal to  $x$ :  $\lceil 1.1 \rceil = 2$ ,  $\lceil 2.0 \rceil = 2$ , and  $\lceil -1.1 \rceil = -1$ .
- 5
- 6 •  $|x|$  indicates the absolute value of  $x$ :  $|-17| = 17$ ,  $|17| = 17$ .
- 7 •  $\oplus$  indicates exclusive OR.
- 8 •  $\min(x, y)$  indicates the minimum of  $x$  and  $y$ .
- 9 •  $\max(x, y)$  indicates the maximum of  $x$  and  $y$ .
- 10 • In figures,  $\otimes$  indicates multiplication. In formulas within the text, multiplication is implicit. For
- 11 example, if  $h(n)$  and  $p_L(n)$  are functions, then  $h(n) p_L(n) = h(n) \otimes p_L(n)$ .
- 12 •  $x \bmod y$  indicates the remainder after dividing  $x$  by  $y$ :  $x \bmod y = x - (y \lfloor x / y \rfloor)$ .
- 13 •  $\text{round}(x)$  is traditional rounding:  $\text{round}(x) = \text{sign}(x) \lfloor |x| + 0.5 \rfloor$ , where

14 
$$\text{sign}(x) = \begin{cases} 1 & x \geq 0 \\ -1 & x < 0 \end{cases}$$

- 15 •  $\sum$  indicates summation. If the summation symbol specifies initial and terminal values, and the
- 16 initial value is greater than the terminal value, then the value of the summation is 0. For example,
- 17 if  $N=0$ , and if  $f(n)$  represents an arbitrary function, then

18 
$$\sum_{n=1}^N f(n) = 0.$$

- 19 • The bracket operator,  $[ ]$ , isolates individual bits of a binary value.  $\text{VAR}[n]$  refers to bit  $n$  of the
- 20 binary representation of the value of the variable  $\text{VAR}$ , such that  $\text{VAR}[0]$  is the least significant bit
- 21 of  $\text{VAR}$ . The value of  $\text{VAR}[n]$  is either 0 or 1.
- 22 • Unless otherwise specified  $\log(x)$  denotes logarithm at base 10 throughout this document.

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## REFERENCES

The following standards contain provisions; through reference in this text constitute provisions of this Standard. At the time of publication, the editions indicated were valid. All standards are subject to revision, and parties to agreements based on this Standard are encouraged to investigate the possibility of applying the most recent editions of the standards indicated below. ANSI and TIA maintain registers of currently valid national standards published by them.

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# 1 SOURCE-CONTROLLED VARIABLE-RATE MULTIMODE TWO-WAY WIDEBAND VOICE COMMUNICATION (VMR-WB)

## 4 1.1 General Description

5 Service Options 62 and 63, the source-controlled variable-rate multimode wideband speech codec  
6 (VMR-WB), provides two-way voice communication between the base station and the mobile station  
7 using the dynamically variable data rate speech codec algorithm described in this standard. The  
8 transmitting speech codec receives voice samples and generates an encoded speech packet for  
9 every Traffic Channel frame\*. The receiving station generates a speech packet from every Traffic  
10 Channel frame and supplies it to the speech codec for decoding into voice samples.

11  
12 It should be noted that the contents of this document describe all operational modes of the VMR-WB  
13 codec inclusive of the AMR-WB interoperable mode and its associated circuit-switched interworking  
14 functions. While both Service Options 62 and 63 as well as the AMR-WB interoperable mode share  
15 the same algorithmic description, Service Option 62 provides methods for initialization and call setup  
16 using VMR-WB modes 0, 1, 2, and 2 with maximum half-rate. Service Option 63 includes VMR-WB  
17 modes 4 and 4 with maximum half-rate. Furthermore Service Options 62 and 63 shall be the primary  
18 Service Options for circuit-switched wideband voice calls in cdma2000<sup>®</sup> terminals that support VMR-  
19 WB codec. The applications and call setup using the AMR-WB interoperable mode are beyond the  
20 scope of this document.

21  
22 VMR-WB modes 0, 1, 2, and 3 communicate at one of four rates: 13300 bps, 6200 bps, 2700 bps and  
23 1000 bps corresponding to Rate-Set II of CDMA systems, whereas VMR-WB mode 4 communicates  
24 at one of three rates: 8550 bps, 4000 bps, and 800 bps corresponding to Rate-Set I of CDMA  
25 systems. Note that Rate-Set I Quarter-Rate is not used.

26 During an interoperable interconnection using VMR-WB mode 3, the operation bandwidth shall be  
27 wideband and switching to other modes shall not be allowed.

28 All implementations shall meet the minimum performance requirements defined in the appropriate  
29 version of the VMR\_WB Minimum Performance Specification.

30 The algorithmic description and the operation of Rate-Set II compliant modes of VMR-WB (i.e., VMR-  
31 WB mode 0, 1, 2, 2 with maximum half-rate, and 3) are identical to those described in [30].

## 32 1.2 Overview of VMR-WB Documentation

33 Release A of the VMR-WB specification family consists of two standards. This standard provides the  
34 algorithmic description of the VMR-WB as well as the master floating-point C-simulation of the codec.

---

\* 3GPP2 C.S0003-0 uses the term "frame" to represent a 20 ms grouping of data on the Fundamental Channel. Common speech codec terminology also uses the term "frame" to represent a quantum of processing. For Service Option 62, the speech codec frame corresponds to speech sampled over 20 ms. The speech samples are processed into a packet. This packet is transmitted in a Traffic Channel frame.

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1 The companion minimum performance standard, “Recommended Minimum Performance Standard  
2 for the Variable-Rate Multimode Wideband Speech Codec, Service Options 62 and 63”, consists of  
3 the master fixed-point C-simulation of the VMR-WB codec as well as minimum performance  
4 specification and the associated test vectors and processing tools. The minimum performance  
5 specification further consists of a set of objective and subjective tests used to verify the quality of any  
6 non bit-exact VMR-WB implementation.

7 Section 2 of this standard provides information on the interfaces between VMR-WB and generic  
8 cdma2000 air-interface (i.e., these technical specifications do not rely upon a particular version of  
9 cdma2000 air-interface). Section 3 provides information on the audio interfaces to VMR-WB. The  
10 specifications given by Sections 4, 5, and 6 of this standard provide the algorithmic description of  
11 VMR-WB. The necessary interworking functions for establishing an interoperable interconnection  
12 between VMR-WB and AMR-WB in circuit-switched networks are described in Section 7. Section 8  
13 provides the detailed description of VMR-WB frame structure and storage file format. The support for  
14 TDD/TTY and low rate in-band data is described in Section 9.  
15

16 The floating-point C-code accompanying this document provides a more detailed complementary  
17 description of VMR-WB. In the case of a discrepancy between the floating-point C-simulation and the  
18 algorithmic description, the floating-point C-simulation will prevail.  
19

### 20 **1.3 VMR-WB Implementation Options**

21  
22 There are two options available for implementation of the VMR-WB:  
23

24 If a bit-exact implementation approach is chosen, the implementation that is developed shall be end-  
25 to-end bit-exact to the reference fixed-point C-simulation. Bit-exactness is verified via application of  
26 the appropriate version of the VMR\_WB Minimum Performance Specification, which includes the  
27 reference fixed-point codec simulation as well as a number of test vectors.  
28

29 Alternatively, an implementation that deviates from the end-to-end bit-exact VMR-WB specification  
30 described above may be developed. Such an implementation shall pass the objective and subjective  
31 tests defined by the appropriate version of the VMR\_WB Minimum Performance Specification.  
32

### 33 **1.4 VMR-WB Algorithmic Options**

34  
35 By default, VMR-WB is interoperable with 3GPP/AMR-WB (ITU-T/G.722.2) only at 12.65 kbps in  
36 mode 3. However, the interoperability can be expanded to 3GPP/AMR-WB (ITU-T/G.722.2) at 6.60  
37 and 8.85 kbps by selecting optional compilation flag, EXPANDED\_INTEROPERABILITY, in the VMR-  
38 WB C simulation. Note that the support of AMR-WB codec modes 6.60 and 8.85 kbps in VMR-WB is  
39 optional.  
40

41 While in the AMR-WB interoperable mode, mode switching is not allowed. There is only one AMR-  
42 WB interoperable mode in VMR-WB. During an AMR-WB interoperable interconnection, the AMR-WB  
43 codec on the GSM/WCDMA side depending on channel conditions may request VMR-WB encoder to  
44 switch between AMR-WB codec modes 0, 1, and 2 corresponding to 6.60, 8.85, and 12.65 kbps. In-  
45 band data embedded in VMR-WB frame structure, as shown in Section 8, is used during an AMR-WB  
46 interoperable interconnection to switch between AMR-WB codec modes 0, 1, or 2 without changing  
47 the operation mode of VMR-WB. Other AMR-WB codec modes that are not specified in this  
48 document are not supported in VMR-WB. While support for AMR-WB codec modes 6.60 and 8.85  
49 kbps is included in VMR-WB, due to insufficient quality of these modes, the use of these modes for  
50 voice services is not recommended.  
51

1 The encoding and decoding procedures of the optional AMR-WB 8.85 and 6.60 kbps codec modes in  
2 VMR-WB C simulation are identical with the corresponding modes in AMR-WB codec. The  
3 algorithmic description of the optional AMR-WB 8.85 and 6.60 kbps codec modes are not included in  
4 this document and can be found in 3GPP/AMR-WB specifications [10-18].  
5

6 By default, the VMR-WB codec is capable of detection and concealment of frames with corrupted rate  
7 information. However, the sensitivity of the bad rate detection algorithm can be increased by selecting  
8 the compilation option BRH\_LEVEL2 in the C simulation of the VMR-WB codec (see Section 6.7).  
9

## 10 **1.5 Narrowband and Wideband Processing Capabilities of VMR-WB**

11  
12 The VMR-WB codec, while a wideband speech codec, by default, has the capability to operate with 8  
13 kHz sampled input/output speech signals in all modes of operation. The VMR-WB decoder does not  
14 require a priori knowledge about the sampling rate of the original media (i.e., speech/audio signals  
15 sampled at 8 or 16 kHz) at the input of the encoder. The VMR-WB decoder, by default, generates 16  
16 kHz wideband output regardless of the encoder input sampling frequency. Depending on the  
17 application, the decoder can be configured to generate 8 kHz output, as well. Also note that the  
18 encoder output packet formats for the narrowband or wideband input are identical and  
19 indistinguishable.  
20

21 The VMR-WB encoder, by default, is capable of processing 16 kHz sampled signals (i.e., signals  
22 band-limited to 4 or 8 kHz). Therefore, as an example, a narrowband speech signal over-sampled at  
23 16 kHz can also be processed by the VMR-WB encoder and a 16 kHz wideband is generated by the  
24 VMR-WB decoder.  
25

26 Therefore, the procedures for initialization and call setup using Service Options 62 and 63 can utilize  
27 either or both narrowband and wideband speech processing capabilities of VMR-WB codec.  
28

29 While dynamic mode switching is allowed between all modes associated with Service Option 62 with  
30 a minimum mode-switching period of 20ms, change of operational bandwidth is not recommended  
31 during a conversation. Furthermore, the implementation of mode-switching for the purpose of  
32 changing the operational bandwidth using the Service Option Control Message is optional (see  
33 Section 2.3.1.2.1). The default operation bandwidth is wideband. Once the operational bandwidth is  
34 specified, the mode of operation can be dynamically switched during a conversation. It should be  
35 noted that dynamic switching of the input/output sampling rate between 8 kHz and 16 kHz is not  
36 described in this specification and is not supported by the accompanied software (floating-point and  
37 fixed-point simulations).  
38

39 Dynamic mode switching is allowed between the two modes associated with Service Option 63 with a  
40 minimum mode-switching period of 20ms.  
41

## 42 **1.6 Service Option Numbers**

43 The variable-rate multimode two-way wideband/narrowband voice Service Options, using the speech  
44 codec algorithm described by this standard, shall use Service Option number 62, for initialization and  
45 call set up using VMR-WB modes 0, 1, 2, or 2 with maximum half-rate, and shall be called Service  
46 Option 62. In addition, Service Option number 63 shall be used for initialization and call set up using  
47 VMR-WB mode 4 or mode 4 with maximum half-rate, and shall be called Service Option 63.  
48

## 49 **1.7 Allowable Delays**

50

1 **1.7.1 Allowable Transmitting Speech Codec Encoding Delay**

2 The transmitting speech codec shall supply a packet to the multiplex sublayer no later than 20 ms  
3 after it has obtained the last input sample for the current speech frame.

4  
5 **1.7.2 Allowable Receiving Speech Codec Decoding Delay**

6 The receiving decoder shall generate the first sample of speech using parameters from a packet  
7 supplied to it by the multiplex sublayer no later than 3 ms after being supplied the packet.

8

9 **1.8 Special Cases**

10

11 **1.8.1 Blanked Packets**

12 A blanked frame occurs when the transmitting station uses the entire frame for either signaling traffic  
13 or secondary traffic. The VMR-WB encoder does no special encoding process during the generation  
14 of a blank packet; i.e., the generated voice packet is simply not used. The decoder, in turn, treats a  
15 blank packet in the same manner as a frame erasure.

16

17 **1.8.2 Null Traffic Channel Data**

18 A Rate 1/8 packet with all bits set to '1' is considered as null Traffic Channel data. This packet is  
19 declared an erased packet and handled as described in Section 6.5. If more than two consecutive all-  
20 ones Rate 1/8 packets are received, the decoder's output shall be muted until a valid packet is  
21 received.

22

23 **1.8.3 All Zeros Packet**

24 Rate 1, Rate 1/2, Rate 1/4, and Rate 1/8 packets with all bits set to '0' shall be considered erased  
25 frames by the decoder and shall be handled as described in Section 6.5.

26

## 1 1.9 Terms and Numeric Information

2 **Adaptive Codebook (ACB).** The adaptive codebook contains excitation vectors that are adapted for  
 3 every subframe. The adaptive codebook is derived from the long-term filter state. The lag value can  
 4 be viewed as an index into the adaptive codebook.

5 **Algebraic Codebook.** A fixed-codebook where algebraic code is used to populate the excitation  
 6 vectors (innovation vectors). The excitation contains a small number of nonzero pulses with  
 7 predefined interlaced sets of potential positions. The amplitudes and positions of the pulses of the  $k$ th  
 8 excitation codevector can be derived from its index  $k$  through a rule requiring no or minimal physical  
 9 storage, in contrast with stochastic codebooks, whereby the path from the index to the associated  
 10 codevector involves look-up tables.

11 **Algebraic Code Excited Linear Predictive Coding (ACELP).** The algorithm that is used by the  
 12 encoder to generate the stochastic component of the excitation where an excitation vector contains a  
 13 few number of non-zero pulses with predefined interlaced sets of positions. The pulses have their  
 14 amplitudes fixed to +1 or -1, and each pulse has a set of possible positions distinct from the positions  
 15 of the other pulses. The set of positions are interlaced. The excitation code is identified by the  
 16 positions of its non-zero pulses. The codebook search is essentially searching the optimum position  
 17 of the non-zero pulses.

18 **Anti-Sparseness Processing.** An adaptive post-processing procedure applied to the fixed-codebook  
 19 vector in order to reduce perceptual artifacts from a sparse fixed-codebook vector.

20 **Autocorrelation Function.** A function showing the relationship of a signal with a time-shifted version  
 21 of itself.

22 **Base Station.** A station in the Public Radio Telecommunications Service, other than a  
 23 personal/mobile station, used for radio communications with personal/mobile stations.

24 **Closed-Loop Pitch Analysis:** This is the adaptive codebook search, i.e. a process of estimating the  
 25 pitch (lag) value from the weighted input speech and the long-term filter state. In the closed-loop  
 26 search, the lag is searched using error minimization loop (analysis-by-synthesis). In the VMR-WB  
 27 codec, closed-loop pitch search is performed for every subframe.

28 **Codec.** The combination of an encoder and decoder in series (encoder/decoder).

29 **Code Excited Linear Predictive Coding (CELP).** A speech-coding algorithm, CELP coders use  
 30 codebook excitation, a long-term pitch prediction filter, and a short-term formant prediction filter.

31 **Codebook.** A set of vectors used by the speech codec. For each speech codec codebook  
 32 subframe, one particular vector is chosen and is used to excite the speech codec's filters. The  
 33 codebook vector is chosen to minimize the weighted error between the original and synthesized  
 34 speech after the pitch and formant synthesis filter coefficients have been determined.

35 **Coder.** Same as "encoder."

36 **Decoder.** Generally, a device for the translation of a signal from a digital representation into an  
 37 analog format. For this standard, a device that converts speech encoded in the format specified in  
 38 this standard to analog or an equivalent PCM representation.

39 **Encoder.** Generally, a device for the translation of a signal into a digital representation. For this  
 40 standard, a device that converts speech from an analog, or from its equivalent PCM representation,  
 41 to the digital representation described in this standard.

42 **Formant.** A resonant frequency of the human vocal tract causing a peak in the short-term spectrum  
 43 of speech.

44 **Fractional Lags.** A set of lag values having sub-sample resolution. In VMR-WB codec a sub-sample  
 45 resolution of 1/4th or 1/2nd of a sample is used.

46 **IIR Filter.** (Infinite-duration impulse response filter) A filter for which the output, in response to an  
 47 impulse input, never totally converges to zero. This term is usually used in reference to digital filters.

1 **Interpolating Filter.** An FIR filter used to produce an estimate of sub-sample resolution samples,  
 2 given an input sampled with integer sample resolution. In this implementation, the interpolating filter  
 3 has low-pass filter characteristics. Thus the adaptive codebook consists of the low-pass filtered  
 4 interpolated past excitation.

5 **Inverse Filter:** This filter removes the short-term correlation from the speech signal. The filter models  
 6 an inverse frequency response of the vocal tract.

7 **Lag:** The long term filter delay. This is typically the true pitch period, or its multiple or submultiples.

8 **LP Analysis Window:** For each frame, the short-term filter coefficients are computed using the high  
 9 pass filtered speech samples within the analysis window. In VMR-WB codec, the length of the  
 10 analysis window is always 384 samples. For all the modes, a single asymmetric window is used to  
 11 generate a single set of LP coefficients. The 10 ms look-ahead is used in the analysis.

12 **Linear Predictive Coding (LPC).** A method of predicting future samples of a sequence by a linear  
 13 combination of the previous samples of the same sequence. Linear Predictive Coding is frequently  
 14 used in reference to a class of speech codecs.

15 **Immitance Spectral Frequencies (ISFs).** A representation of digital filter coefficients in a pseudo-  
 16 frequency domain. This representation has good quantization and interpolation properties.

17 **LSB.** Least significant bit.

18 **Mode.** An operating condition of the codec that corresponds to certain average data rate and  
 19 subjective quality.

20 **MSB.** Most significant bit.

21 **Normalized Autocorrelation Function.** A measure used to determine the pitch period and the  
 22 degree of periodicity of the input speech. This measure is useful in distinguishing voiced from  
 23 unvoiced speech.

24 **Open-Loop Pitch Search:** A process of estimating the near optimal lag directly from the weighted  
 25 speech input. This is done to simplify the pitch analysis and confine the closed-loop pitch search to a  
 26 small number of lags around the open-loop estimated lags.

27 **Packet.** The unit of information exchanged between Service Option applications in the base station  
 28 and the personal/mobile station.

29 **Perceptual Weighting Filter:** This filter is employed in the analysis-by-synthesis search of the  
 30 codebooks. The filter exploits the noise masking properties of the formants (vocal tract resonances)  
 31 by weighting the error less in regions near the formant frequencies and more in regions away from  
 32 them.

33 **Personal/Mobile Station.** A station in the Public Radio Telecommunications Service intended to be  
 34 used while in motion or during halts at unspecified points.

35 **Pitch.** The fundamental frequency in speech caused by the periodic vibration of the human vocal  
 36 cords.

37 **Rate-Set I:** The allowable traffic frame rates in cdma2000 Rate-Set I: Rate 1 frames use the 9600 bps  
 38 rate, Rate  $\frac{1}{2}$  frames use the 4800 bps rate, Rate  $\frac{1}{4}$  frames use the 2400 bps rate, and Rate  $\frac{1}{8}$   
 39 frames use the 1200 bps rate. The allowable speech encoding frame rates in cdma2000 Rate-Set I:  
 40 Rate 1 frames use the 8550 bps rate, Rate  $\frac{1}{2}$  frames use the 4000 bps rate, Rate  $\frac{1}{4}$  frames use 2000  
 41 bps rate, and Rate  $\frac{1}{8}$  frames use the 800 bps rate.  
 42

1 **Rate-Set II:** The allowable traffic frame rates in cdma2000 Rate-set II: Rate 1 frames use the 14400  
 2 bps rate, Rate ½ frames use the 7200 bps rate, Rate ¼ frames use the 3600 bps rate and Rate 1/8  
 3 frames use the 1800 bps rate. The allowable speech encoding frame rates in cdma2000 Rate-set II:  
 4 Rate 1 frames use the 13300 bps rate, Rate ½ frames use the 6200 bps rate, Rate ¼ frames use the  
 5 2700 bps rate and Rate 1/8 frames use the 1000 bps rate.

6 **RDA.** Rate Determination Algorithm.

7 **Relaxation Code Excited Linear Predictive Coding (RCELP).** The speech coding algorithm used  
 8 by the encoder where unlike conventional CELP coders, a modified version of the speech signal that  
 9 conforms to a linearly interpolated pitch contour is encoded, relaxing the frequent pitch update  
 10 constraint in low rate CELP coders. This pitch contour is obtained by estimating the pitch values at  
 11 the analysis frame boundaries and linearly interpolating the pitch across frame.

12 **Residual.** The output signal resulting from an inverse filtering operation

13 **RLR.** Receive Loudness Rating, a measure of receive audio sensitivity, as defined in IEEE Standard  
 14 269-2002. The measurement of the receive loudness rating is described in ITU-T Recommendation  
 15 P.79-1999.

16 **Short-Term Synthesis Filter:** This filter introduces short-term correlation into the excitation signal,  
 17 which models the impulse response of the vocal tract.

18 **SLR.** Send Loudness Rating, a measure of transmit audio sensitivity, as defined in IEEE Standard  
 19 269-2002. The measurement of the send loudness rating is described in ITU-T Recommendation  
 20 P.79-1999.

21 **SPL.** Sound Pressure Level.

22 **Subframe:** A time interval equal to 5 ms (80 samples at 16 kHz sampling rate).

23 **Vector Quantization:** A method of grouping several parameters into a vector and quantizing them  
 24 simultaneously.

25 **Voiced Speech.** Speech generated when the vocal cords are vibrating at a fundamental frequency.  
 26 Characterized by high energy, periodicity, and a large ratio of energy below 2 kHz to energy above 2  
 27 kHz.

28 **Unvoiced Speech.** Speech generated by forcing air through constrictions in the vocal tract without  
 29 vibration of the vocal cords. Characterized by a lack of periodicity, and a near-unity ratio of energy  
 30 below 2 kHz to energy above 2 kHz.

31 **WAEPL.** Weighted Acoustic Echo Path Loss. A measure of the echo performance under normal  
 32 conversation. ANSI/EIA/TIA-579-A98 defines the measurement of WAEPL.

33 **Zero Input Response (ZIR).** The filter output caused by the non-zero initial state of the filter when  
 34 no input is present.

35 **Zero State Response (ZSR).** The filter output caused by an input when the initial state of the filter is  
 36 zero.

37 **ZIR.** See Zero Input Response.

38

39 **ZSR.** See Zero State Response.

40

41

42

43

## 2 REQUIRED MULTIPLEX OPTIONS SUPPORT

Service Options 62 and 63 shall support an interface with Multiplex Options 2 and 1, respectively. Speech packets for Service Options 62 and 63 shall only be transported as primary traffic. Service Options 62 and 63 shall be the primary Service Options for circuit-switched wideband voice calls in cdma2000 terminals that support VMR-WB codec.

### 2.1 Interface to Multiplex Option 2

#### 2.1.1 Transmitted Packets

Using Service Option 62, the speech codec shall generate and shall supply exactly one packet to the multiplex sublayer every 20 milliseconds. The packet contains the Service Option information bits that are transmitted as primary traffic. The packet shall be one of five types as shown in Table 2.1-1. The number of bits supplied to the multiplex sublayer for each type of packet shall also be as shown in Table 2.1-1. Unless otherwise commanded, the speech codec may supply a Rate 1, Rate 1/2, Rate 1/4, or Rate 1/8 packet. Upon command, the speech codec shall generate a Blank packet. Also upon command, the speech codec shall generate a non-blank packet with a maximum rate of Rate 1/2.

A Blank packet contains no bits and is used for blank-and-burst transmission of signaling traffic or secondary traffic (see 3GPP2 C.S0003-0 and 3GPP2 C.S0005-0<sup>†</sup>).

**Table 2.1-1. Packet Types Supplied by Service Option 62 to the Multiplex Sublayer**

Packet Type	Bits per Packet
Rate 1	266
Rate 1/2	124
Rate 1/4	54
Rate 1/8	20
Blank	0

#### 2.1.2 Received Packets

The multiplex sublayer in the receiving station categorizes every received Traffic Channel frame, and supplies the packet type and accompanying bits, if any, to the speech codec as shown in Table 2.1-1. The speech codec processes the bits of the packet as described in Sections 6 and 7. The received packet types shown in Table 2.1-2 correspond to the transmitted packet types shown in Table 2.1-1. The Blank packet type occurs when the receiving station determines that a blank-and-burst frame for signaling traffic or secondary traffic was transmitted. When the multiplex sublayer determines that a received frame is in error, the multiplex sublayer supplies an insufficient frame quality (erasure) packet to the Service Option 62.

<sup>†</sup> The technical specifications described in this document do not rely upon a particular version of cdma2000 air-interface.

1 **Table 2.1-2. Packet Types Supplied by the Multiplex Sublayer to Service Option 62**

Packet Type	Bits per Packet
Rate 1	266
Rate 1/2	124
Rate 1/4	54
Rate 1/8	20
Blank	0
Insufficient frame quality (erasure)	0

2

## 3 **2.2 Interface to Multiplex Option 1**

4

### 5 **2.2.1 Transmitted Packets**

6

7 Using Service Option 63, the speech codec shall generate and shall supply exactly one packet to the  
 8 multiplex sublayer every 20 milliseconds. The packet contains the Service Option information bits  
 9 that are transmitted as primary traffic. The packet shall be one of five types as shown in Table 2.2-1.  
 10 The number of bits supplied to the multiplex sublayer for each type of packet shall also be as shown  
 11 in Table 2.2-1. Unless otherwise commanded, the speech codec may supply a Rate 1, Rate 1/2,  
 12 Rate 1/4, or Rate 1/8 packet. Upon command, the speech codec shall generate a Blank packet. Also  
 13 upon command, the speech codec shall generate a non-blank packet with a maximum rate of Rate  
 14 1/2. Note that Rate 1/4 is not used in Service Option 63.

15 A Blank packet contains no bits and is used for blank-and-burst transmission of signaling traffic or  
 16 secondary traffic (see 3GPP2 C.S0003-0 and 3GPP2 C.S0005-0†).

17 **Table 2.2-1. Packet Types Supplied by Service Option 63 to the Multiplex Sublayer**

Packet Type	Bits per Packet
Rate 1	171
Rate 1/2	80
Rate 1/8	16
Blank	0

18

### 19 **2.2.2 Received Packets**

20

21 The multiplex sublayer in the receiving station categorizes every received Traffic Channel frame, and  
 22 supplies the packet type and accompanying bits, if any, to the speech codec as shown in Table 2.2-1.  
 23 The speech codec processes the bits of the packet as described in Sections 6 and 7. The received  
 24 packet types shown in Table 2.2-2 correspond to the transmitted packet types shown in Table 2.2-1.  
 25 The Blank packet type occurs when the receiving station determines that a blank-and-burst frame for  
 26 signaling traffic or secondary traffic was transmitted. When the multiplex sublayer determines that a  
 27 received frame is in error, the multiplex sublayer supplies an insufficient frame quality (erasure)  
 28 packet to the Service Option 63. The Rate 1 with bit errors packet type occurs when the receiving  
 29 station determines that the frame was transmitted at 9600 bps and the frame has one or more bit  
 30 errors. The insufficient frame quality packet type occurs when the mobile station is unable to decide

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† The technical specifications described in this document do not rely upon a particular version of cdma2000 air-interface.

upon the data rate of the received frame or when the mobile station detects a frame in error, which does not belong to the Rate 1 with bit errors, packet type. Although the Service Option 63 does not utilize Rate 1/4 packets, Multiplex Option 1 is not required to recognize this fact; Service Option 63 is, therefore, responsible for declaring Rate 1/4 frames as having insufficient frame quality (erasure).

**Table 2.2-2. Packet Types Supplied by the Multiplex Sublayer to Service Option 63**

Packet Type	Bits per Packet
Rate 1	171
Rate 1/2	80
Rate 1/4	40
Rate 1/8	16
Blank	0
Rate 1 with bit errors	171
Insufficient frame quality (erasure)	0

6

## 7 2.3 Negotiation for Service Option 62

8 The mobile station and base station can negotiate for Service Option 62 service negotiation, as  
 9 described in 3GPP2 C.S0005-0 [7]. This section describes the service negotiation and call set up for  
 10 modes 0, 1, 2, and 2 with maximum half-rate of VMR-WB standard.

11

### 12 2.3.1 Procedures Using Service Negotiation

13 The mobile station and base station shall perform service negotiation for Service Option 62 as  
 14 described in 3GPP2 C.S0005-0 [7], and the negotiated service configuration shall include only valid  
 15 attributes for the Service Option as specified in Table 2.3-1.

16

**Table 2.3-1. Valid Service Configuration Attributes for Service Option 62**

Service Configuration Attribute	Valid Selections
Forward Multiplex Option	Multiplex Option 2
Reverse Multiplex Option	Multiplex Option 2
Forward Transmission Rates	Rate-Set II with all rates enabled
Reverse Transmission Rates	Rate-Set II with all rates enabled
Forward Traffic Type	Primary Traffic
Reverse Traffic Type	Primary Traffic

17

### 18 2.3.1.1 Initialization and Connection

#### 19 2.3.1.1.1 Mobile Station Requirements

20 If the mobile station accepts a service configuration, as specified in a *Service Connect Message*,  
 21 *General Handoff Direction Message*, or *Universal Handoff Direction Message* that includes a Service  
 22 Option connection using Service Option 62, the mobile station shall perform the following:

- 23 • If the Service Option connection is new (that is, not part of the previous service configuration), the  
 24 mobile station shall perform speech codec initialization at the time specified by the maximum of  
 25 the action time associated with the message carrying the Service Configuration Record, and the  
 26 time that the corresponding Call Control Instance is instantiated. The mobile station shall initialize

1 its VMR-WB encoder mode of operation to a default value of 0 and the operational bandwidth to  
2 wideband. The mobile station shall complete the initialization within 40 ms.

- 3 • Beginning at the time specified by the maximum of the action time associated with the message  
4 carrying the Service Configuration Record, and the time that the corresponding Call Control  
5 Instance is instantiated, and continuing for as long as the service configuration includes the  
6 Service Option connection, Service Option 62 shall process received packets and generate and  
7 supply packets for transmission as follows:
  - 8 - If the Call Control Instance is in the *Conversation Substate*, Service Option 62 shall process  
9 the received packets and generate and supply packets for transmission in accordance with  
10 this standard.
  - 11 - If the Call Control Instance is not in the *Conversation Substate*, Service Option 62 shall  
12 process the received packets in accordance with this standard, and shall generate and  
13 supply Rate 1/8 Packets with all bits set to '1' for transmission, except when commanded to  
14 generate a Blank packet.

#### 15 2.3.1.1.2 Base Station Requirements

16 If the base station establishes a service configuration, as specified in a *Service Connect Message*,  
17 *General Handoff Direction Message*, or *Universal Handoff Direction Message* that includes a Service  
18 Option connection using Service Option 62, the base station shall perform the following:

- 19 • If the Service Option connection is new (that is, not part of the previous service configuration), the  
20 base station shall perform speech codec initialization no later than the time specified by the  
21 maximum of the action time associated with the message carrying the Service Configuration  
22 Record, and the time that the corresponding Call Control Instance is Instantiated. The base  
23 station shall initialize its VMR-WB encoder mode of operation to a default value of 0 and the  
24 operational bandwidth to wideband.
- 25 • Commencing at the time specified by the maximum of the action time associated with the  
26 message carrying the Service Configuration Record, and the time that the corresponding Call  
27 Control Instance is Instantiated, and continuing for as long as the service configuration includes  
28 the Service Option connection, Service Option 62 shall process received packets, and shall  
29 generate and supply packets for transmission in accordance with this standard. The base station  
30 may defer enabling the audio input and output.

### 31 2.3.1.2 Service Option Control Messages

#### 32 2.3.1.2.1 Mobile Station Requirements

33 The mobile station shall support one pending *Service Option Control Message* for Service Option 62.

34 If the mobile station receives a *Service Option Control Message* for Service Option 62, then, at the  
35 action time associated with the message, the mobile station shall process the message as follows:

- 36 1. If the MOBILE\_TO\_MOBILE field is equal to '1', the mobile station should disable the audio  
37 output of the speech codec for 1 second after initialization.  
38 If the MOBILE\_TO\_MOBILE field is equal to '0', the mobile station shall process each received  
39 packet as described in Section 6.
- 40 2. If the INIT\_CODEC field is equal to '1', the mobile station shall perform speech codec  
41 initialization. The mobile station shall complete the initialization within 40 ms.
- 42 3. VMR-WB accepts as input, a mode of operation through the RATE\_REDUC field as defined in  
43 Table 2.3-2 using this mode of operation, VMR-WB generates Rate 1, Rate 1/2, Rate 1/4, and Rate  
44 1/8 packets in a proportion that results in the average data rate given by Table 2.3-2.

45 Service Option 62 shall continue to use these fractions until either of the following events occurs:

- 1       • The mobile station receives a *Service Option Control Message* specifying a different
- 2       RATE\_REDUCE, or
- 3       • Service Option 62 is initialized.

4  
 5 Service Option 62 was developed using reduced rate operation (encoding mode of operation) as a  
 6 network control criteria. The VMR-WB codec selects the encoding rate based upon the instantaneous  
 7 characteristics of the input speech: voiced, unvoiced, transition, etc., as well as the encoding mode  
 8 selected.

9  
 10 While dynamic mode switching is allowed between all modes associated with Service Option 62 with  
 11 a minimum mode-switching period of 20ms, change of operational bandwidth is not recommended  
 12 during a conversation. The default operation bandwidth is wideband. Once the operational bandwidth  
 13 is specified, the mode of operation can be dynamically switched during a conversation.

14  
 15 **Table 2.3-2. VMR-WB Encoding Rate Mode Control Parameters**

RATE_REDUCE (Binary)	Operational Bandwidth	VMR-WB Mode of Operation	Estimated Average Encoding Rate kbps (Source Encoding Rates)
'000'	Wideband	0	9.1404
'001'		1	7.6930
'010'		2	6.2847
'011'		2 (HR Max)	4.7443
'100'	Narrowband	0	9.0435
'101'		1	7.5276
'110'		2	6.2109
'111'		2 (HR Max)	4.7518

16 2.3.1.2.2 Base Station Requirements

17 The base station may send a *Service Option Control Message* to the mobile station. If the base  
 18 station sends a *Service Option Control Message*, the base station shall include the following type-  
 19 specific fields for Service Option 62:

20 **Table 2.3-3. Service Option Control Message Type-Specific Fields**

Field	Length (bits)
RATE_REDUCE	3
RESERVED	3
MOBILE_TO_MOBILE	1
INIT_CODEC	1

- 21
- 22 RATE\_REDUCE - VMR-WB mode of operation.  
 23 The base station shall set this field to the RATE\_REDUCE value from  
 24 Table 2.3-2 corresponding to the mode of operation that the mobile  
 25 station is to operate in.
  - 26 RESERVED - Reserved bits.  
 27 The base station shall set this field to '000'.

MOBILE\_TO\_MOBILE

- Mobile-to-mobile processing.

If the mobile station is to perform mobile-to-mobile processing (Section 2.3.1.2.1), the base station shall set this field to '1'. In addition, if the mobile station is to disable the audio output of the speech codec for 1 second after initialization, the base station shall set the INIT\_CODEEC field and the MOBILE\_TO\_MOBILE field to '1'. If the mobile station is not to perform mobile-to-mobile processing, the base station shall set the MOBILE\_TO\_MOBILE field to '0'.

INIT\_CODEEC - Initialize speech codec.

If the mobile station is to initialize the speech codec, the base station shall set this field to '1'. Otherwise, the base station shall set this field to '0'.

**Table 2.3-4. VMR-WB Encoding Rate Mode Control Parameters**

RATE_REDUC (Binary)	Operational Bandwidth	VMR-WB Mode of Operation	Estimated Average Encoding Rate kbps (Source Encoding Rates)
'000'	Wideband	0	9.1404
'001'		1	7.6930
'010'		2	6.2847
'011'		2 (HR Max)	4.7443
'100'	Narrowband	0	9.0435
'101'		1	7.5276
'110'		2	6.2109
'111'		2 (HR Max)	4.7518

**2.4 Negotiation for Service Option 63**

The mobile station and base station can negotiate for Service Option 63 service negotiation, as described in 3GPP2 C.S0005-0 [7]. This section describes the service negotiation and call set up for modes 4, and 4 with maximum half-rate of VMR-WB standard.

**2.4.1 Procedures Using Service Negotiation**

The mobile station and base station shall perform service negotiation for Service Option 63 as described in 3GPP2 C.S0005-0 [7], and the negotiated service configuration shall include only valid attributes for the Service Option as specified in Table 2.4-1.

**Table 2.4-1. Valid Service Configuration Attributes for Service Option 63**

Service Configuration Attribute	Valid Selections
Forward Multiplex Option	Multiplex Option 1
Reverse Multiplex Option	Multiplex Option 1
Forward Transmission Rates	Rate-Set I with all rates enabled
Reverse Transmission Rates	Rate-Set I with all rates enabled
Forward Traffic Type	Primary Traffic
Reverse Traffic Type	Primary Traffic

## 1    **2.4.1.1        Initialization and Connection**

### 2    2.4.1.1.1    Mobile Station Requirements

3    If the mobile station accepts a service configuration, as specified in a *Service Connect Message*,  
4    *General Handoff Direction Message*, or *Universal Handoff Direction Message* that includes a Service  
5    Option connection using Service Option 63, the mobile station shall perform the following:

- 6    • If the Service Option connection is new (that is, not part of the previous service configuration), the  
7    mobile station shall perform speech codec initialization at the time specified by the maximum of  
8    the action time associated with the message carrying the Service Configuration Record, and the  
9    time that the corresponding Call Control Instance is instantiated. The mobile station shall initialize  
10   its VMR-WB encoder mode of operation to a default value of 4 and the operational bandwidth to  
11   wideband. The mobile station shall complete the initialization within 40 ms.
- 12   • Beginning at the time specified by the maximum of the action time associated with the message  
13   carrying the Service Configuration Record, and the time that the corresponding Call Control  
14   Instance is instantiated, and continuing for as long as the service configuration includes the  
15   Service Option connection, Service Option 63 shall process received packets and generate and  
16   supply packets for transmission as follows:
  - 17       - If the Call Control Instance is in the *Conversation Substate*, Service Option 63 shall process  
18       the received packets and generate and supply packets for transmission in accordance with  
19       this standard.
  - 20       - If the Call Control Instance is not in the *Conversation Substate*, Service Option 63 shall  
21       process the received packets in accordance with this standard, and shall generate and  
22       supply Rate 1/8 Packets with all bits set to '1' for transmission, except when commanded to  
23       generate a Blank packet.

### 24   2.4.1.1.2    Base Station Requirements

25   If the base station establishes a service configuration, as specified in a *Service Connect Message*,  
26   *General Handoff Direction Message*, or *Universal Handoff Direction Message* that includes a Service  
27   Option connection using Service Option 63, the base station shall perform the following:

- 28   • If the Service Option connection is new (that is, not part of the previous service configuration), the  
29   base station shall perform speech codec initialization no later than the time specified by the  
30   maximum of the action time associated with the message carrying the Service Configuration  
31   Record, and the time that the corresponding Call Control Instance is Instantiated. The base  
32   station shall initialize its VMR-WB encoder mode of operation to a default value of 4 and the  
33   operational bandwidth to wideband.
- 34   • Commencing at the time specified by the maximum of the action time associated with the  
35   message carrying the Service Configuration Record, and the time that the corresponding Call  
36   Control Instance is Instantiated, and continuing for as long as the service configuration includes  
37   the Service Option connection, Service Option 63 shall process received packets, and shall  
38   generate and supply packets for transmission in accordance with this standard. The base station  
39   may defer enabling the audio input and output.

## 40   **2.4.1.2        Service Option Control Messages**

### 41   2.4.1.2.1    Mobile Station Requirements

42   The mobile station shall support one pending *Service Option Control Message* for Service Option 63.  
43   If the mobile station receives a *Service Option Control Message* for Service Option 63, then, at the  
44   action time associated with the message, the mobile station shall process the message as follows:

- 1 1. If the MOBILE\_TO\_MOBILE field is equal to '1', the mobile station should disable the audio  
 2 output of the speech codec for 1 second after initialization.
- 3 If the MOBILE\_TO\_MOBILE field is equal to '0', the mobile station shall process each received  
 4 packet as described in Section 6.
- 5 2. If the INIT\_CODEEC field is equal to '1', the mobile station shall perform speech codec  
 6 initialization. The mobile station shall complete the initialization within 40 ms.
- 7 3. VMR-WB accepts as input, a mode of operation through the RATE\_REDUCE field as defined in  
 8 Table 2.4-2 using this mode of operation, VMR-WB generates Rate 1, Rate 1/2, and Rate 1/8  
 9 packets in a proportion that results in the average data rate given by Table 2.4-2. Note that Rate  
 10 1/4 is not used.

11 Service Option 63 shall continue to use these fractions until either of the following events occurs:

- 12 • The mobile station receives a *Service Option Control Message* specifying a different  
 13 RATE\_REDUCE, or
- 14 • Service Option 63 is initialized.

15  
 16 Service Option 63 was developed using reduced rate operation (encoding mode of operation) as a  
 17 network control criteria. The VMR-WB codec selects the encoding rate based upon the instantaneous  
 18 characteristics of the input speech: voiced, unvoiced, transition, etc., as well as the encoding mode  
 19 selected.

20  
 21 If the RATE\_REDUCE field is not equal to a value defined in Table 2.4-2, the mobile station shall reject  
 22 the message by sending a *Mobile Station Reject Order* with the ORDQ field set equal to '00000100'.

23 **Table 2.4-2. VMR-WB Encoding Rate Mode Control Parameters**

RATE_REDUCE (Binary)	VMR-WB Mode of Operation	Estimated Average Encoding Rate kbps (Source Encoding Rates)
'000'	4	5.7786
'001'	4 (HR Max)	3.3641
All other RATE_REDUCE values are reserved		

24 2.4.1.2.2 Base Station Requirements

25 The base station may send a *Service Option Control Message* to the mobile station. If the base  
 26 station sends a *Service Option Control Message*, the base station shall include the following type-  
 27 specific fields for Service Option 63:

28 **Table 2.4-3. Service Option Control Message Type-Specific Fields**

Field	Length (bits)
RATE_REDUCE	3
RESERVED	3
MOBILE_TO_MOBILE	1
INIT_CODEEC	1

29

- 1       RATE\_REDUCE   -   VMR-WB mode of operation.
- 2                                The base station shall set this field to the RATE\_REDUCE value from
- 3                                Table 2.4-2 corresponding to the mode of operation that the mobile
- 4                                station is to operate in.
- 5       RESERVED       -   Reserved bits.
- 6                                The base station shall set this field to '000'.
- 7       MOBILE\_TO\_MOBILE
- 8                                -   Mobile-to-mobile processing.
- 9                                If the mobile station is to perform mobile-to-mobile processing (Section
- 10                                2.4.1.2.1), the base station shall set this field to '1'. In addition, if the
- 11                                mobile station is to disable the audio output of the speech codec for 1
- 12                                second after initialization, the base station shall set the INIT\_CODEC
- 13                                field and the MOBILE\_TO\_MOBILE field to '1'. If the mobile station is
- 14                                not to perform mobile-to-mobile processing, the base station shall set
- 15                                the MOBILE\_TO\_MOBILE field to '0'.
- 16       INIT\_CODECE    -   Initialize speech codec.
- 17                                If the mobile station is to initialize the speech codec, the base station
- 18                                shall set this field to '1'. Otherwise, the base station shall set this field to
- 19                                '0'.

**Table 2.4-4. VMR-WB Encoding Rate Mode Control Parameters**

RATE_REDUCE (Binary)	VMR-WB Mode of Operation	Estimated Average Encoding Rate kbps (Source Encoding Rates)
'000'	4	5.7786
'001'	4 (HR Max)	3.3641
All other RATE_REDUCE values are reserved		

21

1

## 2 3 AUDIO INTERFACES

3

4

In general the basic audio interface characteristics of VMR-WB codec are shown in Table 3-1.

5

**Table 3-1: Basic audio interface characteristics of VMR-WB**

<b>Bandwidth</b>	50 – 7000 Hz (-3dB) @ 16 kHz sampling frequency for wideband operation 100 – 3700 Hz (-3dB) @ 8 kHz sampling frequency for narrowband operation
<b>Nominal Input Speech Level</b>	-25 dBov per ITU-T P.56 Speech Voltmeter
<b>Recommended Maximum Input Speech Level Variation</b>	-15 to -35 dBov per ITU-T P.56 Speech Voltmeter
<b>Minimum Resolution</b>	13 bit linear, 2's complement PCM
<b>Recommended SNR</b>	Greater than 10 dB

6

### 3.1 Input Audio Interface

7

8

The analog-to-digital and digital-to-analog conversion will in principle comprise the following elements:

9

10

1) Analog to uniform digital PCM

11

- Microphone;
- Input level adjustment device;
- Input anti-aliasing filter;
- Sample-hold device sampling at 16 kHz;
- Analog-to-uniform digital conversion to 16-bit representation.

12

13

14

15

16

The uniform format shall be represented in two's complement.

17

18

2) Uniform digital PCM to analog

19

- Conversion from 16-bit/16 kHz uniform PCM to analog;
- A hold device;
- Reconstruction filter including  $x/\sin(x)$  correction;
- Output level adjustment device;
- Earphone or loudspeaker.

20

21

22

23

24

25

In the terminal equipment, the A/D function may be achieved by direct conversion to 16-bit uniform PCM format; For the D/A operation, the inverse operations take place.

26

27

28

Lower uniform PCM precisions such as 14-bit should be adjusted to 16-bit representation by magnitude scaling; i.e., shifting the bits to the left and setting the LSBs to zero.

29

30

31

#### 3.1.1 Input Audio Interface in the Mobile Station

32

The input audio may be either an analog or digital signal.

### 1    **3.1.1.1       Conversion and Scaling**

2    Whether the input is analog or digital by default, the signal presented to the input of the wideband  
3    speech codec should span a frequency range of 50-7000 Hz and shall be sampled at a rate of 16000  
4    samples per second and should be quantized to a uniform PCM format with 16 bits of dynamic range.

5    The wideband codec is also capable of processing narrowband speech signals (100-3700 Hz)  
6    sampled at 8000 Hz with recommended precision of 16-bits and reproduce narrowband outputs  
7    sampled at 8000 Hz with the same precision.

8    The default input sampling rate for the encoder and the decoder of VMR-WB is 16000 Hz. As  
9    appropriate, an input/output sampling rate of 8000 Hz may be signaled to the encoder and the  
10   decoder when they are initialized.

11   The quantities in this standard assume 16-bit integer input normalization with a range from -32,768  
12   through +32,767. The following speech codec discussion assumes this 16-bit integer normalization.  
13   If an input audio interface uses a different normalization scheme, then appropriate scaling should be  
14   used.

### 15   **3.1.1.2       Digital Audio Input**

16   If the input audio is an 8-bit  $\mu$ -Law/A-Law PCM signal, it should be converted to a uniform PCM  
17   format according to Table 2 for  $\mu$ -Law and Table 1 for A-Law in ITU-T Recommendation G.711 "Pulse  
18   Code Modulation (PCM) of Voice Frequencies. After this conversion, the uniform PCM signal should  
19   be scaled in magnitude by shifting to the left by 2 bits (3 bits for A-Law) and setting the 2 (3 for A-  
20   Law) LSBs to zero to form a 16-bit integer. This will ensure normalization of the 8-bit input signal to  
21   the overload point of the 16-bit linear quantization.

### 22   **3.1.1.3       Analog Audio Input**

23   If the input is in analog form, the mobile station should sample the analog speech and should convert  
24   the samples to a digital format for speech codec processing. This may be accomplished by either the  
25   following or an equivalent method. First, the input gain audio level is adjusted. Then, the signal is  
26   bandpass filtered to prevent aliasing. Finally, the filtered signal is sampled and quantized (Section  
27   3.1.1.1).

#### 28    3.1.1.3.1    Transmit Level Adjustment

29   The mobile station should have a Send Loudness Rating (SLR) equal to  $11 \pm 3$ dB, when transmitting  
30   to a reference base station. The send loudness ratings are described in ITU-T Recommendation  
31   P.79-1999 "Calculation of loudness ratings for telephone sets".

#### 32    3.1.1.3.2    Band Pass Filtering

33   If the input speech is wideband, the signal should be filtered by a filter that generally conforms to the  
34   mask shown in ITU-T G.722 (11/1998) Figure 10/G.722, "Attenuation Distortion vs. Frequency".

35   If the input speech is narrowband, the signal should be filtered by a filter that generally conforms to  
36   the mask shown in ITU-T G.712 (11/2001) Figure 4/G.712, "Attenuation/frequency distortion for  
37   channels between a 4-wire analog port and a digital port ( $E_{4in}$  to  $T_{out}$ )".

38   The manufacturer may provide additional anti-aliasing filtering.

#### 39    3.1.1.3.3    Echo Return Loss

40   Provision shall be made to ensure adequate isolation between receive and transmit audio paths in all  
41   modes of operation. When no external transmit audio is present, the speech codec should not  
42   generate packets at rates higher than Rate 1/8 due to acoustic coupling of the receive audio into the  
43   transmit audio path (specifically with the receive audio at full volume). Minimum target levels of 45 dB

1 WAEPL should be met. See ANSI/EIA/TIA-579-A-98, *Telecommunications Telephone Terminal*  
 2 *Equipment Transmission Requirements for Digital Wireline Telephones*, Nov. 1998.

### 4 **3.1.2 Input Audio Interface in the Base Station**

#### 5 **3.1.2.1 Sampling and Format Conversion**

6 The base station converts the input speech (analog,  $\mu$ -Law/A-Law companded Pulse Code  
 7 Modulation, or other format) into a uniform quantized PCM format with 16 bits of dynamic range. The  
 8 sampling rate by default is 16000 samples per second, unless narrowband processing is desired. In  
 9 the case of narrowband input/output speech processing, the sampling rate is 8000 samples per  
 10 second. The sampling and conversion process should be as in Section 3.1.1.1.

#### 11 **3.1.2.2 Transmit Level Adjust**

12 The base station should set the transmit level so that a 1004 Hz tone at a level of 0 dBm0 at the  
 13 network interface produces a level 3.17 dB below maximum amplitude at the output of the quantizer.

#### 14 **3.1.2.3 Line Echo Canceling**

15 In case of narrowband operation, the base station should provide a method to cancel echoes returned  
 16 by the PSTN interface<sup>†</sup>. The echo canceling function should meet ITU-T G.168 requirement.

## 17 **3.2 Output Audio Interface**

### 19 **3.2.1 Output Audio Interface in the Mobile Station**

#### 20 **3.2.1.1 Band Pass Filtering**

21 If the output speech is wideband, the signal should be filtered by a filter that generally conforms to the  
 22 mask shown in ITU-T G.722 (11/1998) Figure 10/G.722, "Attenuation Distortion vs. Frequency".

23 If the output speech is narrowband, the signal should be filtered by a filter that generally conforms to  
 24 the mask shown in ITU-T G.712 (11/2001) Figure 4/G.712, "Attenuation/frequency distortion for  
 25 channels between a 4-wire analog port and a digital port ( $E_{4in}$  to  $T_{out}$ )".

26 The manufacturer may provide additional reconstruction filtering.

#### 27 **3.2.1.2 Receive Level Adjustment**

28 The mobile station should have a nominal Receive Loudness Rating (RLR) equal to  $3 \pm 3$  dB when  
 29 receiving from a reference base station. The receive loudness ratings are described in ITU-T  
 30 Recommendation P.79-1999 "Calculation of loudness ratings for telephone sets".

### 32 **3.2.2 Output Audio Interface in the Base Station**

33 Details of the digital and analog interfaces to the network are outside the scope of this document.

#### 34 **3.2.2.1 Receive Level Adjustment**

35  
 36 The base station should set the audio level so that a received 1004 Hz tone 3.17 dB below maximum  
 37 amplitude produces a level of 0 dBm0 at the network interface.

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† Because of the relatively long delays inherent in the speech coding and transmitting processes, echoes that are not sufficiently suppressed are noticeable to the mobile station user.

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## 1 4 THE VARIABLE-RATE MULTIMODE WIDEBAND SPEECH CODEC (VMR- 2 WB) – INTRODUCTION AND BIT ALLOCATION TABLES

### 3 4.1 Introduction to the VMR-WB Speech Coding Algorithm

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5 VMR-WB is a source-controlled variable-rate multimode codec designed for encoding/decoding of  
6 wideband speech, with capability of processing narrowband speech in all modes of operation. It is  
7 based on 3GPP/AMR-WB (ITU-T/G.722.2) core technology [9]. VMR-WB is fully interoperable with  
8 both standards at 12.65, 8.85\*, and 6.60\* kbps in the AMR-WB interoperable mode of operation.  
9 While support for AMR-WB codec modes 6.60 and 8.85 kbps is included in VMR-WB, due to  
10 insufficient quality of these modes, the use of these modes for voice services is not recommended.

11

12 The VMR-WB algorithm is based on AMR-WB; however, it is optimized to operate efficiently in the  
13 cdma2000 system by using a source-controlled variable-bit-rate paradigm and the addition of Half-  
14 Rate (HR), Quarter-Rate (QR), and Eighth-Rate (ER) encoding schemes that are optimized for  
15 various classes of speech signal to achieve the best subjective quality at various average data rates  
16 (ADR) applicable to cdma2000 Rate-Sets I and II.

17

18 The operation of VMR-WB is controlled by speech signal characteristics (i.e., source-controlled) and  
19 by traffic condition of the network (i.e., network-controlled). Depending on the traffic conditions, one of  
20 5 operational modes is used. Modes 0, 1, 2, and 4 are specific to CDMA systems (i.e., cdmaOne,  
21 cdma2000) with mode 0 providing the highest quality in Rate-Set II and modes 2 and 4 the lowest  
22 ADR in Rate-Set II and Rate-Set I, respectively. Mode 3 is the AMR-WB interoperable mode  
23 operating at an ADR slightly higher than Mode 0 and providing a quality equal or better than that of  
24 AMR-WB at 12.65 kbps when in an interoperable interconnection with AMR-WB at 12.65 kbps.

25

26 There are a number of various encoding types used in the VMR-WB codec. The 12.65/8.85/6.60 kbps  
27 Interoperable Full-Rate (FR) types are interoperable with AMR-WB at 12.65, 8.85, and 6.60 kbps,  
28 respectively. As an example, in the Rate-Set II Generic FR type at 13.3 kbps, the extra 13 bits (i.e.,  
29 the difference between 12.65 and 13.3 kbps) are used to enhance the codec performance in frame  
30 erasure conditions. Thus, the codec can interoperate with ITU-T G.722.2/AMR-WB at 12.65, 8.85,  
31 and 6.60 kbps without compromising the quality and performance in the cdma2000 network.

32

33 The VMR-WB coding system consists of a variable-rate multimode encoder, a decoder, and three  
34 interworking functions, each operating at bit-stream level (i.e. no decoding is necessary). The first  
35 interworking function, associated with Service Options 62 and 63, allows the use of Dim-and-Burst  
36 signaling during VMR-WB mobile-to-mobile calls (e.g., in TFO), or in other cases where a signaling  
37 request for generation of a Half Rate frame type may not be communicated to the originating VMR-  
38 WB encoder. In this case, Full-Rate frames that arrive at the multiplex sub-layer coincident with  
39 signaling requests shall be reduced from Full-Rate to Half-Rate to accommodate the signaling  
40 insertion. Frames that arrive at the multiplex sub-layer during non-signaling frames, or frames that  
41 are sufficiently low in rate to allow Dim-and-Burst signaling shall not be modified.

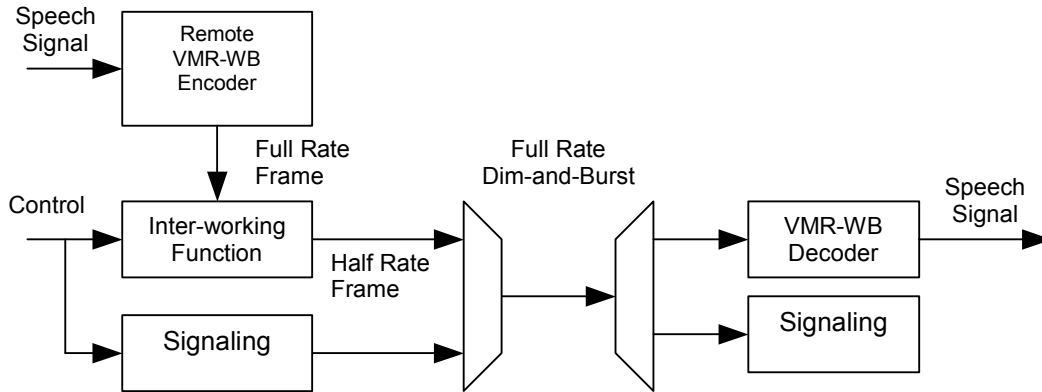
42

43 The frame rate reduction process (or “packet-level signaling”) is facilitated by the Rate-Set I and  
44 Rate-Set II Half-Rate Signaling encoding types. This is substantially a Full-Rate frame with certain  
45 parameter bit fields stripped away. This is shown in more detail later in Table 4.2-1 and Table 4.2-2.  
46 Figure 4.1-1 shows a block diagram of the use of this interworking function.

---

\* By default VMR-WB is only interoperable with AMR-WB at 12.65 kbps; however, activating compilation flag EXPANDED\_INTEROPERABILITY in the C simulation would allow interoperability with AMR-WB at 6.60 and 8.85 kbps, as well. Note that the support of EXPANDED\_INTEROPERABILITY in VMR-WB is optional.

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Figure 4.1-1: Functional block-diagram of the first Interworking Function

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The second interworking function allows for bit-stream conversion from AMR-WB at 12.65 kbps (also 8.85 or 6.60 kbps) to VMR-WB and for inserting CDMA signaling information if necessary (as above). The third interworking function enables bit-stream conversion from the VMR-WB Interoperable mode to AMR-WB at 12.65, 8.85, or 6.60 kbps.

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Depending on the speech signal characteristics in each frame and the selected operating mode, the built-in rate selection mechanism chooses a particular and permissible encoding type operating at one of the bit rates available in CDMA Rate-Set II or Rate-Set I. These rates are Full Rate (FR) at 13.3 kbps, Half Rate (HR) at 6.2 kbps, Quarter-Rate (QR) at 2.7 kbps, and Eighth Rate (ER) at 1.0 kbps in Rate-Set II and Full Rate at 8.55 kbps, Half Rate at 4.0 kbps, and Eighth Rate a 0.8 kbps. Note that Quarter Rate is not used in Rate Set I operation.

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For all operating modes, the analysis frame size is 20 ms. The encoding techniques utilized at FR or HR frames are based on the Algebraic CELP (ACELP) paradigm, while the encoding techniques used at QR or ER exploit Linear Prediction (LP) synthesis filter excited by a random noise with appropriately scaled energy. The encoding types are summarized in Table 4.1-1 and Table 4.1-2 for Rate-Set II and Rate-Set I modes, respectively.

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The VMR-WB encoder general flow chart is shown in Figure 4.1-2. The pre-processing functions comprise sampling conversion, high-pass filtering, and spectral pre-emphasis. Spectral analysis is done twice per frame and provides the energy per critical bands. The critical band energies are used for the Voice Activity Detection (VAD) and the Noise Reduction.

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2  
3**Table 4.1-1: VMR-WB encoding types and their brief description in Rate-Set II**

Encoding Types	Brief Description
Generic FR	General purpose FR codec
12.65 kbps Interoperable FR	General purpose FR codec interoperable with AMR-WB @ 12.65 kbps
8.85 kbps Interoperable FR	General purpose FR codec interoperable with AMR-WB @ 8.85 kbps
6.60 kbps Interoperable FR	General purpose FR codec interoperable with AMR-WB @ 6.60 kbps
Signaling HR	General purpose HR codec used for packet level signaling
12.65 kbps Interoperable HR	General purpose HR codec used for signaling in the 12.65 kbps AMR-WB interoperable mode
8.85 kbps Interoperable HR	General purpose HR codec used for signaling in the 8.85 kbps AMR-WB interoperable mode
6.60 kbps Interoperable HR	General purpose HR codec used for signaling in the 6.60 kbps AMR-WB interoperable mode
Generic HR	General purpose HR codec
Voiced HR	Voiced frame encoding at HR
Unvoiced HR	Unvoiced frame encoding at HR
Unvoiced QR	Unvoiced frame encoding at QR
CNG QR	Comfort noise generator for the AMR-WB interoperable mode at QR
CNG ER	Comfort noise generator at ER

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7**Table 4.1-2: VMR-WB encoding types and their brief description in Rate-Set I**

Encoding Type	Encoding Bit Rate (kbps)	Brief Description
Generic FR	8.55	General purpose FR codec
Voiced FR	8.55	Voiced frame encoding at FR
Signaling Generic HR	4.0	HR codec derived from Generic FR for packet level signaling
Signaling Voiced HR	4.0	HR codec derived from Voiced FR for packet level signaling
Generic HR	4.0	General purpose HR codec
Voiced HR	4.0	Voiced frame encoding at HR
Unvoiced HR	4.0	Unvoiced frame encoding at HR
CNG ER	0.8	Comfort noise generator at ER

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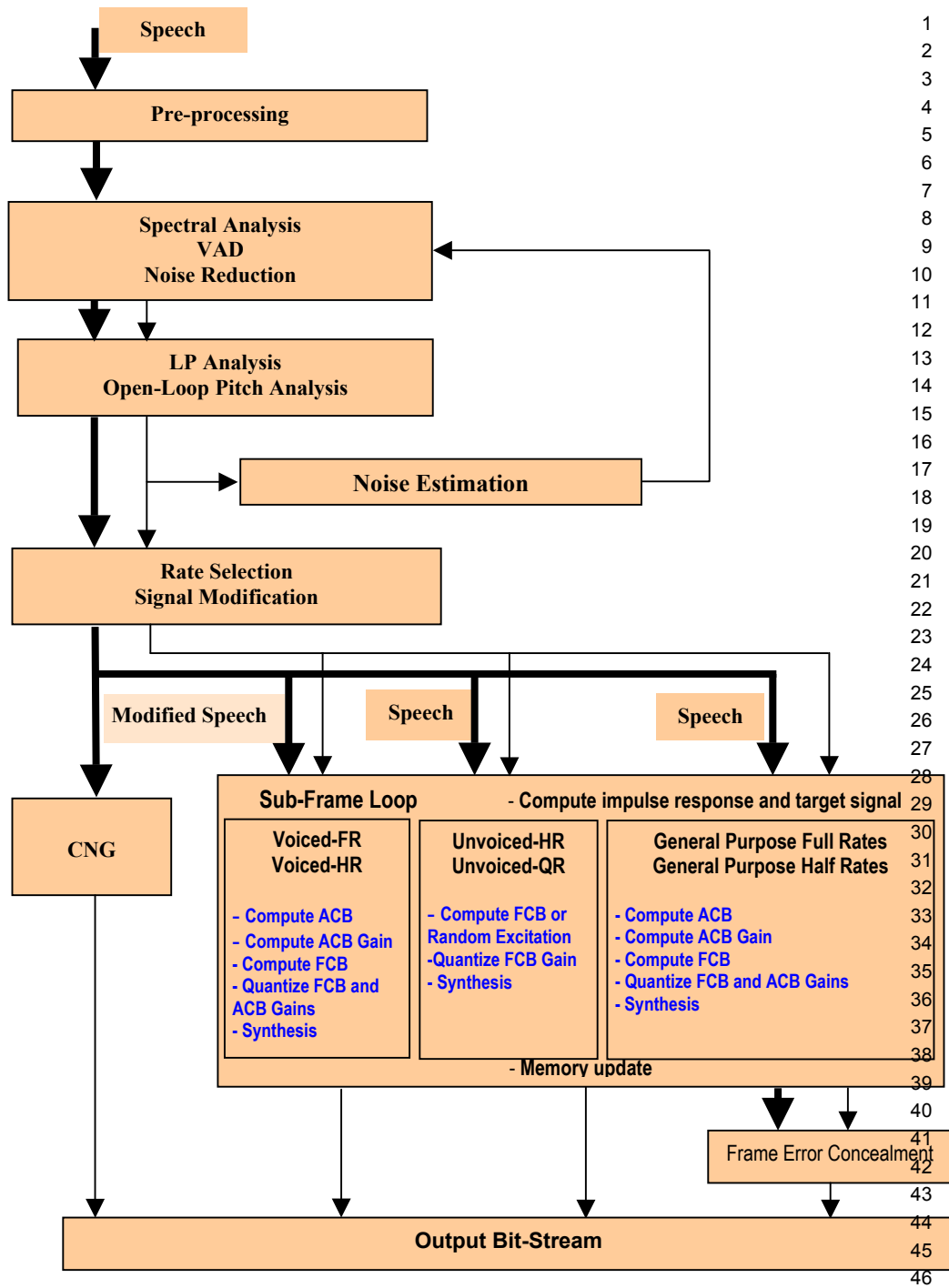
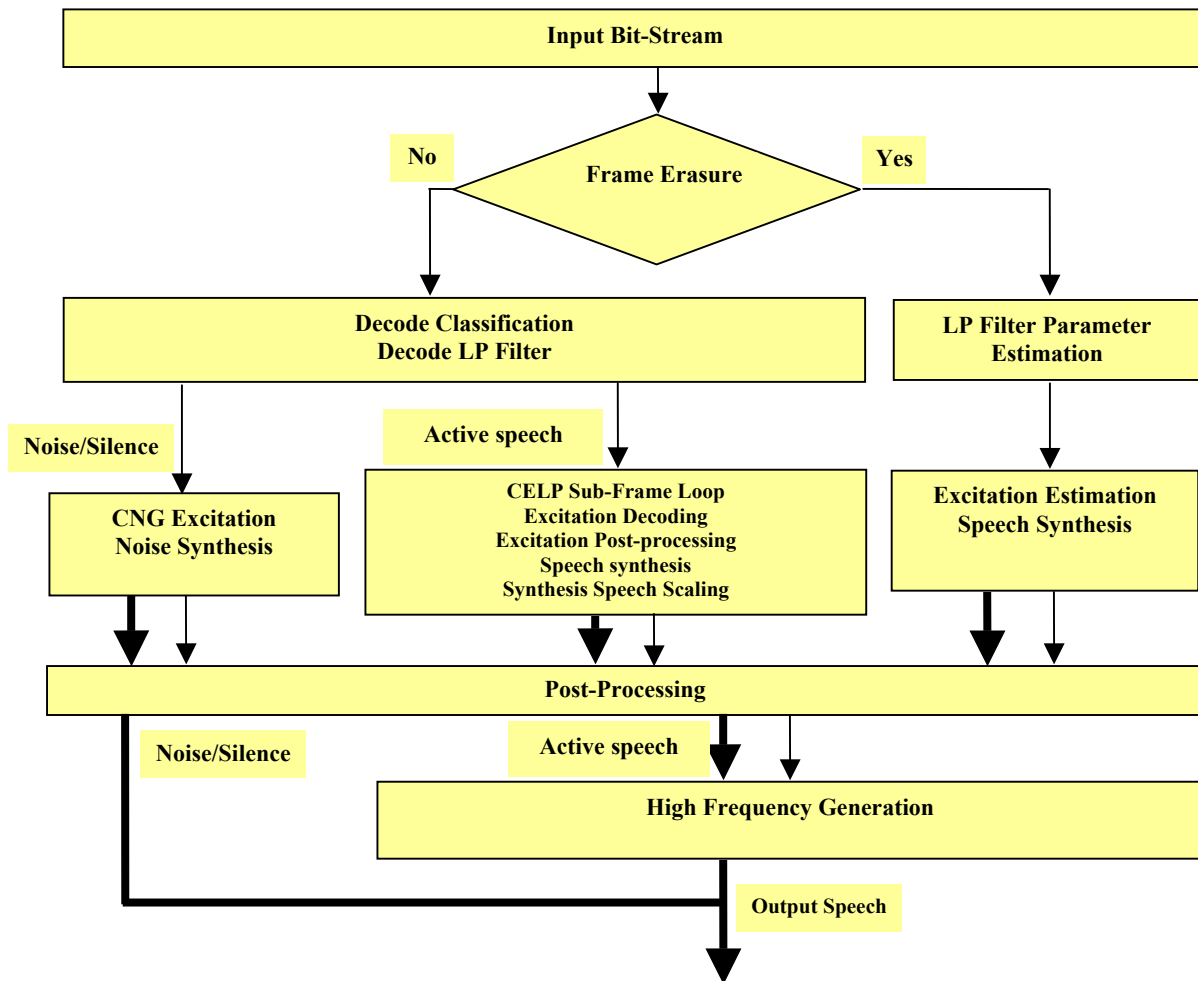


Figure 4.1-2: The VMR-WB Encoder General Flow Chart

The LP analysis is done similar to the AMR-WB standard. The open-loop pitch value is searched three times per frame using a pitch-tracking algorithm. The signal modification function modifies the original signal to make the encoding easier for the voiced encoding types. It also contains an inherent classifier for classification of those frames that are suitable for HR voiced encoding. The other encoding techniques are determined in the rate selection block.

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4 **Figure 4.1-3: The VMR-WB Decoder General Flow Chart**

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7 The active speech is then processed through the sub-frame loop using an appropriate coding  
8 technique. If the RS-II Generic FR or RS-I Voiced FR encoder are selected, supplementary  
9 information is added for better frame error concealment and recovery upon frame erasure detection at  
10 the decoder. The Comfort Noise Generation (CNG) module using CNG QR or CNG ER, depending  
11 whether VMR-WB is engaged in an AMR-WB interoperable call, encodes the inactive speech.

12

13

14 The decoder flow chart is shown in Figure 4.1-3. The erased frames following an active speech  
15 period are processed separately independent of mode or rate. The silence or noise-only frames are  
16 generated using the CNG decoder. The active speech frames are processed with conventional CELP  
17 decoding. The post-processing consists of spectral de-emphasis, sampling conversion to the output  
18 frequency, and low-frequency enhancement. Finally, high frequencies are regenerated and added to  
19 active speech frames if the output is sampled at 16 kHz.

20

21

22 The VMR-WB algorithmic delay is a fixed delay. For wideband operation, the encoder algorithmic  
23 delay is 32.8125 ms accounting for the frame length of 20 ms, 11.875 ms of lookahead and 0.9375  
24 ms of re-sampling filter delay. The decoder delay is 0.9375 ms, which comprises re-sampling filter  
25 delay. The total encoder-decoder delay for wideband operation is 33.75 ms. For narrowband  
26 operation, the encoder delay is 32.875 ms and the decoder delay is 2.1875 ms. The total encoder-

1 decoder delay for narrowband operation is 35.0635 ms. Note that the algorithmic encoding and  
 2 decoding and thereby overall algorithmic delays for Rate-Set I and Rate-Set II modes are the same.

3  
 4 As an example, the following Tables provide rate usage of VMR-WB modes and their corresponding  
 5 average data rates under clean and background noise conditions.

6 **Table 4.1-3: Rate Percentages and Average Data Rates for Clean Speech at Nominal Level**

	Mode 0	Mode 1	Mode 2	Mode 3	Mode 4
Rate 1	56.17%	36.28%	20.51%	60.11%	50.30%
Rate 1/2	3.95%	23.83%	31.22%	0	9.81%
Rate 1/4	0	0	8.38%	6.61%	0
Rate 1/8	39.89%	39.89%	39.89%	33.28%	39.89%
ADR kbps	9.0903	7.6584	6.2211	9.4934	5.7786

8  
 9 **Table 4.1-4: Rate Percentages and Average Data Rates for Speech + Street Noise at 15 dB SNR**

	Mode 0	Mode 1	Mode 2	Mode 3	Mode 4
Rate 1	54.57%	37.77%	22.55%	57.57%	45.00%
Rate 1/2	2.99%	19.80%	24.52%	0	12.56%
Rate 1/4	0	0	10.49%	7.32%	0
Rate 1/8	42.43%	42.43%	42.43%	35.11%	42.43%
ADR kbps	8.8379	7.6279	6.1545	9.1850	5.4326

10 Table 4.1-3 provides the percentages of the selected rates and the ADR for a clean sample speech  
 11 database at nominal level (-22 dB) that was generated during the VMR-WB selection process. The  
 12 voice activity for this database is about 63%; hence, that database does not represent typical  
 13 conversational speech traffic.

14 Table 4.1-4 provides the percentages of the selected rates and the ADR for speech + street noise at  
 15 15 dB signal-to-noise ratio (SNR), which was generated from the same database.

16 Note that Rate 1, Rate 1/2, Rate 1/4, Rate 1/8 for Rate-Set I mode 4 and Rate-Set II modes 0, 1, 2,  
 17 and 3 are associated with different bit rates.

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 19  
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**4.2 Bit Allocation Tables for VMR-WB Modes 0,1,2 and 3**

The general bit allocation for various encoding schemes used in the VMR-WB codec in Rate-Set II are given in Table 4.2-1.

**Table 4.2-1: Bit allocation for all encoding schemes used in the VMR-WB codec in Rate-Set II**

Parameter	Generic FR	12.65 kbps Interoperable FR	8.85 kbps Interoperable FR	6.6 kbps Interoperable FR	Signaling HR	Unvoiced HR	Generic HR
Class Info/Frame Identifier	*	13	13	13	3*	2	1
VAD Flag	0	1	1	1	0	0	0
LP Parameters	46	46	46	36	46	46	36
Pitch Delay	30	30	26	23	30	0	13
Pitch Filtering	4	4	0	0	4	0	0
Gains	28	28	24	24	28	24	26
Fixed- Codebook	144	144	80	48	0	52	48
FER protection bits	14*	0	0	0	8*	0	0
Unused bits	0		76	121	5	0	0
<b>Total</b>	<b>266</b>	<b>266</b>	<b>266</b>	<b>266</b>	<b>124</b>	<b>124</b>	<b>124</b>
Parameter	Voiced HR	12.65 kbps Interoperable HR	8.85 kbps Interoperable HR	6.6 kbps Interoperable HR	Unvoiced QR	CNG-QR	CNG-ER
Class Info/Frame Identifier	3	13	13	13	1	1	0
VAD Flag	0	1	1	1	0	0	0
LP Parameters	36	46	46	36	32	28	14
Pitch Delay	9	30	26	23	0	0	0
Pitch Filtering	2	4	0	0	0	0	0
Gains	26	28	24	24	20	6	6
Fixed- Codebook	48	2	14	27	0	0	0
FER protection bits	0	0	0	0	0	0	0
Unused bits	0	0	0	0	1	19	0
<b>Total</b>	<b>124</b>	<b>124</b>	<b>124</b>	<b>124</b>	<b>54</b>	<b>54</b>	<b>20</b>

\*Note: In Generic FR and Signaling HR encoding types, some FER protection bits are used to transmit signal energy. However, only 63 out of 64 possible combinations are used for energy encoding. The bit combination '111110' (i.e., decimal 62) is reserved and used to distinguish between different interoperable encoding types (This bit combination is part of the Class Info/Frame Identifier bits in the interoperable encoding schemes). More details are provided in Section 8.

The detailed bit allocation for various VMR-WB encoding types in Rate-Set II are given in Table 4.2-2 to Table 4.2-5.

**Table 4.2-2: Detailed bit allocation for Full-Rate encoding schemes used in VMR-WB in Rate-Set II**

Parameter	VMR-WB Encoding Type (4 sub-frames/frame)			
	Generic Full-Rate	12.65 Interoperable Full-Rate	8.85 Interoperable Full-Rate	6.60 Interoperable Full-Rate
<b>VAD Flag</b>	-	1 bit/Frame	1 bit/Frame	1 bit/Frame
<b>ISFs</b>	2 stage MA Prediction Split-VQ 1 <sup>st</sup> stage: 8+8 bits/frame 2 <sup>nd</sup> stage: 6+7+7+5+5 bits/frame	2 stage MA Prediction Split-VQ 1 <sup>st</sup> stage: 8+8 bits/frame 2 <sup>nd</sup> stage: 6+7+7+5+5 bits/frame	2 stage MA Prediction Split-VQ 1 <sup>st</sup> stage: 8+8 bits/frame 2 <sup>nd</sup> stage: 6+7+7+5+5 bits/frame	2 stage MA Prediction Split-VQ 1 <sup>st</sup> stage: 8+8 bits/frame 2 <sup>nd</sup> stage: 7+7+6 bits/frame
<b>Fixed-Codebook</b>	36 bits/sub-frame	36 bits/sub-frame	20 bits/sub-frame	12 bits/sub-frame

<b>Indices</b>				
<b>Adaptive Codebook</b>	9+6+9+6 bits/frame	9+6+9+6 bits/frame	8+5+8+5 bits/frame	8+5+5+5 bits/frame
<b>Codebook Gains</b>	Joint VQ 7 bits/sub-frame	Joint VQ 7 bits/sub-frame	Joint VQ 6 bits/sub-frame	Joint VQ 6 bits/sub-frame
<b>Pitch Filtering</b>	1 bit/sub-frame	1 bit/sub-frame	-	-
<b>FER Protection</b>	2 bits/frame for frame classification 6 bits/frame for first glottal pulse position  6 bits/frame for synthesized speech energy	-	-	-
<b>Total Bits/Frame</b>	266	266	266	266

1 **Table 4.2-3: Detailed bit allocation for Half-Rate encoding schemes used in VMR-WB in Rate-**  
2 **Set II**

Parameter	VMR-WB Encoding Type (4 sub-frames/frame)						
	Generic Half-Rate	12.65 Interoperable Half-Rate	8.85 Interoperable Half-Rate	6.60 Interoperable Half-Rate	Signaling Half-Rate	Voiced Half-Rate	Unvoiced Half-Rate
<b>Class Info</b>	1 bit/frame	3 bits/frame	3 bits/frame	3 bits/frame	3 bits/frame	3 bits/frame	2 bits/frame
<b>VAD Flag</b>	0	1 bit/Frame	1 bit/Frame	1 bit/Frame	0	0	0
<b>ISFs</b>	2 stage MA Prediction Split-VQ 1 <sup>st</sup> stage: 8+8 bits/frame 2 <sup>nd</sup> stage: 7+7+6 bits/frame	2 stage MA Prediction Split-VQ 1 <sup>st</sup> stage: 8+8 bits/frame 2 <sup>nd</sup> stage: 6+7+7+5+5 bits/frame	2 stage MA Prediction Split-VQ 1 <sup>st</sup> stage: 8+8 bits/frame 2 <sup>nd</sup> stage: 6+7+7+5+5 bits/frame	2 stage MA Prediction Split-VQ 1 <sup>st</sup> stage: 8+8 bits/frame 2 <sup>nd</sup> stage: 7+7+6 bits/frame	2 stage MA Prediction Split-VQ 1 <sup>st</sup> stage: 8+8 bits/frame 2 <sup>nd</sup> stage: 6+7+7+5+5 bits/frame	2 stage AR Prediction Split-VQ 1 <sup>st</sup> stage: 8+8 bits/frame 2 <sup>nd</sup> stage: 7+7+6 bits/frame	2 stage MA Prediction Split-VQ 1 <sup>st</sup> stage: 8+8 bits/frame 2 <sup>nd</sup> stage: 6+7+7+5+5 bits/frame
<b>Fixed-Codebook Indices</b>	12 bits/sub-frame	0	0	0	0	12 bits/sub-frame	Gaussian Codebook 13 bits/sub-frame
<b>Adaptive Codebook</b>	8+5 bits/frame	9+6+9+6 bits/frame	8+5+8+5 bits/frame	8+5+5+5 bits/frame	9+6+9+6 bits/frame	9 bits/frame	0
<b>Gain Quantization Flag</b>	1+1 bits/frame	0	0	0	0	1+1 bits/frame	0
<b>Codebook Gains</b>	Joint VQ 6 bits/sub-frame	Joint VQ 7 bits/sub-frame	Joint VQ 6 bits/sub-frame	Joint VQ 6 bits/sub-frame	Joint VQ 7 bits/sub-frame	Joint VQ 6 bits/sub-frame	Scalar Quantization 6 bits/sub-frame
<b>Pitch Filtering</b>	1+1 bits/frame	1 bit/sub-frame	0	0	1 bit/sub-frame	1+1 bits/frame	0
<b>FER Protection</b>	0	0	0	0	2 bits/frame for frame classification 6 bits/frame for synthesized speech energy	0	0
<b>Total Bits/Frame</b>	124	124	124	124	124	124	124

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**Table 4.2-4: Detailed bit allocation for Quarter-Rate encoding schemes used in VMR-WB in Rate-Set II**

Parameter	VMR-WB Encoding Type	
	CNG Quarter-Rate (1 sub-frame/frame)	Unvoiced Quarter-Rate (4 sub-frames/frame)
Class Info	0	2 bits/frame
ISFs	Single stage Split-VQ 6+6+6+5+5 bits/frame	2 stage MA Prediction Split-VQ 1 <sup>st</sup> stage: 8+8 bits/frame 2 <sup>nd</sup> stage: 5+5+6 bits/frame
Excitation Gain	6 bits/frame	5 bits/sub-frame
Total Bits/Frame	54	54

**Table 4.2-5: Detailed bit allocation for Eighth-Rate encoding scheme used in VMR-WB in Rate-Set II**

Parameter	VMR-WB Encoding Type (1 sub-frame/frame)
	CNG Eighth-Rate
ISFs	Single stage Split-VQ 5+5+4 bits/frame
Excitation Gain	6 bits/frame
Total Bits/Frame	20

**4.3 Bit Allocation Tables for VMR-WB Mode 4 in Rate-Set I**

The general bit allocation for various encoding schemes used in the VMR-WB codec in Rate-Set I are given in Table 4.3-1.

**Table 4.3-1: Bit allocation for all encoding schemes used in the VMR-WB codec in Rate-Set I**

Parameter	Generic FR	Voiced FR	Signaling Generic HR	Signaling Voiced HR	Generic HR	Voiced HR	Unvoiced HR	CNG ER
Class Info/Frame Identifier	1	2	3	3	1	3	4	-
LP Parameters	36	36	23	36	30	30	46	11
Pitch Delay	26	9	26	9	11	9	-	-
Sub-frame Selection Index	-	-	-	-	2	2	-	-
Gains	28	32	28	32	12	12	26	5
Fixed Codebook	80	80	-	-	24	24	-	-
FER protection bits	-	6	-	-	-	-	-	-
Formant shaping flag	-	-	-	-	-	-	1	-
Unused bits	-	6	-	-	-	-	3	-
<b>Total</b>	<b>171</b>	<b>171</b>	<b>80</b>	<b>80</b>	<b>80</b>	<b>80</b>	<b>80</b>	<b>16</b>

The detailed bit allocation for various VMR-WB encoding types in Rate-Set I are given in Table 4.3-2 to Table 4.3-4.

**Table 4.3-2: Detailed bit allocation for Full-Rate encoding schemes used in VMR-WB in Rate-Set I**

Parameter	VMR-WB Encoding Type (4 sub-frames/frame)	
	Generic Full-Rate	Voiced Full-Rate
Class Info	1 bit/frame	2 bits/frame
ISFs	2 stage MA Prediction Split-VQ 1 <sup>st</sup> stage: 8+8 bits/frame 2 <sup>nd</sup> stage: 7+7+6 bits/frame	2 stage AR Prediction Split-VQ 1 <sup>st</sup> stage: 8+8 bits/frame 2 <sup>nd</sup> stage: 7+7+6 bits/frame
Fixed-Codebook Indices	20 bits/sub-frame	20 bits/sub-frame

<b>Adaptive Codebook</b>	8+5+8+5 bits/frame	9 bits/frame
<b>Codebook Gains &amp; Pitch Filtering</b>	Joint VQ 7 bits/sub-frame	Joint VQ 8 bits/sub-frame
<b>FER Protection</b>	0	6 bits/frame for synthesized speech energy
<b>Total Bits/Frame</b>	171	171

1 **Table 4.3-3: Detailed bit allocation for Half-Rate encoding schemes used in VMR-WB in Rate-**  
2 **Set I**

<b>VMR-WB Encoding Type (4 sub-frames/frame)</b>					
<b>Parameter</b>	<b>Signaling Generic Half-Rate</b>	<b>Signaling Voiced Half-Rate</b>	<b>Generic Half-Rate</b>	<b>Voiced Half-Rate</b>	<b>Unvoiced Half-Rate</b>
<b>Class Info</b>	3 bits/frame	3 bits/frame	1 bit/frame	3 bits/frame	4 bits/frame
<b>ISFs</b>	2 stage MA Prediction Split-VQ 1 <sup>st</sup> stage: 8+8 bits/frame 2 <sup>nd</sup> stage (1 <sup>st</sup> 5 ISFs): 7 bits/frame	2 stage AR Prediction Split-VQ 1 <sup>st</sup> stage: 8+8 bits/frame 2 <sup>nd</sup> stage: 7+7+6 bits/frame	2 stage MA Prediction Split-VQ 1 <sup>st</sup> stage: 8+8 bits/frame 2 <sup>nd</sup> stage (1 <sup>st</sup> 9 ISFs): 7+7 bits/frame	2 stage AR Prediction Split-VQ 1 <sup>st</sup> stage: 8+8 bits/frame 2 <sup>nd</sup> stage (1 <sup>st</sup> 9 ISFs): 7+7 bits/frame	2 stage MA Prediction Split-VQ 1 <sup>st</sup> stage: 8+8 bits/frame 2 <sup>nd</sup> stage: 6+7+7+5+5 bits/frame
<b>Fixed-Codebook Indices</b>	0	0	12+12 bits/frame	12+12 bits/frame	0
<b>Adaptive Codebook</b>	8+5+8+5 bits/frame	9 bits/frame	7+4 bits/frame	9 bits/frame	0
<b>Sub-frame Selection Index</b>	0	0	1+1 bits/frame	1+1 bits/frame	0
<b>Codebook Gains</b>	Joint VQ 7 bits/sub-frame	Joint VQ 8 bits/sub-frame	Joint VQ 6+6 bits/frame	Joint VQ 6+6 bits/frame	13+13 bits/frame
<b>Pitch Filtering</b>	Included in gain quantization	Included in gain quantization	0	0	0
<b>Formant shaping flag</b>	0	0	0	0	1 bit/frame
<b>Total Bits/Frame</b>	80	80	80	80	80

3 **Table 4.3-4: Detailed bit allocation for Eighth-Rate encoding scheme used in VMR-WB in Rate-**  
4 **Set I**

<b>VMR-WB Encoding Type (1 sub-frame/frame)</b>	
<b>Parameter</b>	<b>CNG Eighth-Rate</b>
<b>ISFs</b>	Single stage Split-VQ 4+4+3 bits/frame
<b>Excitation Gain</b>	5 bits/frame
<b>Total Bits/Frame</b>	16

#### 5 **4.4 VMR-WB Symbol Table**

6 This specification uses the following symbols:  
7

<b>Symbol</b>	<b>Description</b>
$A(z)$	The inverse filter with unquantized coefficients
$\hat{A}(z)$	The inverse filter with quantized coefficients
$H(z) = \frac{1}{\hat{A}(z)}$	The speech synthesis filter with quantized coefficients
$a_i$	The unquantized linear prediction parameters (direct form coefficients)
$\hat{a}_i$	The quantified linear prediction parameters
$m$	The order of the LP model
$W(z)$	The perceptual weighting filter (unquantized coefficients)

$\gamma_1$	The perceptual weighting factor
$T$	The integer pitch lag nearest to the closed-loop fractional pitch lag of the subframe
$\beta$	The adaptive pre-filter coefficient (the quantified pitch gain)
$H_{h1}(z)$	Preprocessing high-pass filter
$w(n)$	LP analysis window
$L_1$	Length of the first part of the LP analysis window $w(n)$
$L_2$	Length of the second part of the LP analysis window $w(n)$
$r_c(k)$	The autocorrelations of the windowed speech $s'(n)$
$w_{lag}(i)$	Lag window for the autocorrelations (60 Hz bandwidth expansion)
$f_0$	The bandwidth expansion in Hz
$f_s$	The sampling frequency in Hz
$r'(k)$	The modified (bandwidth expanded) autocorrelations
$E(i)$	The prediction error in the $i$ th iteration of the Levinson algorithm
$k_i$	The $i$ th reflection coefficient
$a_j^{(i)}$	The $j$ th direct form coefficient in the $i$ th iteration of the Levinson algorithm
$F_1'(z)$	Symmetric ISF polynomial
$F_2'(z)$	Asymmetric ISF polynomial
$F_1(z)$	Polynomial $F_1'(z)$
$F_2(z)$	Polynomial $F_2'(z)$ with roots $z = 1$ and $z = -1$ eliminated
$q_i$	The Immitance spectral pairs (ISFs) in the cosine domain
$\mathbf{q}$	An ISF vector in the cosine domain
$\hat{\mathbf{q}}_i^{(n)}$	The quantified ISF vector at the $i$ th subframe of the frame $n$
$\omega_j$	The Immitance spectral frequencies (ISFs)
$T_m(x)$	A $m$ th order Chebyshev polynomial
$f_1(i), f_2(i)$	The coefficients of the polynomials $F_1(z)$ and $F_2(z)$
$f_1'(i), f_2'(i)$	The coefficients of the polynomials $F_1'(z)$ and $F_2'(z)$
$f(i)$	The coefficients of either $F_1(z)$ or $F_2(z)$
$C(x)$	Sum polynomial of the Chebyshev polynomials
$x$	Cosine of angular frequency $\omega$
$\lambda_k$	Recursion coefficients for the Chebyshev polynomial evaluation
$f_i$	The Immitance spectral frequencies (ISFs) in Hz
$\mathbf{f}^t = [f_1 f_2 \dots f_{16}]$	The vector representation of the ISFs in Hz
$\mathbf{z}(n)$	The mean-removed ISF vector at frame $n$
$\mathbf{r}(n)$	The ISF prediction residual vector at frame $n$
$\mathbf{p}(n)$	The predicted ISF vector at frame $n$
$\hat{\mathbf{r}}(n-1)$	The quantified residual vector at the past frame
$\hat{\mathbf{r}}_i^k$	The quantified ISF subvector $i$ at quantization index $k$
$d_i$	The distance between the Immitance spectral frequencies $f_{i+1}$ and $f_{i-1}$
$h(n)$	The impulse response of the weighted synthesis filter
$H(z)W(z)$	The weighted synthesis filter
$T_1$	The integer nearest to the fractional pitch lag of the previous (1st or 3rd) subframe
$s'(n)$	The windowed speech signal
$s_w(n)$	The weighted speech signal
$\hat{s}(n)$	Reconstructed speech signal
$x(n)$	The target signal for adaptive codebook search

$x_2(n), \mathbf{x}_2^t$	The target signal for algebraic codebook search
$r(n)$	The LP residual signal
$c(n)$	The fixed-codebook vector
$v(n)$	The adaptive codebook vector
$y(n) = v(n)*h(n)$	The filtered adaptive codebook vector
$y_k(n)$	The past filtered excitation
$u(n)$	The excitation signal
$\hat{u}'(n)$	The gain-scaled emphasized excitation signal
$T_{op}$	The best open-loop lag
$t_{min}$	Minimum lag search value
$t_{max}$	Maximum lag search value
$R(k)$	Correlation term to be maximized in the adaptive codebook search
$R(k)_t$	The interpolated value of $R(k)$ for the integer delay $k$ and fraction $t$
$A_k$	Correlation term to be maximized in the algebraic codebook search at index $k$
$C_k$	The correlation in the numerator of $A_k$ at index $k$
$E_{Dk}$	The energy in the denominator of $A_k$ at index $k$
$\mathbf{d} = \mathbf{H}^t \mathbf{x}_2$	The correlation between the target signal $x_2(n)$ and the impulse response $h(n)$ , i.e. backward filtered target
$\mathbf{H}$	The lower triangular Toeplitz convolution matrix with diagonal $h(0)$ and lower diagonals $h(1), \dots, h(63)$
$\Phi = \mathbf{H}^t \mathbf{H}$	The matrix of correlations of $h(n)$
$d(n)$	The elements of the vector $\mathbf{d}$
$\phi(i, j)$	The elements of the symmetric matrix $\Phi$
$c_k$	The innovation vector
$\mathbf{C}$	The correlation in the numerator of $A_k$
$m_i$	The position of the $i$ th pulse
$\mathcal{G}_i$	The amplitude of the $i$ th pulse
$N_p$	The number of pulses in the fixed-codebook excitation
$E_D$	The energy in the denominator of $A_k$
$res_{LTP}(n)$	The normalized long-term prediction residual
$b(n)$	The signal used for presetting the signs in algebraic codebook search
$s_b(n)$	The sign signal for the algebraic codebook search
$d'(n)$	Sign extended backward filtered target
$\phi'(i, j)$	The modified elements of the matrix $\Phi$ , including sign information
$\mathbf{z}^t, \mathbf{z}(n)$	The fixed-codebook vector convolved with $h(n)$
$E(n)$	The mean-removed innovation energy (in dB)
$\bar{E}$	The mean of the innovation energy
$\tilde{E}(n)$	The predicted energy
$[b_1 b_2 b_3 b_4]$	The MA prediction coefficients
$\hat{R}(k)$	The quantified prediction error at subframe $k$
$E_l$	The mean innovation energy
$R(n)$	The prediction error of the fixed-codebook gain quantization
$E_Q$	The quantization error of the fixed-codebook gain quantization
$e(n)$	The states of the synthesis filter $1/\hat{A}(z)$
$e_w(n)$	The perceptually weighted error of the analysis-by-synthesis search
$\eta$	The gain scaling factor for the emphasized excitation
$g_c$	The fixed-codebook gain
$g'_c$	The predicted fixed-codebook gain
$\hat{g}_c$	The quantified fixed-codebook gain

$g_p$	The adaptive codebook gain
$\hat{g}_p$	The quantified adaptive codebook gain
$\gamma_{gc} = g_c / g'_c$	A correction factor between the gain $g_c$ and the estimated one $g'_c$
$\hat{\gamma}_{gc}$	The optimum value for $\gamma_{gc}$
$\gamma_{sc}$	Gain scaling factor
$s'(n)$	Preprocessed signal
$s(n)$	Denosed signal
$x_w(n)$	Windowed signal for spectral analysis
$X^{(1)}(k)$	FFT of $x_w(n)$ (first spectral analysis)
$X^{(2)}(k)$	FFT of $x_w(n)$ (second spectral analysis)
$X^{(0)}(k)$	Second spectral analysis in the past frame
$E_{CB}^{(j)}(i)$	Average energy per critical band from spectral analysis j (1 or 2)
$E_{BIN}^{(j)}(i)$	Energy per bin from spectral analysis j (1 or 2)
$E_t$	Total frame energy in dB
$E_{av}(i)$	Average energy per critical band in a speech frame
$SNR_{CB}(i)$	SNR per critical band
$N_{CB}(i)$	Estimated noise energy per critical band
$SNR_{LT}$	Long term SNR
$N_{tot}$	Total noise energy per frame in dB
$\bar{E}_f$	Long term average frame energy
$\bar{N}_f$	Long term average noise energy
$\bar{E}_{CB}(i)$	Frame energy per critical band
$N_{imp}(i)$	Temporary updated noise energy
$r_e$	Noise correction factor
$NR_{max}$	Maximum allowed noise reduction in dB (default 14 dB)
$g_{dB}$	Maximum allowed noise attenuation level (in linear scale)
$g_s$	Scaling gain per critical band
$X'_R(k)$ and $X'_I(k)$	Scaled spectrum
$K_{voic}$	Number of voiced critical bands
$g_{CB,LP}$	Smoothed scaling gain used for noise reduction per critical band
$g_{BIN,LP}$	Smoothed scaling gain used for noise reduction per bin
$M_{CB}(i)$	Number of bins per critical band i
$\tilde{d}(n)$	Delay contour
$\tilde{r}(n)$	Modified residual
$\bar{d}_k$	Pitch delay at the end of present frame in signal modification
$\bar{d}_{k-1}$	Pitch delay at the end of previous frame in signal modification
$C(z)$	Harmonic Noise Weighting (HNW) filter
$g'_p$	HNW gain
$\epsilon_p$	HNW coefficient
$B_{lp}(z)$	Variable filter for RS-I adaptive codebook excitation
$v_1(n)$	Adaptive codebook vector $v(n)$

$v_2(n)$	$v(n+1) + v(n-1)$
$y_1(n)$	$v_1(n) * h(n)$
$y_2(n)$	$v_2(n) * h(n)$
$\lambda_j$	Join search weighting factor in joint algebraic codebook search in RS-I
$\tilde{\Phi}$	Modified correlation matrix in RS-I joint codebook search
$d'_c(n)$	Modified backward filtered target in RS-I joint codebook search
$G_i$	Subframe gain in NELP
$G'$	Gain normalization factor in NELP
$UVG1CB[16]$	Quantization codebook of gain normalization factor
$\hat{G}'$	Quantized gain normalization factor in NELP
$G''[i]$	Normalized gains in NELP
$UVG2CB[128][4]$	Quantization codebook for normalized gains
$\hat{G}''[i]$	Quantized normalized gains
$\hat{G}[i]$	Quantized NELP gains
$e_{rand}(n)$	Random NELP excitation vector
$e_{sparse}(i, n)$	Sparse random excitation vectors
$SF1(z)$	NELP excitation shaping filter
$flag_{F(z)}$	Formant shaping flag for Rate-Set I operation

1

2 **4.5 Abbreviations**

3

4 This specification uses the following abbreviations:

5

Abbreviation	Description
3GPP	3 <sup>rd</sup> Generation Partnership Project
3GPP2	3 <sup>rd</sup> Generation Partnership Project 2
AMR-WB	Adaptive Multi-Rate Wideband
AR	Auto Regressive
CNG	Comfort Noise Generation
ER	Eighth-Rate
FFT	Fast Fourier Transform
FIR	Finite Impulse Response
FR	Full-Rate
HNW	Harmonic Noise Weighting
HR	Half-Rate
IETF	Internet Engineering Task Force
IIR	Infinite Impulse Response
ISP	Immittance Spectral Pairs
ITU-T	International Telecommunication Union/Telecommunication Standardization Sector
LP	Linear Predictive Coding
LPC	Linear Prediction
LTP	Long-Term Prediction

MA	Moving Average
MIME	Multi-Purpose Internet Mail Extension
Modified-IRS	Modified Intermediate Response System
NELP	Noise Excited Linear Prediction
QR	Quarter-Rate
RS-I	CDMA Rate-Set I
RS-II	CDMA Rate-Set II
SID	Silence Descriptor
SNR	Signal to Noise Ratio
VAD	Voice Activity Detection
VMR-WB	Variable-Rate Multimode Wideband

1

## 5 FUNCTIONAL DESCRIPTION OF THE VMR-WB ENCODER

The VMR-WB encoder operates on 20 ms frames. The encoding procedure consists of the following:

- Preprocessing which consists of sampling conversion to 12800 samples/second, high-pass filtering and pre-emphasis.
- Spectral analysis which is used for voice activity detection and noise reduction
- Detection of narrow-band inputs
- Voice activity detection
- Noise estimation
- Noise reduction
- Linear prediction analysis, LP to ISF conversion, and interpolation.
- Computation of weighted speech signal
- Open-loop pitch analysis
- Background noise update
- Rate selection algorithm
- Signal modification and refinement of rate selection decision
- Frame encoding using the selected rate and encoding type (e.g. Rate-Set II generic FR, voiced HR, unvoiced HR, CNG-QR, CNG-ER, etc., or Rate-Set I generic FR, voiced FR, unvoiced HR, unvoiced HR, CNG-ER, etc.)

The details of the above listed encoding types will be described in the following sections. Most procedures listed above are common for both Rate-Set I and Rate-Set II operation. Procedures specific for Rate-Set I (VMR-WB mode 4) are described in separate subsections throughout the text. If for a given encoding type the Rate-Set is not mentioned, it is assumed to be Rate-Set II by default. Rate-Set I encoding types are always preceded by RS-I, e.g. RS-I generic FR, RS-I unvoiced HR, etc.

### 5.1 Pre-Processing

#### 5.1.1 Sampling Conversion

**Routine Name:** modify\_Fs

**Inputs:**

- $S_8(n)$  or  $S_{16}(n)$ : The input speech signal sampled at 8 or 16 kHz depending on the operational bandwidth, respectively.

**Outputs:**

- $s_{12.8}(n)$ : The re-sampled and filtered speech signal.

**Initialization:**

- The FIR filter memory is set to all zeros at initialization.

The linear predictive (LP) analysis, long-term prediction (LTP), and computation of fixed-codebook parameters are performed by the VMR-WB encoder at 12.8 kHz sampling rate. Therefore, the wideband input signal is decimated from 16 to 12.8 kHz. The sampling conversion is performed by first up-sampling by 4, then filtering the output through low-pass FIR filter  $H_{dec,16}(z)$  that has the cut off frequency at 6.4 kHz. Then, the signal is down-sampled by 5. The filtering delay is 15 samples at 16 kHz sampling frequency.

If the input signal sampled at 16 kHz is denoted  $s_{16}(n)$  then sampling conversion is performed as follows using a 121-tap FIR filter  $H_{dec,16}(z)$ . First, the signal is up sampled to 64 kHz by inserting three zero-valued samples between each 2 samples for each 20ms frame of 320 samples at 16 kHz sampling

1

$$s_{64}(n) = \begin{cases} s_{16}(n/4), & \text{if } n/4 = \lfloor n/4 \rfloor \\ 0, & \text{otherwise} \end{cases} \quad 0 \leq n < 1280 \quad (5.1.1-1)$$

2 where  $s_{16}(n)$  is the signal at 16 kHz sampling and  $s_{64}(n)$  is the signal at 64 kHz sampling. Then, the  
 3 signal  $s_{64}(n)$  is filtered through the filter  $H_{dec,16}(z)$  and decimated by 5 by keeping one out of 5  
 4 samples. The filter  $H_{dec,16}(z)$  is a 121-tap linear phase FIR filter having a cut-off frequency at 6.4  
 5 kHz in the 64 kHz in the up-sampled domain. The filtering and decimation can be done using the  
 6 relation  
 7

$$s_{12.8}(n) = \sum_{i=-60}^{60} s_{64}(5n+i)h_{16}(i), \quad n = 0, \dots, 255 \quad (5.1.1-2)$$

9

10 The operations in Equations (5.1.1-1) and (5.1.1-2) can be implemented in one step by using only  
 11 fourth of the filter coefficients at a time with an initial phase related to the sampling instant  $n$ . That is  
 12

$$s_{12.8}(n) = \sum_{i=-K+1}^K s_{16}(m+i)h_{16}(4i-f), \quad n = 0, \dots, 255 \quad (5.1.1-3)$$

14

15 where,  $m = \lfloor \frac{5}{4}n \rfloor$  and  $K=60/4=15$ . The initial phase  $f$  for the re-sampling ratio 4/5 is given by.

$$f = 5n - 4 \left\lfloor \frac{5}{4}n \right\rfloor = 5n - 4m \quad (\text{In this case } f = n \bmod 4).$$

17

18 The VMR-WB codec can also process narrowband input speech sampled at 8 kHz. In this case and  
 19 assuming that the input speech signal has a modified-IRS spectral characteristics, the input speech is  
 20 first filtered to compensate for the modified-IRS characteristics using a filter with the following transfer  
 21 function. Note that the VMR-WB codec has been designed to operate internally on FLAT spectrum  
 22 inputs.  
 23

$$H_{IRS\text{mod}}(z) = \frac{1 - 0.5z^{-1}}{(1 - f_{IRS\text{mod}}z^{-1})}, \quad (5.1.1-4)$$

25

26 The constant  $f_{IRS\text{mod}}$  is set to 0.95 by default. If the narrowband input speech characteristics is not  
 27 modified-IRS, the optimal performance of the codec for narrowband inputs can be tuned by changing  
 28 this constant in a range between 0.95 and 0.5 corresponding to modified-IRS and FLAT narrowband  
 29 input, respectively. The sampling conversion then consists of up-sampling from 8 kHz to 12.8 kHz.  
 30 This is performed by first up-sampling by 8, then filtering the output at 64 kHz sampling rate through a  
 31 low-pass FIR filter  $H_{dec,8}(z)$  that has the cut off frequency at 4 kHz. Then, the signal is down-  
 32 sampled by 5. A linear phase 129-tap FIR filter is used. The filtering delay is 64 samples at the 64  
 33 kHz frequency, which is equivalent to 8 samples at 8 kHz sampling frequency.  
 34

34

35 The filtering is performed similar to Equation (5.1.1-3) but with the new re-sampling ratio 8/5. That is  
 36

$$s_{64}(n) = \begin{cases} s_8(n/8), & \text{if } n/8 = \lfloor n/8 \rfloor \\ 0, & \text{otherwise} \end{cases} \quad 0 \leq n < 1280$$

38

$$s_{12.8}(n) = \sum_{i=-64}^{64} s_{64}(5n+i)h_8(i), \quad n = 0, \dots, 255$$

40

$$s_{12.8}(n) = \sum_{i=-K+1}^K s_8(m+i)h_8(8i-f), \quad n=0,\dots,255 \quad (5.1.1-5)$$

where  $m = \left\lfloor \frac{5}{8}n \right\rfloor$ ,  $K=64/8=16$ , and  $s_8(n)$  is the preprocessed signal at 8 kHz sampling. The initial

phase  $f$  for the re-sampling ratio 8/5 is given by  $f = 5n - 8 \left\lfloor \frac{5}{8}n \right\rfloor = 5n - 8m$ . The upper half of the

coefficients of the filter  $H_{dec,16}(z)$  ( $n=0$  to 60) are given as

$$h_{16}(n) = \{0.999980, 0.934870, 0.754870, 0.501632, 0.231474, -0.000000, \\ -0.152337, -0.209502, -0.181536, -0.098630, 0.000000, \\ 0.078607, 0.114831, 0.104252, 0.058760, -0.000000, \\ -0.049374, -0.073516, -0.067781, -0.038681, 0.000000, \\ 0.033082, 0.049550, 0.045881, 0.026258, -0.000000, \\ -0.022499, -0.033672, -0.031122, -0.017761, 0.000000, \\ 0.015088, 0.022452, 0.020614, 0.011674, -0.000000, \\ -0.009736, -0.014331, -0.012999, -0.007264, 0.000000, \\ 0.005872, 0.008488, 0.007546, 0.004123, -0.000000, \\ -0.003163, -0.004431, -0.003804, -0.001997, 0.000000, \\ 0.001388, 0.001829, 0.001459, 0.000702, -0.000000, \\ -0.000383, -0.000424, -0.000267, -0.000091, 0.000000\};$$

with  $h_{16}(-n) = h_{16}(n)$ ,  $n = 1, \dots, 60$ .

The upper half of the coefficients of the filter  $H_{dec,8}(z)$  ( $n=0$  to 64) are given as

$$h_8(n) = \{ \\ 0.625000, 0.608656, 0.561206, 0.487214, 0.393681, 0.289232, 0.183124, 0.084212, \\ 0.000000, -0.064103, -0.105241, -0.123231, -0.120363, -0.100899, -0.070383, -0.034841, \\ 0.000000, 0.029389, 0.050024, 0.060296, 0.060272, 0.051463, 0.036420, 0.018228, \\ 0.000000, -0.015586, -0.026613, -0.032111, -0.032070, -0.027308, -0.019240, -0.009571, \\ 0.000000, 0.008043, 0.013583, 0.016181, 0.015927, 0.013342, 0.009229, 0.004498, \\ 0.000000, -0.003604, -0.005918, -0.006835, -0.006500, -0.005240, -0.003472, -0.001613, \\ 0.000000, 0.001150, 0.001758, 0.001870, 0.001615, 0.001161, 0.000670, 0.000262, \\ 0.000000, -0.000113, -0.000116, -0.000066, -0.000018, 0.000000, 0.000000, 0.00, 0.00\};$$

with  $h_8(-n) = h_8(n)$ ,  $n = 1, \dots, 64$ .

### 5.1.2 High-Pass Filtering and Pre-emphasis

**Routine Name:** hp50, preemph

**Inputs:**

- $S_{12.8}(n)$ : The re-sampled speech signal at 12.8 kHz.

**Outputs:**

- $s'(n)$ : The high-pass filtered and pre-emphasized speech signal.

**Initialization:**

- The memory of  $H_{hp}(z)$  and  $H_{pre-emph}(z)$  filters are set to all zeros at initialization.

Following the sampling conversion stage, the re-sampled signal  $s_{12.8}(n)$  is pre-processed to obtain the signal  $s'(n)$ . These pre-processing functions are applied to the signal prior to the encoding process: high-pass filtering and pre-emphasizing.

1 A high-pass filter is used to suppress undesired low frequency components of the input signal. The  
2 transfer function of the filter with cut off frequency of 50 Hz is given by

$$3 \quad H_{hp}(z) = \frac{0.982910156 - 1.965820313z^{-1} + 0.982910156z^{-2}}{1 - 1.965820313z^{-1} + 0.966308593z^{-2}} \quad (5.1.2-1)$$

4  
5 In the pre-emphasis, a first order high-pass filter is used to emphasize higher frequencies of the input  
6 speech and it is given by

$$7 \quad H_{pre-emph}(z) = 1 - 0.68z^{-1} \quad (5.1.2-2)$$

## 8 5.2 Spectral Analysis

9  
10 **Routine Name:** `analy_sp`

11 **Inputs:**

- 12 •  $s'(n)$ : The high-pass filtered and pre-emphasized speech signal,

13 **Outputs:**

- 14 •  $X(k)$ : 256-point FFT of the pre-processed input speech
- 15 •  $E_{CB}(i)$ : Average energy in  $i$ th critical band
- 16 •  $E_{BIN}(k)$ : Energy in the  $k$ th frequency bin
- 17 •  $E_t$ : Total frame energy

18 **Initialization:**

- 19 • None

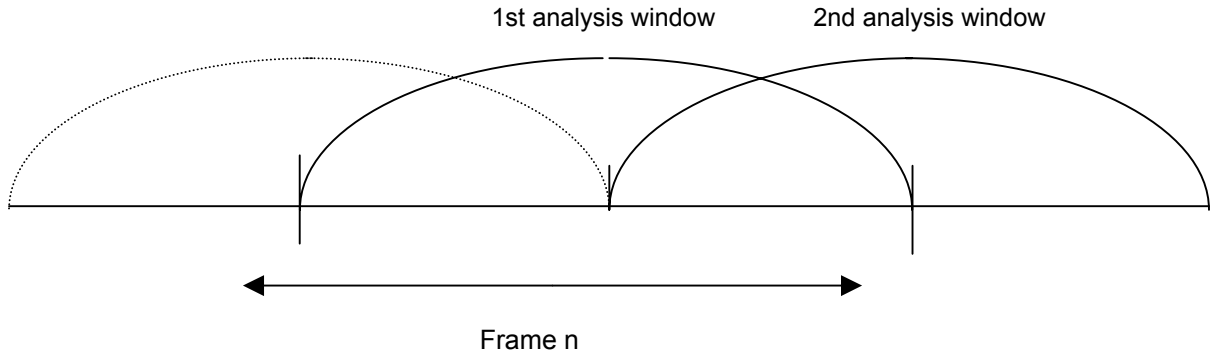
20  
21 Spectral analysis is used for the VAD, noise suppression, and signal classification functions.

22  
23 The Discrete Fourier Transform is used to perform the spectral analysis and spectral energy  
24 estimation. The frequency analysis is done twice per frame using 256-point Fast Fourier Transform  
25 (FFT) with a 50 percent overlap. The positions of the analysis windows are appropriately selected so  
26 that all lookahead information is exploited. The beginning of the first window is placed 24 samples  
27 after the beginning of the current frame. The second window is placed 128 samples farther. A square  
28 root of a Hanning window (which is equivalent to a sine window) is used to weight the input signal for  
29 the frequency analysis. This window is particularly well suited for overlap-add methods. Thus, this  
30 particular spectral analysis is used in the noise suppression algorithm based on spectral subtraction  
31 and overlap-add analysis/synthesis. (Instead of using a Hanning window at the analysis stage and a  
32 rectangular window at synthesis stage, a square-root Hanning window is used at both stages). The  
33 square root Hanning window is given by

$$34 \quad w_{FFT}(n) = \sqrt{0.5 - 0.5 \cos\left(\frac{2\pi n}{L_{FFT}}\right)} = \sin\left(\frac{\pi n}{L_{FFT}}\right), \quad n = 0, \dots, L_{FFT} - 1$$

(5.2-1)

36  
37 where  $L_{FFT}=256$  is the size of FFT analysis. Note that only half the window is computed and stored  
38 since it is symmetric (from 0 to  $L_{FFT}/2$ ).



**Figure 5.2-1: Relative positions of the spectral analysis windows**

The windowed signal for both spectral analysis are obtained as

$$\begin{aligned} x_w^{(1)}(n) &= w_{FFT}(n)s'(n+24), & n &= 0, \dots, L_{FFT} - 1 \\ x_w^{(2)}(n) &= w_{FFT}(n)s'(n+24 + L_{FFT}/2), & n &= 0, \dots, L_{FFT} - 1 \end{aligned} \quad (5.2-2)$$

where  $s'(0)$  is the first sample in the present frame. The superscript (1) and (2) used to denote the first and the second frequency analysis, respectively, are dropped for simplicity up to Equation (5.2-5).

FFT is performed on both windowed signals to obtain two sets of spectral parameters per frame:

$$X(k) = \sum_{n=0}^{N-1} x_w(n) e^{-j2\pi \frac{kn}{N}}, \quad k = 0, \dots, L_{FFT} - 1 \quad (5.2-3)$$

The output of the FFT provides the real and imaginary parts of the spectrum denoted by  $X_R(k)$ ,  $k=0$  to 128, and  $X_I(k)$ ,  $k=1$  to 127. Note that  $X_R(0)$  corresponds to the spectrum at 0 Hz (DC) and  $X_R(128)$  corresponds to the spectrum at 6400 Hz. The spectrum at these points is only real valued and usually ignored in the subsequent analysis.

After FFT analysis, the resulting spectrum is divided into critical bands [20] using the intervals having the following upper limits (20 bands in the frequency range 0-6400 Hz): Critical bands = {100.0, 200.0, 300.0, 400.0, 510.0, 630.0, 770.0, 920.0, 1080.0, 1270.0, 1480.0, 1720.0, 2000.0, 2320.0, 2700.0, 3150.0, 3700.0, 4400.0, 5300.0, 6350.0} Hz.

The 256-point FFT results in a frequency resolution of 50 Hz (i.e., 6400/128). Thus after ignoring the DC component of the spectrum, the number of frequency bins per critical band is  $M_{CB} = \{2, 2, 2, 2, 2, 2, 3, 3, 3, 4, 4, 5, 6, 6, 8, 9, 11, 14, 18, 21\}$ , respectively.

The average energy in critical band is computed as

$$E_{CB}(i) = \frac{1}{(L_{FFT}/2)^2 M_{CB}(i)} \sum_{k=0}^{M_{CB}(i)-1} (X_R^2(k+j_i) + X_I^2(k+j_i)), \quad i = 0, \dots, 19 \quad (5.2-4)$$

where  $X_R(k)$  and  $X_I(k)$  are, respectively, the real and imaginary parts of the  $k$ th frequency bin and  $j_i$  is the index of the first bin in the  $i$ th critical band given by  $j_i = \{1, 3, 5, 7, 9, 11, 13, 16, 19, 22, 26, 30, 35, 41, 47, 55, 64, 75, 89, 107\}$ .

The spectral analysis module also computes the energy per frequency bin,  $E_{BIN}(k)$ , for the first 17 critical bands (74 bins excluding the DC component)

$$E_{BIN}(k) = X_R^2(k) + X_I^2(k), \quad k = 0, \dots, 73 \quad (5.2-5)$$

Finally, the spectral analysis module computes the average total energy for both FFT analyses in a 20 ms frame by adding the average critical band energies  $E_{CB}$ . That is, the spectrum energy for a certain spectral analysis is computed as

$$E_{frame} = \sum_{i=0}^{19} E_{CB}(i) \quad (5.2-6)$$

and the total frame energy (in dB) is computed as the average of spectrum energies of both spectral analysis in a frame. That is

$$E_t = 10 \log(0.5(E_{frame}(0) + E_{frame}(1))) \quad (5.2-7)$$

The output parameters of the spectral analysis module (both spectral analyses), that is average energy per critical band, the energy per frequency bin, and the total energy, are used in VAD, noise reduction, and rate selection modules.

Note that for narrowband inputs sampled at 8000 samples/second, after sampling conversion to 12800 samples/second, there is no content at both ends of the spectrum, thus the first lower frequency critical band as well as the last three high frequency bands are not considered in the computation of output parameters (only bands from  $i=1$  to 16 are considered).

### 5.2.1 Detection of narrowband inputs

When processing wideband input speech sampled at 16 kHz and following the sampling conversion stage, the spectrum up to 6400 Hz is considered in the spectral analysis. Thus the minimum and maximum critical bands are set to  $b_{min}=0$  and  $b_{max}=19$ . However, in case of narrowband inputs sampled at 8 kHz, after sampling conversion to 12.8 kHz, the maximum and minimum bands are set to  $b_{min}=1$  and  $b_{max}=16$ . This implies that the spectrum below 150 Hz and above 3700 Hz is not considered in spectral analysis. For the cases where the input speech is over-sampled at 16 kHz while the input is actually a narrowband signal, a simple detection algorithm is applied to detect such signals and avoid processing them as wideband signals.

The detection is based on computing the smoothed energy in the upper two critical bands and comparing it to a certain threshold.

The mean energy in the upper two bands in a frame is given by

$$E_{18,19} = 0.25(E_{CB}^{(1)}(18) + E_{CB}^{(1)}(19) + E_{CB}^{(2)}(18) + E_{CB}^{(2)}(19)) \quad (5.2.1-1)$$

where  $E_{CB}^{(1)}(i)$  and  $E_{CB}^{(2)}(i)$  are the critical band energies from both spectral analysis. The smoothed energy in the upper two bands is then computed as

$$\bar{E}_{18,19} = 0.99\bar{E}_{18,19} + 0.01E_{18,19} \quad (5.2.1-2)$$

1 with initial value  $\bar{E}_{18,19} = 2$ .

2

3 The decision for the minimum and maximum used critical bands  $b_{min}$  and  $b_{max}$  is made as follows  
4 (using hysteresis):

5

6 If ( $\bar{E}_{18,19} < 1$ ) then  $b_{min}=1$  and  $b_{max}=16$

7 Else If ( $\bar{E}_{18,19} > 2$ ) then  $b_{min}=0$  and  $b_{max}=19$

8

9 This operation is performed only if the input signal is sampled at 16 kHz and if the mean energy in  
10 bands 15 and 16 (computed similar to Equation (5.2.1-1)) is higher than 4.

### 11 5.3 Voice Activity Detection

12

13 **Routine Name:** `wb_vad`

14 **Inputs:**

- 15 •  $E_{CB}(i)$ : Average energy in  $i$ th critical band
- 16 •  $N_{CB}(i)$ : Noise estimate in  $i$ th critical band
- 17 •  $\bar{E}_f$ : Long-term average frame energy
- 18 •  $\bar{N}_f$ : Long-term average noise energy

19 **Outputs:**

- 20 • VAD\_flag and local VAD flag

21 **Initialization:**

- 22 •  $\bar{E}_f$  is initialized to 45 dB.  $\bar{N}_f$  is initialized to total noise energy  $N_{tot}$ .  $N_{tot}$  initialization is  
23 provided in Section 5.4. VAD internal parameters of active speech counter and VAD  
24 hangover counter are initialized to 3 and 0, respectively.

25

26 The spectral analysis described in Section 5.2 is performed twice per frame. Let  $E_{CB}^{(1)}(i)$  and  $E_{CB}^{(2)}(i)$   
27 denote the energy per critical band for the first and second spectral analysis, respectively (as  
28 computed in Equation (5.2-4)). The average energy per critical band for the whole frame and part of  
29 the previous frame is computed as

30

$$31 \quad E_{av}(i) = 0.2E_{CB}^{(0)}(i) + 0.4E_{CB}^{(1)}(i) + 0.4E_{CB}^{(2)}(i) \quad (5.3-1)$$

32

33 where  $E_{CB}^{(0)}(i)$  denote the energy per critical band from the second analysis of the previous frame.

34 The signal-to-noise ratio (SNR) per critical band is then computed as

35

$$36 \quad SNR_{CB}(i) = E_{av}(i) / N_{CB}(i) \quad \text{Constrained by } SNR_{CB} \geq 1. \quad (5.3-2)$$

37

38 where  $N_{CB}(i)$  is the estimated noise energy per critical band as will be explained in Section 5.4.2.

39 The average SNR per frame, in dB, is then computed as

40

$$41 \quad SNR_{av} = 10 \log \left( \sum_{i=b_{min}}^{b_{max}} SNR_{CB}(i) \right), \quad (5.3-3)$$

42

1 where  $b_{min}=0$  and  $b_{max}=19$  in case of wideband signals, and  $b_{min}=1$  and  $b_{max}=16$  in case of  
 2 narrowband signals.

3  
 4 The voice activity is detected by comparing the average SNR per frame to a certain threshold, which  
 5 is a function of the long-term SNR. The long-term SNR is given by

$$6 \quad SNR_{LT} = \bar{E}_f - \bar{N}_f \quad (5.3-4)$$

7  
 8  
 9 where  $\bar{E}_f$  and  $\bar{N}_f$  are computed using equations (5.4.3-1) and (5.4.3-2), respectively. The initial  
 10 value of  $\bar{E}_f$  is 45 dB.

11  
 12 The threshold is a piecewise linear function of the long-term SNR. Different functions are used for  
 13 narrowband signals in Rate-Set I, for narrowband signals in Rate-Set II, and for wideband signals.

14  
 15 For wideband signals, if  $SNR_{LT} < 35$  (noisy speech) then

$$16 \quad th_{VAD} = 0.4346 SNR_{LT} + 13.9575$$

17 Else (clean speech)

$$18 \quad th_{VAD} = 1.0333 SNR_{LT} - 7$$

19 For narrowband signals in Rate-Set II, if  $SNR_{LT} < 29.5$  (noisy speech) then

$$20 \quad th_{VAD} = 0.313 SNR_{LT} + 14.6$$

21 Else (clean speech)

$$22 \quad th_{VAD} = 1.0333 SNR_{LT} - 7$$

23  
 24 For narrowband signals in Rate-Set I, the long-term average noise energy  $\bar{N}_f$  is first lower-  
 25 limited to 2 dB prior to the computation of  $SNR_{LT}$  (equation 5.3-4). If  $SNR_{LT} < 20$  (noisy  
 26 speech) then

$$27 \quad th_{VAD} = 0.435 SNR_{LT} + 15.8$$

28  
 29 Else (clean speech)

$$30 \quad th_{VAD} = 0.62 SNR_{LT} + 12.1$$

31  
 32  
 33 Further, a hysteresis in the VAD decision is added to prevent frequent switching at the end of an  
 34 active speech period. It is applied when the frame is in a soft hangover period or if the last frame is an  
 35 active speech frame. The soft hangover period consists of the first 10 frames after each active  
 36 speech interval longer than 2 consecutive frames. For narrowband signals in Rate-Set I, the  
 37 hysteresis decrements the VAD decision threshold only if  $SNR_{LT} > 0$ , using the formula

$$38 \quad th_{VAD} = th_{VAD} - 0.32 SNR_{LT} \quad (5.3-5)$$

39  
 40  
 41 The decrement is upper-bounded to 11. For wideband signals or Rate-Set II operation, the hysteresis  
 42 is applied as follows.

43  
 44 In case of noisy speech ( $SNR_{LT} < 35$ ) the hysteresis decreases the VAD decision threshold by

$$45 \quad th_{VAD} = 0.95 th_{VAD} \quad (5.3-6)$$

46  
 47  
 48 In case of clean speech, the hysteresis decrements the VAD decision threshold by

$$th_{VAD} = th_{VAD} - 11 \quad (5.3-7)$$

If the average SNR per frame is larger than the VAD decision threshold, that is, if  $SNR_{av} > th_{VAD}$ , then the frame is declared as an active speech frame and the VAD flag and a local VAD flag are set to 1. Otherwise the VAD flag and the local VAD flag are set to 0. However, in case of noisy speech, the VAD flag is forced to 1 in hard hangover frames, i.e. one or two inactive frames following a speech period longer than 2 consecutive frames. (The local VAD flag is then equal to 0 but the VAD flag is set to 1).

## 5.4 Primary Noise Parameter Estimation and Update

**Routine Name:** noise\_est\_down, long\_enr, correlation\_shift

**Inputs:**

- $E_{CB}(i)$ : Average energy in  $i$ th critical band
- $N_{CB}(i)$ : Noise estimate in  $i$ th critical band

**Outputs:**

- $N_{tot}$ : Total noise energy
- $E_{rel}$ : Relative frame energy
- $\bar{N}_f$ : Long-term average noise energy
- $\bar{E}_f$ : Long-term average frame energy
- $N_{CB}(i)$ : Noise estimate in  $i$ th critical band
- $r_e$ : Noise correction factor

**Initialization:**

- $N_{CB}(i)$  is initialized to 0.03 unless otherwise described in the following section.  $N_{tot}$  is computed as a  $10\log$  of the sum of  $N_{CB}(i)$ .

In this section, the total noise energy, relative frame energy, update of long-term average noise energy and long-term average frame energy, average energy per critical band, and a noise correction factor are computed. Furthermore, noise energy initialization and update are described.

### 5.4.1 Total noise and Relative Frame Energy Estimation

The total noise energy per frame is computed as follows:

$$N_{tot} = 10 \log \left( \sum_{i=0}^{19} N_{CB}(i) \right) \quad (5.4.1-1)$$

where  $N_{CB}(i)$  is the estimated noise energy per critical band.

The relative energy of the frame is calculated by the difference between the frame energy in dB and the long-term average energy. The relative frame energy is given by

$$E_{rel} = E_t - \bar{E}_f \quad (5.4.1-2)$$

1 where  $E_t$  is given in Equation (5.2-7) and  $\bar{E}_f$  is given in Equation (5.4.3-1).

#### 2 3 **5.4.2 Frame Energy per Critical Band, Noise Initialization, and Noise Update**

4  
5 The frame energy per critical band for the entire frame is computed by averaging the energies from  
6 both spectral analyses in the frame. That is,

$$7 \quad \bar{E}_{CB}(i) = 0.5E_{CB}^{(1)}(i) + 0.5E_{CB}^{(2)}(i) \quad (5.4.2-1)$$

8  
9  
10 The noise energy per critical band  $N_{CB}(i)$  is usually initialized to 0.03. However, in the first 5  
11 subframes, if the signal energy is not too high or if the signal does not have strong high frequency  
12 components, then the noise energy is initialized using the energy per critical band so that the noise  
13 reduction algorithm can efficiently function from the beginning of the process. Two high frequency  
14 ratios are computed,  $r_{15,16}$ , as the ratio between the average energy of critical bands 15 and 16 and  
15 the average energy in the first 10 bands (i.e., mean of both spectral analyses), and  $r_{18,19}$ , as the ratio  
16 between the average energy of critical bands 18 and 19 and the average energy in the first 10 bands.

17  
18 In the first 5 frames, if  $E_t < 49$  and  $r_{15,16} < 2$  and  $r_{18,19} < 1.5$  then for the first 3 frames, then

$$19 \quad N_{CB}(i) = \bar{E}_{CB}(i), \quad i = 0, \dots, 19 \quad (5.4.2-2)$$

20  
21 and for the following two frames  $N_{CB}(i)$  is updated by

$$22 \quad N_{CB}(i) = 0.33N_{CB}(i) + 0.66\bar{E}_{CB}(i), \quad i = 0, \dots, 19 \quad (5.4.2-3)$$

23  
24  
25 For the following frames, only noise energy update is performed for the critical bands where the  
26 signal energy is less than background noise energy. First, the temporary updated noise energy is  
27 computed as

$$28 \quad N_{tmp}(i) = 0.9N_{CB}(i) + 0.1(0.25E_{CB}^{(0)}(i) + 0.75\bar{E}_{CB}(i)) \quad (5.4.2-4)$$

29  
30  
31 where  $E_{CB}^{(0)}(i)$  corresponds to the second spectral analysis from the previous frame. Then the critical  
32 bands with their energy less than the background noise energy are updated as

$$33 \quad \text{For } i=0 \text{ to } 19, \text{ if } N_{tmp}(i) < N_{CB}(i) \text{ then } N_{CB}(i) = N_{tmp}(i)$$

34  
35  
36  
37 A second level of noise update is performed later in Section 5.9 by setting  $N_{CB}(i) = N_{tmp}(i)$ , if the  
38 frame is declared as inactive speech frame. The noise energy update has been divided into two  
39 stages because the noise parameter update can be performed only during inactive speech frames  
40 (i.e., silence intervals) and all the necessary parameters for the speech activity detection are thereby  
41 needed. These parameters are, however, dependent on LP analysis and open-loop pitch analysis,  
42 which are performed on the denoised speech signal. For the noise reduction algorithm in order to  
43 have as accurate noise estimate as possible, the noise estimation is thus updated by lowering the  
44 noise estimate, if necessary, before the noise reduction is performed and later by increasing the noise  
45 estimate, if necessary during silence intervals. Therefore, the noise level update is not increased in  
46 the first stage and the update can be done independent of the speech activity.

#### 47 48 **5.4.3 Long-Term Average Noise Energy and Frame Energy Update**

1 Either the long-term average noise energy or the long-term average frame energy is updated in every  
 2 frame. In case of active speech frames (VAD\_flag = 1), the long-term average frame energy is  
 3 updated as follows:

$$4 \quad \bar{E}_f = 0.99\bar{E}_f + 0.01E_t \quad (5.4.3-1)$$

6 with initial value  $\bar{E}_f = 45dB$ . In case of inactive speech frames (VAD flag = 0), the long-term  
 7 average noise energy is updated by

$$8 \quad \bar{N}_f = 0.99\bar{N}_f + 0.01N_{tot} \quad (5.4.3-2)$$

10 The initial value of  $\bar{N}_f$  is set equal to  $N_{tot}$  for the first 4 frames. Furthermore, in the first 4 frames,  
 11 the value of  $\bar{E}_f$  is constrained by  $\bar{E}_f \geq \bar{N}_{tot} + 10$ .

#### 12 5.4.4 Noise Correction Factor

13 A correction factor  $r_e$  is added to the normalized correlation in order to compensate for the decrease  
 14 of normalized correlation in the presence of background noise. It has been found that the  
 15 dependence between this decrease  $r_e$  and the total background noise energy in dB is approximately  
 16 exponential and can be estimated using following equation:

$$17 \quad r_e = 0.00024492 e^{0.1596 (N_{tot} - g_{dB})} - 0.022 \quad \text{constrained by } r_e \geq 0. \quad (5.4.4-1)$$

18 where  $g_{dB}$  is the maximum allowed noise attenuation level in dB, which is set to 14 dB by default. It  
 19 should be noted that under normal operation of the noise reduction algorithm  $g_{dB}$  is sufficiently high  
 20 and thereby  $r_e$  is practically zero. It is only relevant when the noise reduction is disabled or if the  
 21 background noise level is significantly higher than the maximum attenuation level.

## 22 5.5 Noise Suppression

23 **Routine Name:** noise\_sup

24 **Inputs:**

- 25 •  $E_{CB}(i)$ : Average energy per critical band
- 26 •  $E_{BIN}(k)$ : Energy per frequency bin
- 27 •  $N_{CB}(i)$ : Noise estimate in  $i$ th critical band
- 28 •  $X(k)$ : 256-point FFT of the pre-processed input speech
- 29 • VAD\_flag and local VAD flag
- 30 •  $K_{voic}$ : Voiced critical bands

31 **Outputs:**

- 32 •  $s(n)$ : The denoised speech signal

33 **Initialization:**

- 34 • The noise suppression buffers are initially set to zero. The number of voiced critical bands is  
 35 initialized to 0. The smoothed scaling gains  $g_{BIN,LP}(k)$  and  $g_{CB,LP}(i)$  are initially set to 1.

36 Noise reduction is performed in the spectral domain. The denoised signal is then reconstructed using  
 37 the overlap and add method. The reduction is performed by scaling the spectrum in each critical band

1 with a scaling gain constrained between  $g_{min}$  and 1, which is derived from the signal-to-noise ratio  
 2 (SNR) in that critical band. For frequencies lower than a certain frequency, the processing is  
 3 performed on frequency bin basis and not on critical band basis. Thus, a scaling gain is applied on  
 4 every frequency bin derived from the SNR in that bin (i.e., the SNR is computed using the bin energy  
 5 divided by the noise energy of the critical band including that bin). This feature preserves the energy  
 6 at frequencies close to harmonics and further prevents distortion while strongly reducing the noise  
 7 between the harmonics. This technique is only utilized for voiced speech and, given the frequency  
 8 resolution of the frequency analysis used, for speech segments with relatively short pitch period.  
 9 These are scenarios where the noise between harmonics is most perceptible.

10  
 11 The minimum scaling gain  $g_{min}$  is derived from the maximum allowed noise reduction in dB,  $g_{dB}$ . The  
 12 maximum allowed reduction is an input parameter to the noise suppression unit with default value of  
 13 14 dB. Thus minimum-scaling gain is given by

$$14 \quad g_{min} = 10^{-g_{dB}/20} \quad (5.5-1)$$

15  
 16 and it is equal to 0.19953 for the default value of 14 dB.

17  
 18 In case of inactive speech frames with  $VAD\_flag=0$ , the same scaling is applied over the entire  
 19 spectrum and is given by  $g_s = 0.9g_{min}$  if noise suppression is activated (if  $g_{min}$  is lower than 1). That  
 20 is, the scaled real and imaginary components of the spectrum are given by

$$21 \quad X'_R(k) = g_s X_R(k), \quad k = 1, \dots, 128, \quad \text{and} \quad X'_I(k) = g_s X_I(k), \quad k = 1, \dots, 127. \quad (5.5-2)$$

22  
 23 Note that for narrowband input speech, the maximum value of index  $k$  in Equation (5.5-2) is set to 79  
 24 (up to 3950 Hz). For purpose of noise suppression, the input signal is considered as narrowband only  
 25 if the input sampling frequency is equal to 8000 Hz; i.e. in this case, the narrowband detector decision  
 26 described in Section 5.2.1 is disregarded.

27  
 28 For active speech frames, the scaling gain is computed related to the SNR per critical band or per bin  
 29 for the first voiced bands. If  $K_{VOIC} > 0$  then per bin noise suppression is performed on the first  
 30  $K_{VOIC}$  bands. Per band noise suppression is used on the rest of the bands. In case  $K_{VOIC} = 0$  per  
 31 band noise suppression is used on the entire spectrum. The value of  $K_{VOIC}$  is updated as described  
 32 in Section 5.9.1. The maximum value of  $K_{VOIC}$  is 17; therefore, per bin processing can be applied  
 33 only on the first 17 critical bands corresponding to a maximum frequency of 3700 Hz. The maximum  
 34 number of bins for which per bin processing can be used is 74 (the number of bins in the first 17  
 35 bands). An exception is considered for hard hangover frames that will be described later in this  
 36 Section.

37  
 38 The scaling gain in a certain critical band, or for a certain frequency bin, is computed as a function of  
 39 SNR and given by

$$40 \quad (g_s)^2 = k_s SNR + c_s, \quad \text{constrained by} \quad g_{min} \leq g_s \leq 1 \quad (5.5-3)$$

41  
 42 The values of  $k_s$  and  $c_s$  are determined such that  $g_s = g_{min}$  for  $SNR = 1$ , and  $g_s = 1$  for  $SNR =$   
 43 45. That is, for  $SNR \leq 1$ , the scaling is limited to  $g_s$  and for  $SNR \geq 45$ ; no noise suppression is  
 44 performed in the given critical band ( $g_s = 1$ ). Note that the  $SNR$  is expressed as a ratio of energies.  
 45 Thus, considering these two end points, the values of  $k_s$  and  $c_s$  in Equation (5.5-3) are given by

$$k_s = (1 - g_{\min}^2) / 44 \quad \text{and} \quad c_s = (45g_{\min}^2 - 1) / 44. \quad (5.5-4)$$

The variable  $SNR$  in Equation (5.5-3) is either the SNR per critical band,  $SNR_{CB}(i)$ , or the SNR per frequency bin,  $SNR_{BIN}(k)$ , depending on the type of processing.

The SNR per critical band, corresponding to the first spectral analysis of the frame, is computed as follows:

$$SNR_{CB}(i) = \frac{0.2E_{CB}^{(0)}(i) + 0.6E_{CB}^{(1)}(i) + 0.2E_{CB}^{(2)}(i)}{N_{CB}(i)} \quad i = 0, \dots, 19 \quad (5.5-5)$$

and the SNR per critical band, corresponding to the second spectral analysis of the frame, is computed as follows:

$$SNR_{CB}(i) = \frac{0.4E_{CB}^{(1)}(i) + 0.6E_{CB}^{(2)}(i)}{N_{CB}(i)} \quad i = 0, \dots, 19 \quad (5.5-6)$$

where  $E_{CB}^{(1)}(i)$  and  $E_{CB}^{(2)}(i)$  denote the energy per critical band for the first and second spectral analysis, respectively (as computed in Equation (5.2-4)),  $E_{CB}^{(0)}(i)$  denotes the energy per critical band from the second spectral analysis of the previous frame, and  $N_{CB}(i)$  denotes the noise energy estimate per critical band (see Section 5.9 for the update of  $N_{CB}(i)$ ).

The SNR per critical bin in the  $i$ th critical band, corresponding to the first spectral analysis of the frame, is computed as

$$SNR_{BIN}(k) = \frac{0.2E_{BIN}^{(0)}(k) + 0.6E_{BIN}^{(1)}(k) + 0.2E_{BIN}^{(2)}(k)}{N_{CB}(i)}, \quad k = j_i, \dots, j_i + M_{CB}(i) - 1 \quad (5.5-7)$$

and for the second spectral analysis, the SNR is computed as

$$SNR_{BIN}(k) = \frac{0.4E_{BIN}^{(1)}(k) + 0.6E_{BIN}^{(2)}(k)}{N_{CB}(i)}, \quad k = j_i, \dots, j_i + M_{CB}(i) - 1 \quad (5.5-8)$$

where  $E_{BIN}^{(1)}(k)$  and  $E_{BIN}^{(2)}(k)$  denote the energy per frequency bin for the first and second spectral analysis, respectively (as computed in Equation (5.2-5)),  $E_{BIN}^{(0)}(k)$  denote the energy per frequency bin from the second spectral analysis of the previous frame,  $N_{CB}(i)$  denote the noise energy estimate per critical band,  $j_i$  is the index of the first bin in the  $i$ th critical band and  $M_{CB}(i)$  is the number of bins in critical band  $i$  defined in Section 5.2.

In case of per critical band processing for a frequency band with index  $i$ , after determining the scaling gain as in Equation (5.5-5), and using SNR as defined in Equations (5.5-5) or (5.5-6), the actual scaling is performed using a smoothed scaling gain updated in every frequency analysis as

$$g_{CB,LP}(i) = \alpha_{gs} g_{CB,LP}(i) + (1 - \alpha_{gs}) g_s \quad (5.5-9)$$

1 where the smoothing factor is inversely proportional to the gain and is given by  $\alpha_{gs} = 1 - g_s$ . That  
 2 implies the smoothing is stronger for smaller gains  $g_s$ . The scaling in the critical band is performed  
 3 as

$$\begin{aligned}
 X'_R(k + j_i) &= g_{CB,LP}(i) X_R(k + j_i), \quad \text{and} \\
 X'_I(k + j_i) &= g_{CB,LP}(i) X_I(k + j_i), \quad k = 0, \dots, M_{CB}(i) - 1,
 \end{aligned}
 \tag{5.5-10}$$

6  
 7 where  $j_i$  is the index of the first bin in the critical band  $i$  and  $M_{CB}(i)$  is the number of bins in that  
 8 critical band.

9  
 10 In case of per bin processing in a frequency band with index  $i$ , after determining the scaling gain as in  
 11 Equation (5.5-3), and using SNR as defined in Equations (5.5-7) or (5.5-8), the actual scaling is  
 12 performed using a smoothed scaling gain updated in every frequency analysis which is calculated as  
 13 follows:

$$g_{BIN,LP}(k) = \alpha_{gs} g_{BIN,LP}(k) + (1 - \alpha_{gs}) g_s \tag{5.5-11}$$

16  
 17 where  $\alpha_{gs} = 1 - g_s$  similar to Equation (5.5-9).

18 While temporal smoothing of the gains prevents audible energy oscillations, control of the smoothing  
 19 using  $\alpha_{gs}$  prevents distortion in high-SNR speech segments preceded by low-SNR speech frames, as  
 20 it is the case for voiced onsets for example. The scaling in the  $i$ th critical band is performed as

$$\begin{aligned}
 X'_R(k + j_i) &= g_{BIN,LP}(k + j_i) X_R(k + j_i), \quad \text{and} \\
 X'_I(k + j_i) &= g_{BIN,LP}(k + j_i) X_I(k + j_i), \quad k = 0, \dots, M_{CB}(i) - 1,
 \end{aligned}
 \tag{5.5-12}$$

23  
 24 where  $j_i$  is the index of the first bin in the critical band  $i$  and  $M_{CB}(i)$  is the number of bins in that  
 25 critical band. The smoothed scaling gains  $g_{BIN,LP}(k)$  and  $g_{CB,LP}(i)$  are initially set to 1. Each time  
 26 an inactive speech frame is processed (VAD\_flag=0), the values of the smoothed gains are reset to  
 27  $g_{min}$  as defined in Equation (5.5-1).

28  
 29 As mentioned above, if  $K_{VOIC} > 0$  per bin, noise suppression is performed on the first  $K_{VOIC}$   
 30 frequency bands, and per band noise suppression is performed on the remaining frequency bands  
 31 using the procedures described above. Note that in every spectral analysis, the smoothed scaling  
 32 gains  $g_{CB,LP}(i)$  are updated for all critical bands (even for voiced bands processed with per bin  
 33 processing - in this case  $g_{CB,LP}(i)$  is updated with an average of  $g_{BIN,LP}(k)$  associated with the  
 34 band  $i$ ). Similarly, scaling gains  $g_{BIN,LP}(k)$  are updated for all frequency bins in the first 17 bands (up  
 35 to bin 74). For frequency bands processed with per band processing, they are updated by setting  
 36 them equal to  $g_{CB,LP}(i)$  in these 17 specific bands.

37  
 38 Note that for clean speech, noise suppression is not performed in active speech frames  
 39 (VAD\_flag=1). This is detected by finding the maximum noise energy in all critical bands,  
 40  $\max(N_{CB}(i))$ ,  $i = 0, \dots, 19$ , and if this value is less or equal 15 then no noise suppression is  
 41 performed.

42  
 43 As mentioned above, for inactive speech frames (VAD\_flag=0), a scaling of  $0.9 g_{min}$  is applied over  
 44 the entire spectrum, which is equivalent to removing a constant noise floor. For VAD short-hangover

frames (VAD\_flag=1 and local\_VAD=0), per band processing is applied to the first 10 bands as described above (corresponding to 1700 Hz), and for the rest of the spectrum, a constant noise floor is subtracted by scaling the rest of the spectrum by a constant value  $g_{\min}$ . This measure reduces significantly high frequency noise energy oscillations. For these bands above the 10<sup>th</sup> band, the smoothed scaling gains  $g_{CB,LP}(i)$  are not reset but updated using Equation (5.5-9) with  $g_s = g_{\min}$  and the per bin smoothed scaling gains  $g_{BIN,LP}(k)$  are updated by setting them equal to  $g_{CB,LP}(i)$  in the corresponding critical bands.

In case of processing of narrowband speech signals (up-sampled to 12800 Hz), the noise suppression is performed on the first 17 bands (up to 3700 Hz). For the remaining 5 frequency bins between 3700 Hz and 4000 Hz, the spectrum is scaled using the last scaling gain  $g_s$  at the bin at 3700 Hz. For the remaining of the spectrum (from 4000 Hz to 6400 Hz), the spectrum is zeroed.

### 5.5.1 Reconstruction of Denoised Signal

After calculation of the scaled spectral components,  $X'_R(k)$  and  $X'_I(k)$ , inverse FFT of the scaled spectrum is taken to obtain the windowed denoised signal in the time domain.

$$x_{w,d}(n) = \frac{1}{N} \sum_{k=0}^{N-1} X(k) e^{j2\pi \frac{kn}{N}}, \quad n = 0, \dots, L_{FFT} - 1 \quad (5.5.1-1)$$

This is repeated for both spectral analyses in the frame to obtain the denoised windowed signals  $x_{w,d}^{(1)}(n)$  and  $x_{w,d}^{(2)}(n)$ . For every half frame, the signal is reconstructed using an overlap-add scheme for the overlapping portions of the analysis. Since a square root Hanning window is used on the original signal prior to spectral analysis, the same window is applied at the output of the inverse FFT prior to overlap-add operation. Thus, the doubled windowed denoised signal is given by

$$\begin{aligned} x_{ww,d}^{(1)}(n) &= w_{FFT}(n) x_{w,d}^{(1)}(n), \quad n = 0, \dots, L_{FFT} - 1 \\ x_{ww,d}^{(2)}(n) &= w_{FFT}(n) x_{w,d}^{(2)}(n), \quad n = 0, \dots, L_{FFT} - 1 \end{aligned} \quad (5.5.1-2)$$

For the first half of the analysis window, the overlap-add scheme for constructing the denoised signal is formulated as

$$s(n + 24) = x_{ww,d}^{(0)}(n + L_{FFT} / 2) + x_{ww,d}^{(1)}(n), \quad n = 0, \dots, L_{FFT} / 2 - 1 \quad (5.5.1-3)$$

and for the second half of the analysis window, the overlap-add operation for constructing the denoised signal is as follows:

$$s(n + 24 + L_{FFT} / 2) = x_{ww,d}^{(1)}(n + L_{FFT} / 2) + x_{ww,d}^{(2)}(n), \quad n = 0, \dots, L_{FFT} / 2 - 1 \quad (5.5.1-4)$$

where  $x_{ww,d}^{(0)}(n)$  is the double windowed denoised signal from the second analysis in the previous frame.

Note that with the overlap-add scheme described above, the denoised signal can be reconstructed up to 24 samples from the lookahead in addition to the present frame. However, another 128 samples are still needed to complete the lookahead needed for LP analysis and open-loop pitch analysis. This part is temporary obtained by inverse windowing the second half of the denoised windowed signal  $x_{w,d}^{(2)}(n)$  without performing overlap-add operation. That is

$$s(n + 24 + L_{FFT}) = x_{w,d}^{(2)}(n + L_{FFT}/2) / w_{FFT}^2(n + L_{FFT}/2), \quad n = 0, \dots, L_{FFT}/2 - 1$$

(5.5.1-5)

Note that this portion of the signal is properly re-computed in the next frame using the overlap-add method as described above. In the following sections, the denoised signal  $s(n)$  is used as the input signal.

## 5.6 Linear Prediction Analysis and ISP Conversion

**Routine Name:** `analy_lp`

**Inputs:**

- $s(n)$ : The denoised speech signal
- $q_i^{(n-1)}$ : The imittance spectral pairs of previous frame

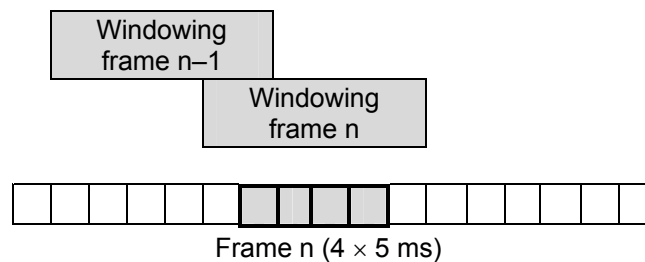
**Outputs:**

- $a_i$ : LP filter coefficients
- $q_i^{(n)}$ : The imittance spectral pairs for current frame
- $E(i)$ : LP residual energies

**Initialization:**

- $q_i^{(n-1)}$  are initialized such that corresponding ISFs are equi-distant.

Short-term prediction, or LP analysis is performed once per speech frame using the autocorrelation approach with 30 ms asymmetric windows. An overhead of 5 ms is used in the autocorrelation computation. The frame structure is depicted below.



**Figure 5.6-1: Relative positions and length of the LP analysis windows**

The autocorrelation of windowed speech is converted to the LP coefficients using the Levinson-Durbin algorithm. Then the LP coefficients are transformed to the Imittance Spectral Pairs (ISP) domain for quantization and interpolation purposes. The interpolated quantized and unquantized filters are converted back to the LP filter coefficients (to construct the synthesis and weighting filters for each subframe).

### 5.6.1 Windowing and Autocorrelation Computation

The LP analysis is performed once per frame using an asymmetric window. The window is centered at the fourth subframe and it consists of two parts: the first part is a half of a Hamming window and the second part is a quarter of a cosine cycle. The window is given by

$$w(n) = 0.54 - 0.46 \cos\left(\frac{2\pi n}{2L_1 - 1}\right), \quad n = 0, \dots, L_1 - 1$$

$$= \cos\left(\frac{2\pi(n - L_1)}{4L_2 - 1}\right), \quad n = L_1, \dots, L_1 + L_2 - 1$$
(5.6.1-1)

where the values  $L_1=256$  and  $L_2=128$  are used.

The autocorrelations of the windowed speech  $s''(n) = s'(n-64)w(n)$ ,  $n = 0, \dots, 383$  are computed by

$$r_c(k) = \sum_{n=k}^{383} s''(n) s''(n-k), \quad k = 0, \dots, 16$$
(5.6.1-2)

and a 60 Hz bandwidth expansion is used by lag windowing the autocorrelations using the window

$$w_{lag}(i) = \exp\left[-\frac{1}{2}\left(\frac{2\pi f_0 i}{f_s}\right)^2\right], \quad i = 1, \dots, 16$$
(5.6.1-3)

where  $f_0 = 60$  Hz is the bandwidth expansion,  $f_s = 12800$  Hz is the sampling frequency, and  $r_c(0) \geq 100$ . Furthermore, for wideband input signals,  $r_c(0)$  is multiplied by the white noise correction factor 1.0001 which is equivalent to adding a noise floor at  $-40$  dB. In case of narrowband inputs,  $r_c(0)$  is multiplied by a stronger factor of 1.0031 to ease the LP coefficients estimation on a spectrum with a sharp cut-off frequency at the ends of the narrowband spectrum.

### 5.6.2 Levinson-Durbin Algorithm

The modified autocorrelations  $r'(0) = 1.0001r_c(0)$  or  $r'(0) = 1.0031r_c(0)$  and  $r'(k) = r_c(k)w_{lag}(k)$ ,  $k = 1, \dots, 16$ , are used to obtain the LP filter coefficients  $a_k$ ,  $k = 1, \dots, 16$  by solving the set of equations.

$$\sum_{k=1}^{16} a_k r'(i-k) = -r'(i), \quad i = 1, \dots, 16$$
(5.6.2-1)

The set of equations in (5.6.2-1) is solved using the Levinson-Durbin algorithm [22]. This algorithm uses the following recursion:

$$E(0) = r'(0)$$

For  $i = 1$  to 16 do

$$k_i = -\left[r'(i) + \sum_{j=1}^{i-1} a_j^{i-1} r'(i-j)\right] / E(i-1)$$

$$a_i^{(i)} = k_i \quad (5.6.2-2)$$

For  $j = 1$  to  $i-1$  do

$$a_j^{(i)} = a_j^{(i-1)} + k_i a_{i-j}^{(i-1)}$$

$$E(i) = (1 - k_i^2) E(i-1)$$

The final solution is given as  $a_j = a_j^{(16)}$ ,  $j = 1, \dots, 16$ . The LP filter coefficients are converted to the ISP representation [23] for quantization and interpolation purposes. The conversions to the ISP domain and back to the LP filter domain are described in following sections.

### 5.6.3 LP to ISP conversion

The LP filter coefficients  $a_k$ ,  $k = 1, \dots, 16$  are converted to the ISP representation for quantization and interpolation purposes. For a 16th order LP filter, the ISPs are defined as the roots of the sum and difference polynomials

$$f_1'(z) = A(z) + z^{-16} A(z^{-1}) \quad (5.6.3-1)$$

and

$$f_2'(z) = A(z) - z^{-16} A(z^{-1}) \quad (5.6.3-2)$$

respectively. (The polynomials  $f_1'(z)$  and  $f_2'(z)$  are symmetric and asymmetric, respectively). It can be proven that all roots of these polynomials are on the unit circle and interlaced. Polynomial  $f_2'(z)$  has two roots at  $z = 1$  ( $\omega = 0$ ) and  $z = -1$  ( $\omega = \pi$ ). To eliminate these two roots, we define the new polynomials

$$f_1(z) = f_1'(z) \quad (5.6.3-3)$$

and

$$f_2(z) = f_2'(z) / (1 - z^{-2}) \quad (5.6.3-4)$$

Polynomials  $f_1(z)$  and  $f_2(z)$  have 8 and 7 conjugate roots on the unit circle  $(e^{\pm j\omega_i})$ , respectively. Therefore, the polynomials can be written as

$$F_1(z) = (1 + a_{16}) \prod_{i=0,2,\dots,14} (1 - 2q_i z^{-1} + z^{-2}) \quad (5.6.3-5)$$

and

$$F_2(z) = (1 + a_{16}) \prod_{i=1,3,\dots,13} (1 - 2q_i z^{-1} + z^{-2}) \quad (5.6.3-6)$$

where  $q_i = \cos(\omega_i)$  with  $\omega_i$  being the Immitance Spectral Frequencies (ISF) and  $a_{16}$  is the last predictor coefficient. The ISFs satisfy the ordering property  $0 < \omega_0 < \omega_1 < \dots < \omega_{14} < \pi$ . We refer to  $q_i$  as the ISPs in the cosine domain and  $\omega_i$  as the ISPs in the frequency domain.

1 Since both polynomials  $f_1(z)$  and  $f_2(z)$  are symmetric only the first 8 and 7 coefficients of each  
 2 polynomial, respectively, and the last predictor coefficient need to be computed. The coefficients of  
 3 these polynomials are found by the following recursive relations

$$4 \quad f_1(8) = 2a_8$$

5 for  $i = 0$  to 7

$$6 \quad f_1(i) = a_i + a_{m-i}$$

$$7 \quad f_2(i) = a_i - a_{m-i} + f_2(i-2)$$

(5.6.3-7)

8 where  $m = 16$  is the predictor order, and  $f_2(-2) = f_2(-1) = 0$ .

9  
 10 The ISPs are found by evaluating the polynomials  $F_1(z)$  and  $F_2(z)$  at 100 points equally spaced  
 11 between 0 and  $\pi$  and checking for sign changes. A sign change signifies the existence of a root and  
 12 the sign change interval is then divided 4 times to more precisely track the root. The Chebyshev  
 13 polynomials are used to evaluate  $F_1(z)$  and  $F_2(z)$  [24]. In this method the roots are found directly in  
 14 the cosine domain  $\{q_i\}$ . The polynomials  $F_1(z)$  and  $F_2(z)$  evaluated at  $z = e^{j\omega}$  can be written as

$$15 \quad F_1(\omega) = 2e^{-j8\omega}C_1(x) \text{ and } F_2(\omega) = 2e^{-j7\omega}C_2(x) \quad (5.6.3-8)$$

16 with

$$17 \quad C_1(x) = \sum_{i=0}^7 f_1(i)T_{8-i}(x) + f_1(8)/2, \text{ and } C_2(x) = \sum_{i=0}^6 f_2(i)T_{8-i}(x) + f_2(7)/2 \quad (5.6.3-9)$$

18  
 19 where  $T_m = \cos(m\omega)$  is the  $m$ th order Chebyshev polynomial,  $f(i)$  are the coefficients of either  $F_1(z)$  or  
 20  $F_2(z)$ , computed using the equations in (5.6.3-7). The polynomial  $C(x)$  is evaluated at a certain value  
 21 of  $x = \cos(\omega)$  using the recursive expression

for  $k = n_f - 1$  down to 1

$$22 \quad b_k = 2xb_{k+1} - b_{k+2} + f(n_f - k)$$

(5.6.3-10)

end

$$C(x) = xb_1 - b_2 + f(n_f)/2$$

23  
 24 where  $n_f = 8$  in case of  $C_1(x)$  and  $n_f = 7$  in case of  $C_2(x)$ , with initial values  $b_{n_f} = f(0)$  and  $b_{n_f+1} = 0$ . The  
 25 detail of the Chebyshev polynomial evaluation method can be found in [24].

#### 26 5.6.4 ISP to LP Conversion

27  
 28  
 29 Once the ISPs are quantized and interpolated, the quantized and unquantized ISPs are converted  
 30 back to the LP coefficient domain  $\{a_k\}$ . The conversion to the LP domain is done as follows. The  
 31 coefficients of  $F_1(z)$  and  $F_2(z)$  are found by expanding Equations (5.6.3-5) and (5.6.3-6) knowing the  
 32 quantized and interpolated ISPs  $q_i, i = 0, \dots, m-1$ , where  $m = 16$ . The following recursive relation is  
 33 used to compute  $f_1(z)$

for  $i = 2$  to  $m/2$

$$f_1(i) = -2q_{2i-2}f_1(i-1) + 2f_1(i-2)$$

for  $j = i-1$  down to 2

$$f_1(j) = f_1(j) - 2q_{2i-2}f_1(j-1) + f_1(j-2)$$

end

$$f_1(1) = f_1(1) - 2q_{2i-2}$$

end

$$(5.6.4-1)$$

3

4 with initial values  $f_1(0) = 1$  and  $f_1(1) = -2q_0$ . The coefficients  $f_2(i)$  are computed similarly by replacing  
5  $q_{2i-2}$  by  $q_{2i-1}$  and  $m/2$  by  $m/2-1$ , and with initial conditions  $f_2(0) = 1$  and  $f_2(1) = -2q_1$ .

6

7 Once the coefficients  $f_1(z)$  and  $f_2(z)$  are found,  $F_2(z)$  is multiplied by  $1-z^{-2}$ , to obtain  $F_2'(z)$ ; that is

$$\begin{aligned} f_2'(i) &= f_2(i) - f_2(i-2), & i = 2, \dots, m/2-1 \\ f_1'(i) &= f_1(i) & i = 0, \dots, m/2 \end{aligned} \quad (5.6.4-2)$$

9

10 Then  $F_1'(z)$  and  $F_2'(z)$  are multiplied by  $1+q_{m-1}$  and  $1-q_{m-1}$ , respectively. That is

$$\begin{aligned} f_2'(i) &= (1-q_{m-1})f_2'(i), & i = 0, \dots, m/2-1 \\ f_1'(i) &= (1+q_{m-1})f_1'(i) & i = 0, \dots, m/2 \end{aligned} \quad (5.6.4-3)$$

12

13 Finally the LP coefficients are found by

$$\begin{aligned} a_i &= 0.5f_1'(i) + 0.5f_2'(i), & i = 1, \dots, m/2-1 \\ &0.5f_1'(i) - 0.5f_2'(i), & i = m/2+1, \dots, m-1 \\ &0.5f_1'(m/2), & i = m/2 \\ &q_{m-1}, & i = m \end{aligned} \quad (5.6.4-4)$$

15

16 This is directly derived from the equation  $A(z) = (F_1'(z) + F_2'(z))/2$ , and considering the fact that  
17  $F_1'(z)$  and  $F_2'(z)$  are symmetric and asymmetric polynomials, respectively.

18

### 19 5.6.5 Interpolation of ISPs

20

21 The set of LP parameters is used for the fourth subframe whereas the first, second, and third  
22 subframes use a linear interpolation of the parameters in the adjacent frames. The interpolation is  
23 performed on the ISPs in the  $\mathbf{q}$  domain. Let  $\mathbf{q}_4^{(n)}$  be the ISP vector at the 4th subframe of the frame,  
24 and  $\mathbf{q}_4^{(n-1)}$  the ISP vector at the 4th subframe of the past frame  $n-1$ . The interpolated ISP vectors at  
25 the 1st, 2nd, and 3rd subframes are given by

$$\begin{aligned} \mathbf{q}_1^{(n)} &= 0.55\mathbf{q}_4^{(n-1)} + 0.45\mathbf{q}_4^{(n)} \\ \mathbf{q}_2^{(n)} &= 0.2\mathbf{q}_4^{(n-1)} + 0.8\mathbf{q}_4^{(n)} \\ \mathbf{q}_3^{(n)} &= 0.04\mathbf{q}_4^{(n-1)} + 0.96\mathbf{q}_4^{(n)} \end{aligned} \quad (5.6.5-1)$$

26

1 The same formula is used for interpolation of both quantized and unquantized ISPs. The interpolated  
 2 ISP vectors are used to compute a different LP filter at each subframe (both quantized and  
 3 unquantized) using the ISP to LP conversion method described in 5.6.4.

## 4 5.7 Perceptual Weighting

5

6 **Routine Name:** `find_wsp`

7 **Inputs:**

- 8 •  $s(n)$ : The denoised input speech
- 9 •  $a_j$ : The LP filter coefficients

10 **Outputs:**

- 11 •  $s_w(n)$ : The perceptually weighted speech signal

12 **Initialization:**

- 13 • The memory of the filter is set to all zeros at initialization.

14

15 The encoding parameters such as adaptive codebook delay and gain, fixed-codebook index and gain  
 16 are searched by minimizing the error between the input signal and synthesized signal in a  
 17 perceptually weighted domain. Perceptual weighting is performed by filtering the signal through a  
 18 perceptual weighting filter derived from the LP synthesis filter coefficients. The perceptual weighted  
 19 signal is also used in open-loop pitch analysis and signal modification modules. The traditional  
 20 perceptual weighting filter  $W(z) = A(z/\gamma_1)/A(z/\gamma_2)$  has inherent limitations in modeling the  
 21 formant structure and the required spectral tilt concurrently.

22

23 The spectral tilt is more pronounced in wideband signals due to the wide dynamic range between low  
 24 and high frequencies. A solution to this problem is to introduce a pre-emphasis filter at the input for  
 25 filtering the wideband signal to produce a pre-emphasized signal with enhanced high frequency  
 26 content, calculate the LP synthesis filter coefficients from the pre-emphasized signal  $s(n)$ , and  
 27 produce the perceptually weighted speech signal by filtering the pre-emphasized signal through a  
 28 perceptual weighting filter having a transfer function derived from the LP filter coefficients and with a  
 29 denominator having fixed coefficients similar to the pre-emphasis filter so that the weighting of the  
 30 wideband signal in the formant regions is decoupled from the spectral tilt of the wideband signal as  
 31 will be shown below.

32

33 A weighting filter of the form

34

$$35 \quad W(z) = A(z/\gamma_1)H_{\text{de-emph}}(z) = A(z/\gamma_1)/(1 - \beta_1 z^{-1}) \quad (5.7-1)$$

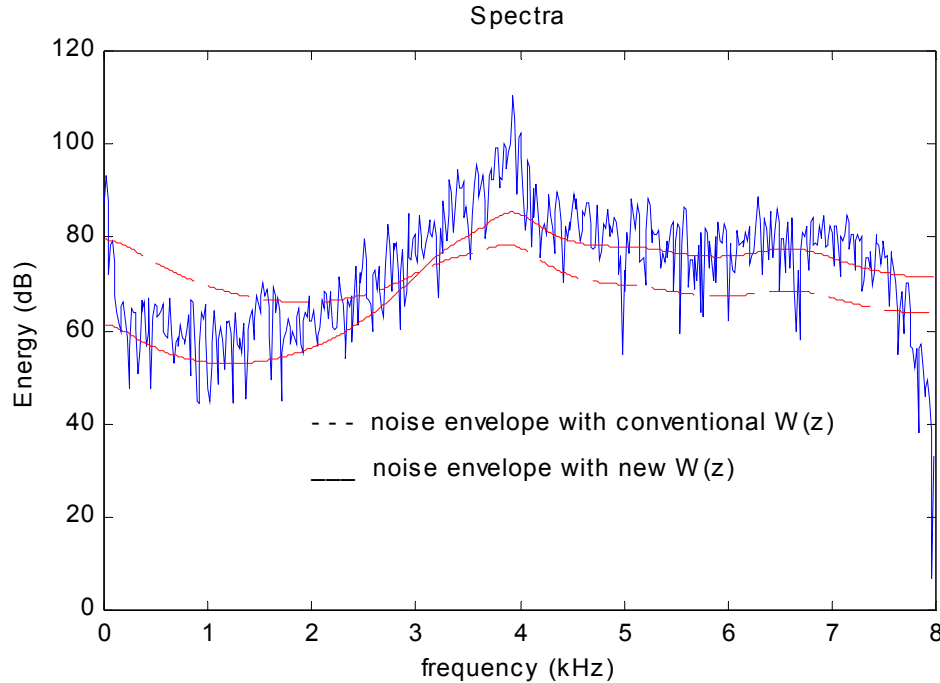
36

37 is used, where  $H_{\text{de-emph}} = \frac{1}{1 - \beta_1 z^{-1}}$  and  $\beta_1$  is fixed and equal to 0.68.

38

39 Because  $A(z)$  is computed based on the pre-emphasized speech signal  $s(n)$ , the tilt of the filter  
 40  $1/A(z/\gamma_1)$  is less pronounced compared to the case when  $A(z)$  is computed based on the original  
 41 speech (as the pre-emphasized signal itself exhibit less spectral tilt than the original wideband signal).  
 42 Since de-emphasis is performed at the decoder end, it can be shown that the quantization error  
 43 spectrum is shaped by a filter having a transfer function  $W^{-1}(z)H_{\text{de-emph}}(z) = 1/A(z/\gamma_1)$ . Thus, the  
 44 spectrum of the quantization error is shaped by a filter whose transfer function is  $1/A(z/\gamma_1)$ , with  $A(z)$   
 45 computed based on the pre-emphasized speech signal. Figure 5.7-1 compares the noise shaping of  
 46 the traditional and new weighting filters in case of an unvoiced speech segment. In the traditional  
 47 case, the filter is given by  $W(z) = A(z/0.9)/A(z/0.6)$  where  $A(z)$  is computed using the original signal  
 48 without pre-emphasis. The proposed filter is given by  $W(z) = A(z/0.9)/(1 - 0.68z^{-1})$  where  $A(z)$  is  
 49 computed using the original signal after pre-emphasis with  $1 - 0.68z^{-1}$ . In the traditional case, the  
 spectrum of the coding noise is shaped using the filter  $1/W(z) = A(z/0.6)/A(z/0.9)$  while in this case, it

1 is shaped using the filter  $1/A(z/0.9)$ . Also in this case, the weighting filter shapes the coding noise in a  
 2 way that it follows better the original speech spectrum. This is shown in Figure 5.7-1 where the  
 3 traditional filter fails to shape the noise properly at low frequencies (about 15 dB difference between  
 4 the two filters at low frequencies). This better noise shaping reflects the decoupling of the weighting  
 5 from the spectral tilt.



6  
 7 **Figure 5.7-1: Spectrum of an unvoiced signal along with the quantization noise envelope after**  
 8 **shaping by the conventional weighting filter and the new weighting filter.**

9  
 10 The perceptual weighting is performed for the 20 ms frame while updating the LP filter coefficients on  
 11 a 5-ms subframe basis using the interpolated filter parameters. In a subframe of size  $L$ , the weighted  
 12 speech is given by

$$13 \quad s_w(n) = s(n) + \sum_{i=1}^{16} a_i \gamma_1^i s(n-i) + \beta_1 s_w(n-1), \quad n = 0, \dots, L-1 \quad (5.7-2)$$

14  
 15 Furthermore, for the open-loop pitch analysis, the computation is extended for a period of 10 ms  
 16 using the lookahead from the future frame. This is done using the filter coefficients of the 4<sup>th</sup> subframe  
 17 in the present frame. Note that this extended weighted signal is used only in the analysis of the  
 18 present frame.

## 19 5.8 Open-loop Pitch Analysis and Pitch Tracking

20  
 21 **Routine Name:** `pitch_ol`

22 **Inputs:**

- 23 •  $s_w(n)$ : The weighted speech signal
- 24 •  $r_e$ : Noise correction factor
- 25 •  $E_{rel}$ : Relative frame energy

26 **Outputs:**

- 27 •  $d_0, d_1,$  and  $d_2$ : The pitch lags in each half-frame

- $C_{norm}(d)$ : The normalized correlation at pitch lags  $d_0$ ,  $d_1$ , and  $d_2$ .

#### Initialization:

- The buffers and internal memories are set to zero at initialization.

Open-loop pitch analysis calculates three estimates of the pitch lag for each frame. This is done in order to smooth the pitch evolution contour and to simplify the pitch analysis by confining the closed-loop pitch search of generic encoding types to a small number of lags around the open-loop estimated lags. In RCELP-based voiced encoding types, the open-loop pitch lags are used for the initialization of the signal modification algorithm.

Open-loop pitch estimation is based on the weighted speech signal  $s_w(n)$  computed as in Equation (5.7-2). The open-loop pitch analysis is performed on the weighted signal decimated by two. The decimated signal is obtained by filtering  $s_w(n)$  through a 5<sup>th</sup> order FIR filter with coefficients {0.13, 0.23, 0.28, 0.23, 0.13} and then down-sampling the output by 2 to obtain the decimated weighted signal  $s_{wd}(n)$ .

Open-loop pitch analysis is performed 3 times per frame every 10 ms to find three estimates of the pitch lag: two in the present frame and one for the lookahead.

### 5.8.1 Correlation Function Computation

The pitch delay range (decimated by 2) is divided into the following four segments: [10,16], [17,31], [32, 61], and [62,115]. The first segment [10,16] is, however, used only under special circumstances to avoid quality degradation for pitch lags below the lowest pitch quantization limit. The autocorrelation function is first computed for each pitch lag value by

$$C(d) = \sum_{n=0}^{L_{sec}} s_{wd}(n) s_{wd}(n-d), \quad (5.8.1-1)$$

where the summation limit depends on the delay section according to

$$\begin{aligned} L_{sec} &= 40 & \text{for } & d = 10, \dots, 16 \\ L_{sec} &= 40 & \text{for } & d = 17, \dots, 31 \\ L_{sec} &= 62 & \text{for } & d = 32, \dots, 61 \\ L_{sec} &= 115 & \text{for } & d = 62, \dots, 115 \end{aligned} \quad (5.8.1-2)$$

This will ensure that for a given delay value at least one pitch cycle is included in the correlation computation.

### 5.8.2 Correlation Reinforcement with Past Pitch Values

The autocorrelation function is then weighted to emphasize the function for delays in the neighborhood of pitch lag parameters determined from the previous frame, and the delay is identified based on the maximum of the weighted correlation function.

The weighting is given by a triangular window of size 27 and it is centered around the extrapolated pitch lags in the current frame based on lags determined in the previous frame. The extrapolated pitch lags are pitch lags estimated for the current frame in the neighborhood of those determined for the previous frame as will be shown below. The pitch neighborhood weighting function is given by

$$w_{pn}(13+i) = w_{pn}(13-i) = 1 + \alpha_{pn}(1-i/14), \quad i = 0, \dots, 13. \quad (5.8.2-1)$$

where  $\alpha_{pn}$  is a scaling factor based on the voicing measure from the previous frame (the normalized pitch correlation) and the pitch stability in the previous frame. During voiced segments with smooth

1 pitch evolution, the scaling factor is updated from previous frame by adding a value of  $0.16\overline{R}_{xy}$ , and it  
 2 is upper-limited to 0.7.  $\overline{R}_{xy}$  is the average of the normalized correlation in the two half frames of the  
 3 previous frame. The scaling factor  $\alpha_{pn}$  is reset to zero (no weighting) if  $\overline{R}_{xy}$  is less than 0.4 or if the  
 4 pitch lag evolution in the previous frame is unstable or if the difference between the energy of the  
 5 previous frame and the long-term average energy of active speech is more than a certain threshold.  
 6 The pitch instability is determined by testing the pitch coherence between consecutive half frames.  
 7 The pitch values of two consecutive half frames are considered coherent if the following condition is  
 8 satisfied:

$$10 \quad (\max\_value < 1.4 \min\_value) \text{ AND } ((\max\_value - \min\_value) < 14)$$

11  
 12 where max\_value and min\_value denote the maximum and minimum of the two pitch values,  
 13 respectively. The pitch evolution in a certain frame is considered as stable if pitch coherence is  
 14 satisfied for both the first half frame and the last half frame of previous frame as well as the two  
 15 halves of the present frame.

16  
 17 The extrapolated pitch lag in the first half frame,  $\tilde{d}_0$ , is computed as the pitch lag from the second  
 18 half frame of the previous frame plus a pitch evolution factor  $f_{evol}$  computed from the previous frame  
 19 (as described in Section 5.8.6). The extrapolated pitch lags in the second half frame and in the  
 20 lookahead,  $\tilde{d}_1$  and  $\tilde{d}_2$ , are computed as the pitch lag from the second half frame of the previous  
 21 frame plus twice the pitch evolution factor. That is

$$23 \quad \tilde{d}_0 = d_{-1} + f_{evol}, \quad \tilde{d}_1 = d_{-1} + 2f_{evol} \quad \text{and} \quad \tilde{d}_2 = \tilde{d}_1 \quad (5.8.2-2)$$

24  
 25  
 26 where  $d_{-1}$  is the pitch lag in the second half-frame of previous frame.

27  
 28 The pitch evolution factor is obtained by averaging the pitch differences of consecutive half frames  
 29 that are determined as coherent (according to the coherence rule described above). The  
 30 autocorrelation function weighted around the extrapolated pitch lag  $\tilde{d}$  is given by

$$32 \quad C^w(\tilde{d} + i) = C(\tilde{d} + i)w_{pn}(13 + i), \quad i = -13, \dots, 13 \quad (5.8.2-3)$$

### 34 5.8.3 Normalized Correlation Computation

35  
 36 After weighting the correlation function with the triangular window of Equation (5.8.2-1) centered at  
 37 the extrapolated pitch lag, the maxima of the weighted correlation function in each of the  $N_{sec}$  sections  
 38 described above,  $C^w(d_{max}^{(i)})$ ,  $i=0, 1, 2$  and  $3$ , are determined.  $N_{sec}$  equals 3 or 4, depending whether  
 39 section 0 is used. Section 0 is used only during stable high-pitched speech segments, i.e. if the open-  
 40 loop pitch period of the 2<sup>nd</sup> half frame of the previous frame is lower or equal to 24 and the scaling  
 41 factor  $\alpha_{pn}$  is higher or equal to 0.1. The correlations at  $N_{sec}$  lag positions (for the  $N_{sec}$  sections) are  
 42 normalized according to

$$44 \quad C_{norm}(d_{max}) = \frac{C(d_{max})}{\sqrt{\sum_{n=0}^{L_{sec}} s_{wd}^2(n) \sum_{n=0}^{L_{sec}} s_{wd}^2(n - d_{max})}}, \quad (5.8.3-1)$$

1 where the summation limit depends on the pitch delay section as defined in Equation (5.8.1-2). At this  
 2 point,  $N_{sec}$  candidate pitch lags  $d_{max}^{(k)}(i)$  have been determined (one per section) for each half frame  
 3 and look-ahead with corresponding values of normalized correlations (both weighted and raw), where  
 4  $i$  is the half-frame index and  $k$  is the section index. All remaining processing is performed using only  
 5 these selected values, greatly reducing the overall complexity.

6  
 7 Note that the last section (long pitch periods) is not searched for the look-ahead. Instead, the  
 8 normalized correlation values and the corresponding pitch are obtained from the last section search  
 9 of the 2<sup>nd</sup> half frame. The reason is that the summation limit in the last section is much larger than the  
 10 available lookahead and a limitation of the computational complexity.

#### 11 **5.8.4 Correlation Reinforcement with Pitch Lag Multiples**

12  
 13 In order to avoid selecting pitch multiples, the weighted normalized correlation in a lower pitch delay  
 14 section is further emphasized if one of its multiples is in the neighborhood of the pitch lag at the  
 15 maximum weighted correlation in a higher section. That is,  
 16

$$17 \text{ If } (|k \times d_{max}^{(2)} - d_{max}^{(3)}| \leq k) \text{ then } C_{norm}^w(d_{max}^{(2)}) = \alpha_{mult} C_{norm}^w(d_{max}^{(2)}), \alpha_{mult} = (\alpha_{mult})^2$$

$$18 \text{ If } (|k \times d_{max}^{(1)} - d_{max}^{(2)}| \leq k) \text{ then } C_{norm}^w(d_{max}^{(1)}) = \alpha_{mult} C_{norm}^w(d_{max}^{(1)}),$$

19  
 20 where  $\alpha_{mult}=1.17$ . In this way, if a pitch period multiple is found in a higher section, the maximum  
 21 weighted correlation in the lower section is emphasized by a factor 1.17. If however the maximum  
 22 correlation in sections 3 is at pitch period multiple of the maximum correlation of the section 2 and the  
 23 maximum correlation in sections 2 is at pitch period multiple of the maximum correlation of the section  
 24 1, the maximum weighted correlation in section 1 is emphasized twice.  
 25

26  
 27 It can be seen that the “neighborhood” is larger with the number of multiples  $k$  to take into account an  
 28 increasing uncertainty on the pitch length (the pitch length is estimated roughly with integer precision  
 29 at 6400 Hz sampling frequency). The uncertainty is even higher for the section 3 of the lookahead,  
 30 when the maximum correlation value and the corresponding pitch period were simply copied from the  
 31 corresponding section of the 2<sup>nd</sup> half frame. For this reason, the condition in the first relation above is  
 32 modified for lookahead as follows  
 33

$$34 \text{ If } (|k \times d_{max}^{(1)} - d_{max}^{(2)}| \leq 2(k-1))$$

35  
 36 Note that section 0 is not considered here, i.e. the maximum normalized correlation in section 0 is  
 37 never emphasized.

#### 38 **5.8.5 Initial Pitch Lag Determination and Reinforcement Based on Pitch Coherence with** 39 **other Half-frames**

40  
 41 For each half frame and lookahead, an initial pitch lag  $d_{init}(i)$ ,  $i$  being half-frame index, and  
 42 corresponding raw (unweighted) normalized correlation are determined by searching the maximum of  
 43  $N_{sec}$  emphasized normalized correlations (corresponding to  $N_{sec}$  sections).  
 44

45  
 46 In the previously described pitch search and tracking, the correlations were weighted at the  
 47 neighborhood of a pitch lag extrapolated from previous frame pitch lags. Now, to further track the right  
 48 pitch value, another level of weighting is performed of the correlations in each section and each half-  
 49 frame based on the pitch coherence of initially determined pitch lags  $d_{init}(i)$  with pitch lags  
 50  $d_{max}^{(k)}(j \neq i)$  of each section in the other half-frames. That is, if the initial pitch lag in a half-frame  $i$  is  
 51 coherent with pitch lag of section  $k$  in half-frame  $j$ , then the corresponding weighted normalized

1 correlation of section  $k$  in half-frame  $j$  is further emphasized by weighting it by the value  
 2  $1 + \alpha(1 - \delta_{pit} / 14)$  where  $\delta_{pit}$  is the absolute difference between  $d_{init}(i)$  and  $d_{max}^{(k)}(j \neq i)$ , and  
 3  $\alpha = 0.4(C_{norm}(d_{init}^{(i)}) + 0.5r_e)$  where  $\alpha$  is upper-bounded to 0.4,  $C_{norm}(d)$  is the normalized correlation  
 4 as defined in Equation (5.8.3-1), but without weighting and  $r_e$  is a correction added to the normalized  
 5 correlation in order to compensate for the decrease of normalized correlation in the presence of  
 6 background noise (defined in Equation (5.4.4-1)). This procedure will further help in avoiding selecting  
 7 pitch multiples and insure pitch continuity in adjacent half-frames.

### 5.8.6 Pitch Lag Determination and Parameter Update

11 Finally, the pitch lags in each half-frame,  $d_0$ ,  $d_1$ , and  $d_2$ , are determined. Again, the further  
 12 emphasized normalized correlation in the last section of look-ahead is copied from the last section of  
 13 the 2<sup>nd</sup> half frame before computing the final pitch lags. They are determined by searching the  
 14 maximum of the emphasized normalized correlations corresponding to each of the  $N_{sec}$  sections. After  
 15 determining the pitch lags, the parameters needed for the next frame pitch search are updated. The  
 16 average normalized correlation  $\bar{R}_{xy}$  is updated by:

$$\bar{R}_{xy} = 0.5(C_{norm}(d_0) + C_{norm}(d_1)) + 0.5r_e \quad \text{constrained by } \bar{R}_{xy} \leq 1, \quad (5.8.6-1)$$

21 Finally, the pitch evolution factor  $f_{evol}$  to be used in computing the extrapolated pitch lags in the next  
 22 frame is updated. The pitch evolution factor is given by averaging the pitch differences of consecutive  
 23 half frames that are determined as coherent. If  $d_{-1}$  is the pitch lag in the second half of the previous  
 24 frame then pitch evolution is given by

$$\begin{aligned} & \delta_{pitch} = 0 \\ & cnt = 0 \\ & \text{For } i=0 \text{ to } 2 \text{ do} \\ & \quad \text{if } d_i \text{ and } d_{i-1} \text{ are coherent} \\ & \quad \quad \delta_{pitch} = \delta_{pitch} + d_i - d_{i-1} \\ & \quad \quad cnt = cnt + 1 \\ & \text{if } (cnt > 0) \quad f_{evol} = \delta_{pitch} / cnt \quad \text{else } f_{evol} = 0 \end{aligned}$$

34 Since the search is performed on the decimated weighted signal, the determined pitch lags  $d_0$ ,  $d_1$ ,  
 35 and  $d_2$  are multiplied by 2 to obtain the open loop pitch lags for the 3 half-frames.

37 If a selected pitch lag value is lower or equal to the minimum pitch quantization level (34 samples at  
 38 12800 Hz sampling frequency), the pitch value is multiplied by two. The reason is that the calculated  
 39 pitch value can saturate to this minimum pitch lag when the real pitch period is below that value. This  
 40 may occur in case of high-pitched female or child speakers. Pitch periods under the lower limit of 34,  
 41 but very close to it, would saturate into 34 causing constant pitch period to be generated in the  
 42 decoder and consequently quality degradation.

## 5.9 Noise Energy Estimate Update and Voiced Critical Band Determination

45 **Routine Name:** noise\_est

46 **Inputs:**

- 1 •  $E_{CB}(i)$ : Average energy in  $i$ th critical band
- 2 •  $N_{CB}(i)$ : Noise estimate in  $i$ th critical band
- 3 •  $r_e$ : Noise correction factor
- 4 •  $d_0, d_1$ , and  $d_2$ : The pitch lags in each half-frame
- 5 •  $C_{norm}(d)$ : The normalized correlation at pitch lags  $d_0, d_1$ , and  $d_2$ .
- 6 •  $E_i$ : Total frame energy
- 7 •  $E(i)$ : LP residual energies

#### 8 **Outputs:**

- 9 •  $N_{CB}(i)$ : Noise estimate in  $i$ th critical band
- 10 •  $K_{voic}$ : Voiced critical bands

#### 11 **Initialization:**

- 12 •  $N_{CB}(i)$  and  $E_{CB,LT}(i)$  are initialized to 0.03 (unless otherwise described for  $N_{CB}(i)$  in Section
- 13 5.4. Previous frame pitch lag  $d_{-1}$  is initialized to zero. The noise update decision hangover is
- 14 initialized to 6.

15

16 This module updates the noise energy estimates per critical band for noise suppression. The update  
 17 is performed during inactive speech intervals. However, the VAD decision obtained in Section 5.4,  
 18 which is based on the SNR per critical band, is not used for determining whether the noise energy  
 19 estimates are updated. Another decision is performed based on other parameters independent of the  
 20 SNR per critical band. The parameters used for the noise update decision are: pitch stability, signal  
 21 non-stationarity, voicing, and ratio between 2<sup>nd</sup> order and 16<sup>th</sup> order LP residual error energies. These  
 22 parameters have generally low sensitivity to the noise level variations.

23

24 The reason for not using the encoder VAD decision for noise update is to make the noise estimation  
 25 robust to rapidly changing noise levels. If the encoder VAD decision were used for the noise update,  
 26 a sudden increase in noise level would cause an increase of SNR even for inactive speech frames,  
 27 preventing the noise estimator to update, which in turn would maintain the SNR high in following  
 28 frames. Consequently, the noise update would be blocked and some other logic would be needed to  
 29 resume the noise adaptation.

30

31 The pitch stability counter is computed as

32

$$33 \quad pc = |d_0 - d_{-1}| + |d_1 - d_0| + |d_2 - d_1| \quad (5.9-1)$$

34

35 where  $d_0, d_1$ , and  $d_2$ , are the open-loop pitch lags for the first half-frame, second half-frame, and  
 36 the lookahead, respectively, and  $d_{-1}$  is the lag of the second half-frame of the pervious frame. Since  
 37 for pitch lags larger than 122, the open-loop pitch search module sets  $d_2 = d_1$ , then for such lags the  
 38 value of  $pc$  in equation (5.9-1) is multiplied by 3/2 to compensate for the missing third term in the  
 39 equation. The pitch stability is true if the value of  $pc$  is less than 12. Further, for frames with low  
 40 voicing,  $pc$  is set to 12 to indicate pitch instability. That is

41

$$42 \quad \text{If } (C_{norm}(d_0) + C_{norm}(d_1) + C_{norm}(d_2))/3 + r_e < 0.7 \text{ then } pc = 12, \quad (5.9-2)$$

43

44 where  $C_{norm}(d)$  is the normalized raw correlation as defined in Equation (5.8.3-1) but without  
 45 weighting and  $r_e$  is a correction added to the normalized correlation in order to compensate for the

1 decrease of normalized correlation in the presence of background noise (defined in Equation (5.4.4-  
2 1)).

3  
4 The signal non-stationarity estimation is performed based on the product of the ratios between the  
5 energy per critical band and the average long-term energy per critical band.

6  
7 The average long-term energy per critical band is updated by

$$9 \quad E_{CB,LT}(i) = \alpha_e E_{CB,LT}(i) + (1 - \alpha_e) \bar{E}_{CB}(i), \quad \text{For } i=b_{min} \text{ to } b_{max}, \quad (5.9-3)$$

10  
11 where  $b_{min}=0$  and  $b_{max}=19$  in case of wideband signals, and  $b_{min}=1$  and  $b_{max}=16$  in case of  
12 narrowband signals, and  $\bar{E}_{CB}(i)$  is the frame energy per critical band defined in Equation (5.4.2-1).

13 The update factor  $\alpha_e$  is a linear function of the total frame energy, defined in Equation (5.2-7), and it  
14 is given as follows:

15  
16 For wideband signals:  $\alpha_e = 0.0245E_{tot} - 0.235$  constrained by  $0.5 \leq \alpha_e \leq 0.99$ .

17 For narrowband signals:  $\alpha_e = 0.00091E_{tot} + 0.3185$  constrained by  $0.5 \leq \alpha_e \leq 0.999$ .

18  
19 The frame non-stationarity is given by the product of the ratios between the frame energy and  
20 average long-term energy per critical band. That is

$$21 \quad nonstat = \prod_{i=b_{min}}^{b_{max}} \frac{\max(\bar{E}_{CB}(i), E_{CB,LT}(i))}{\min(\bar{E}_{CB}(i), E_{CB,LT}(i))} \quad (5.9-4)$$

22  
23  
24 The voicing factor for noise update is given by

$$25 \quad voicing = (C_{norm}(d_0) + C_{norm}(d_1)) / 2 + r_e. \quad (5.9-5)$$

26  
27  
28 Finally, the ratio between the LP residual energy after 2<sup>nd</sup> order and 16<sup>th</sup> order analysis is given by

$$29 \quad resid\_ratio = E(2) / E(16) \quad (5.9-6)$$

30  
31  
32 where  $E(2)$  and  $E(16)$  are the LP residual energies after 2<sup>nd</sup> order and 16<sup>th</sup> order analysis, and  
33 computed in the Levinson-Durbin recursion of Equation (5.6.2-2). This ratio reflects the fact that to  
34 represent a signal spectral envelope a higher order of LP is generally needed for speech signal than  
35 for noise. In other words, the difference between  $E(2)$  and  $E(16)$  is expected to be lower for noise  
36 than for active speech.

37  
38 The update decision is determined based on a variable *noise\_update*, which is initially set to 6, and it  
39 is decremented by 1 if an inactive frame is detected and incremented by 2 if an active frame is  
40 detected. Further, *noise\_update* is bounded by 0 and 6. The noise energies are updated only when  
41 *noise\_update*=0.

42  
43 The value of the variable *noise\_update* is updated in each frame as follows:

44  
45 If ( $nonstat > th_{stat}$ ) OR ( $pc < 12$ ) OR ( $voicing > 0.85$ ) OR ( $resid\_ratio > th_{resid}$ )  
46  $noise\_update = noise\_update + 2$   
47 Else  
48  $noise\_update = noise\_update - 1$

49  
50 where for wideband signals,  $th_{stat}=350000$  and  $th_{resid}=1.9$ , and for narrowband signals,  $th_{stat}=500000$   
51 and  $th_{resid}=11$ .

In other words, frames are declared inactive for noise update when

$$(nonstat \leq th_{stat}) \text{ AND } (pc \geq 12) \text{ AND } (voicing \leq 0.85) \text{ AND } (resid\_ratio \leq th_{resid})$$

and a hangover of 6 frames is used before noise update takes place.

Thus, if  $noise\_update=0$  then

$$\text{for } i=0 \text{ to } 19 \quad N_{CB}(i) = N_{tmp}(i)$$

where  $N_{tmp}(i)$  is the temporary updated noise energy already computed in Equation (5.4.2-4).

### 5.9.1 Update of Voicing Cutoff Frequency

The cut-off frequency below which a signal is considered voiced is updated. This frequency is used to determine the number of critical bands for which noise suppression is performed using per bin processing.

First, a voicing measure is computed as

$$v_g = 0.4C_{norm}(d_1) + 0.6C_{norm}(d_2) + r_e \quad (5.9.1-1)$$

and the voicing cut-off frequency is given by

$$f_c = 0.00017118 e^{17.9772v_g} \quad \text{constrained by } 325 \leq f_c \leq 3700 \quad (5.9.1-2)$$

Then, the number of critical bands,  $K_{voic}$ , having an upper frequency not exceeding  $f_c$  is determined. The bounds of  $325 \leq f_c \leq 3700$  are such that per bin processing is performed on a minimum of 3 bands and a maximum of 17 bands (refer to the critical bands upper limits in Section 5.2). Note that in the voicing measure calculation, more weight is given to the normalized correlation of the lookahead since the determined number of voiced bands will be used in the next frame.

Thus, in the following frame, for the first  $K_{voic}$  critical bands, the noise suppression will use per bin processing as described in Section 5.5.

Note that for frames with low voicing and for large pitch delays, only per critical band processing is used and thus  $K_{voic}$  is set to 0. The following condition is used:

$$\text{If } (0.4C_{norm}(d_1) + 0.6C_{norm}(d_2) \leq 0.72) \text{ OR } (d_1 > 116) \text{ OR } (d_2 > 116) \text{ then } K_{voic} = 0.$$

## 5.10 Unvoiced Signal Classification: Selection of Unvoiced-HR and Unvoiced-QR

**Routine Name:** rate\_select

**Inputs:**

- $s(n)$ : The denoised speech signal
- Mode of operation and HR-maximum signaling flag

- 1 •  $C_{norm}(d)$ : The normalized correlation at pitch lags  $d_0$ ,  $d_1$ , and  $d_2$ .
- 2 • VAD\_flag and local VAD flag
- 3 •  $r_e$ : Noise correction factor
- 4 •  $E_{rel}$ : Relative frame energy
- 5 •  $\bar{N}_f$ : Long-term average noise energy
- 6 • Previous frame encoding scheme

7 **Outputs:**

- 8 •  $e_{tilt}(i)$ : spectral tilt
- 9 • Encoding type

10 **Initialization:**

- 11 •  $e_{old}$ : The tilt in the second half of the previous frame is initialized to 10. Previous frame
- 12 encoding type is initialized to Generic FR.

13  
 14 Figure 5.10-1 shows a simplified high-level description of the signal classification procedure. If voice  
 15 activity is not detected, CNG-ER encoding type is utilized. If voice activity is detected, the voiced  
 16 versus unvoiced classification is performed. If the frame is classified as unvoiced, it is encoded with  
 17 either Unvoiced HR or Unvoiced QR encoding types. If the frame is not classified as unvoiced, then  
 18 stable voiced classification is applied. If the frame is classified as stable voiced, it can be encoded  
 19 using Voiced HR encoding type. Otherwise, the frame is likely to contain a non-stationary speech  
 20 segment such as a voiced onset or rapidly evolving voiced speech signal. These frames typically  
 21 require a general-purpose coding model at high bit rate for sustaining good speech quality. Thus in  
 22 this case an appropriate FR encoding type is mainly used. Frames with very low energy and not  
 23 detected as non-speech, unvoiced or stable voiced can be encoded using Generic HR coding in order  
 24 to reduce the average data rate.

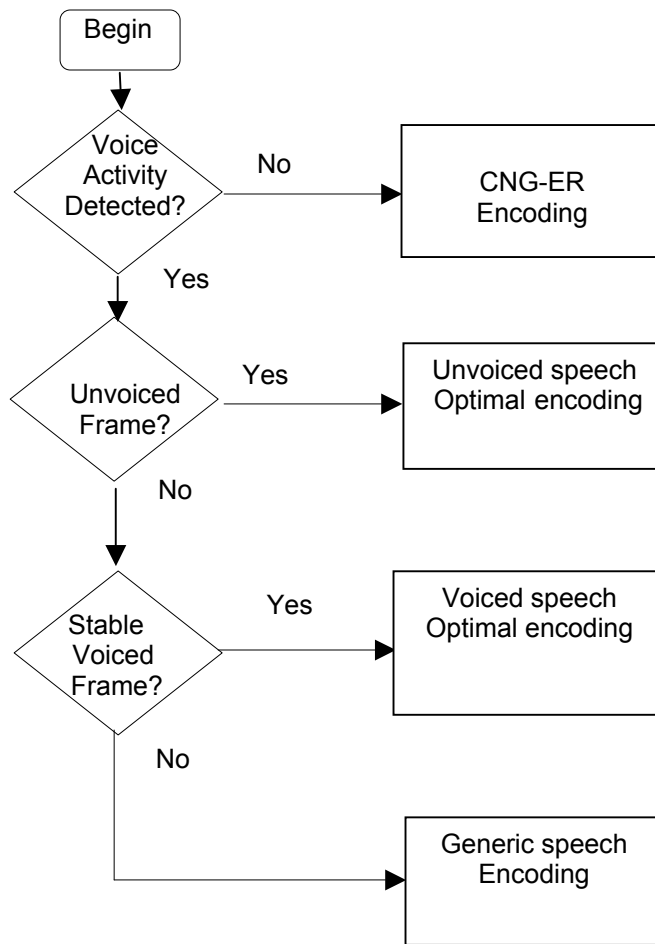
25  
 26 This is a simplified description of the rate determination procedure. The actual choice of encoding  
 27 type in a certain frame is based both on the frame classification and the required mode of operation.  
 28 In VMR-WB mode 0 for instance, HR Voiced encoding type is not used and the Unvoiced QR  
 29 encoding type is used only in VMR-WB mode 2. Thus, the classification thresholds depend on the  
 30 VMR-WB mode of operation. Furthermore, the rate determination procedure has to comply with  
 31 maximum and minimum rate constraints.

32  
 33 The VAD has been described in Section 5.3. In this section, unvoiced signal classification will be  
 34 described. The stable voiced signal classification will be described as part of the signal modification  
 35 procedure (see Section 5.11.4). Finally, the last stage of classification to determine if the frame can  
 36 be encoded using Generic HR encoding type is described in Section 5.12.

37  
 38 The unvoiced parts of the signal are characterized by missing periodic component and can be further  
 39 divided into unstable frames, where the energy and the spectrum changes rapidly, and stable frames  
 40 where these characteristics remain relatively stable. The classification of unvoiced frames exploits the  
 41 following parameters:

- 42 • A voicing measure, computed as an averaged normalized correlation,  $\bar{R}_{xy,3}$ ,
- 43 • Spectral tilt measures,  $e_{tilt}(0)$  and  $e_{tilt}(1)$ , for both spectral analysis per frame,
- 44 • A signal energy ratio ( $dE$ ) used to assess the frame energy variation within the frame  
 45 and thus the frame stability, and
- 46 • Relative frame energy

47  
 48



1  
2 **Figure 5.10-1: Functional block diagram of the high-level signal classification logic.**

3 **5.10.1 Voicing Measure**

4  
5 The normalized correlation, used to determine the voicing measure, is computed as part of the open-  
6 loop search module described in Section 5.8. The normalized correlation is computed similar to  
7 Equation (5.8.6-1) with also considering the lookahead half-frame. That is

$$8 \quad \bar{R}_{xy3} = \frac{1}{3}(C_{norm}(d_0) + C_{norm}(d_1) + C_{norm}(d_2)) + r_e \quad (5.10.1-1)$$

9 where  $C_{norm}(d)$  is computed as in Equation (5.8.3-1) and  $r_e$  as in Equation (5.4.4-1).

10  
11 **5.10.2 Spectral Tilt**

12  
13 The spectral tilt parameter contains the information about the frequency distribution of energy. The  
14 spectral tilt is estimated in the frequency domain as a ratio between the energy concentrated in low  
15 frequencies and the energy concentrated in high frequencies.

16  
17 The energy in high frequencies is computed as the average of the energies of the last two critical  
18 bands

$$19 \quad \bar{E}_h = 0.5[E_{CB}(b_{max} - 1) + E_{CB}(b_{max})] \quad (5.10.2-1)$$

20  
21 where  $E_{CB}(i)$  are the critical band energy computed in Equation (5.2-4) and  $b_{max}=19$  for WB inputs  
22 and  $b_{max}=16$  for NB inputs.  
23

1  
2 The energy in low frequencies is computed as the average of the energies in the first 9 critical bands.  
3 The middle critical bands have been excluded from the computation to improve the discrimination  
4 between frames with high-energy concentration in low frequencies (generally voiced) and with high-  
5 energy concentration in high frequencies (generally unvoiced). In between, the energy content is not  
6 informative for any of the classes and increases the decision uncertainty.

7  
8 The energy in low frequencies is computed differently for long pitch periods and short pitch periods.  
9 For voiced female speech segments, the harmonic structure of the spectrum is exploited to increase  
10 the voiced-unvoiced discrimination. Thus for short pitch periods,  $E_l$  is computed bin-wise and only  
11 frequency bins sufficiently close to the speech harmonics are taken into account in the summation.  
12 That is

$$\bar{E}_l = \frac{1}{cnt} \sum_{k=K_{min}}^{24} E_{BIN}(k) w_h(k) \quad (5.10.2-2)$$

13  
14  
15  
16 where  $K_{min}$  is the first bin ( $K_{min}=0$  for WB inputs and  $K_{min}=2$  for NB inputs) and  $E_{BIN}(k)$  are the bin  
17 energies, as defined in Equation (5.2-5), in the first 25 frequency bins (the DC component is not  
18 considered). Note that these 25 bins correspond to the first 10 critical bands and that the first 2 bins  
19 not included in the case of NB input constitute the 1<sup>st</sup> critical band. In the summation above, only the  
20 terms related to the bins close to the pitch harmonics are considered, so  $w_h(k)$  is set to 1, if the  
21 distance between the nearest harmonics is not larger than a certain frequency threshold (50 Hz) and  
22 is set to 0 otherwise. The counter  $cnt$  is the number of the non-zero terms in the summation. Only  
23 bins closer than 50 Hz to the nearest harmonics are taken into account. Hence, if the structure is  
24 harmonic in low frequencies, only high-energy term will be included in the sum. On the other hand, if  
25 the structure is not harmonic, the selection of the terms will be random and the sum will be smaller.  
26 Thus even unvoiced sounds with high energy content in low frequencies can be detected. This  
27 processing cannot be done for longer pitch periods, as the frequency resolution is not sufficient. For  
28 pitch values larger than 128 or for a priori unvoiced sounds the low frequency energy is computed per  
29 critical band as

$$\bar{E}_l = \frac{1}{10} \sum_{k=0}^9 E_{CB}(k) \quad \text{or} \quad \bar{E}_l = \frac{1}{9} \sum_{k=1}^9 E_{CB}(k) \quad (5.10.2-3)$$

30  
31  
32 for WB and NB inputs, respectively. A priori unvoiced sounds are determined when  
33  $\frac{1}{2}(C_{norm}(d_1) + C_{norm}(d_3)) + r_e < 0.6$ .

34  
35 The resulting low and high frequency energies are obtained by subtracting estimated noise energy  
36 from the values  $\bar{E}_l$  and  $\bar{E}_h$  calculated above. That is

$$E_h = \bar{E}_h - N_h \quad (5.10.2-4)$$

$$E_l = \bar{E}_l - N_l \quad (5.10.2-5)$$

37  
38  
39  
40  
41 where  $N_h$  and  $N_l$  are the averaged noise energies in the last 2 critical bands and first 10 critical bands  
42 (or 9 for NB inputs), respectively, computed similar to Equations (5.10.2-1) and (5.10.2-3). The  
43 estimated noise energies have been added to the tilt computation to account for the presence of  
44 background noise.

45  
46 Finally, the spectral tilt is given by

$$e_{tilt}(i) = \frac{E_l}{E_h} \quad , \quad (5.10.2-6)$$

Note that the spectral tilt computation is performed twice per frame to obtain  $e_{ilt}(0)$  and  $e_{ilt}(1)$  corresponding to both spectral analysis per frame. The average spectral tilt used in unvoiced frame classification is given by

$$e_t = \frac{1}{3}(e_{old} + e_{ilt}(0) + e_{ilt}(1)) \quad (5.10.2-7)$$

where  $e_{old}$  is the tilt in the second half of the previous frame

### 5.10.3 Energy Variation

The energy variation  $dE$  is evaluated on the denoised speech signal  $s(n)$ , where  $n=0$  corresponds to the beginning of the current frame. The signal energy is evaluated twice per subframe, i.e. 8 times per frame, based on short-time segments of length 32 samples. Further, the short-term energies of the last 32 samples from the previous frame and the first 32 samples from next frame are also computed. The short-time maximum energies are computed as

$$E_{st}^{(1)}(j) = \max_{i=0}^{31}(s^2(i + 32j)), \quad j = -1, \dots, 8, \quad (5.10.3-1)$$

where  $j=-1$  and  $j=8$  correspond to the end of previous frame and the beginning of next frame. Another set of 9 maximum energies is computed by shifting the speech indices in Equation (5.10.3-1) by 16 samples. That is

$$E_{st}^{(2)}(j) = \max_{i=0}^{31}(s^2(i + 32j - 16)), \quad j = 0, \dots, 8, \quad (5.10.3-2)$$

The maximum energy variation  $dE$  is computed as the maximum of the following:

$$\begin{aligned} & E_{st}^{(1)}(0) / E_{st}^{(1)}(-1) \\ & E_{st}^{(1)}(7) / E_{st}^{(1)}(8) \\ & \frac{\max(E_{st}^{(1)}(j), E_{st}^{(1)}(j-1))}{\min(E_{st}^{(1)}(j), E_{st}^{(1)}(j-1))} \quad \text{For } j=1 \text{ to } 7 \\ & \frac{\max(E_{st}^{(2)}(j), E_{st}^{(2)}(j-1))}{\min(E_{st}^{(2)}(j), E_{st}^{(2)}(j-1))} \quad \text{For } j=1 \text{ to } 8 \end{aligned}$$

### 5.10.4 Relative Frame Energy $E_{rel}$

The relative frame energy is used to identify low energy frames where the unvoiced classification thresholds can be relaxed, allowing these frames to be classified as unvoiced more easily. The relative frame energy is also used later to identify low energy frames, which has not been classified as background noise frames or unvoiced frames. These frames can be encoded with a generic HR encoding type in order to reduce the average data rate (Section 5.12). The relative frame energy is computed in Equation (5.4.1-2).

### 5.10.5 Unvoiced Speech Classification in Rate-Set II

The classification of unvoiced speech frames is based on the parameters described above, namely: the voicing measure  $\bar{R}_{xy3}$ , the spectral tilt  $e_t$ , the energy variation within a frame  $dE$ , and the relative frame energy  $E_{rel}$ . The decision is made based on at least three of these parameters. The decision

1 thresholds are set based on the operating mode. For operating modes with lower allowable data  
 2 rates, the thresholds are set to favor more unvoiced classification (since a half-rate or a quarter-rate  
 3 encoding type will be used to encode the frame). Unvoiced frames are usually encoded with  
 4 Unvoiced HR encoder. However, in VMR-WB mode 2, Unvoiced QR is also used in order to further  
 5 reduce the average data rate, if additional certain conditions are satisfied.

6  
 7 In VMR-WB mode 0, the frame is encoded as Unvoiced HR if the following condition is satisfied

$$9 \quad (\overline{R}_{xy3} < th_1) \text{ AND } (e_t < th_2) \text{ AND } (dE < th_3)$$

10  
 11 where  $th_1 = 0.5$ ,  $th_2 = 1$ , and  $th_3 = 0$  if the long-term average noise energy  $\overline{N}_f > 21$  and  $th_3 = 4$   
 12 otherwise. Furthermore, for WB inputs only,  $th_3 = 3.2$  if  $\overline{N}_f > 34$ .

13  
 14 In VAD, a decision hangover is used. Thus, after active speech periods, when the algorithm decides  
 15 that the frame is an inactive speech frame, a local VAD is set to zero but the actual VAD\_flag is set to  
 16 zero only after a certain number of frames are elapsed (the hangover period). This avoids clipping of  
 17 speech offsets. In both VMR-WB modes 1 and 2, if the local VAD is zero, the frame is classified as an  
 18 Unvoiced frame.

19  
 20 In VMR-WB mode 1, the frame is encoded as Unvoiced HR if local VAD=0 OR if the following  
 21 condition is satisfied

$$22 \quad (\overline{R}_{xy3} < th_4) \text{ AND } (e_t < th_5) \text{ AND } ((dE < th_6) \text{ OR } (E_{rel} < th_7))$$

23  
 24 where  $th_4 = 0.695$ ,  $th_5 = 4$ ,  $th_6 = 40$ , and  $th_7 = -14$ . Note that  $dE$  is increased by 34 in case of  
 25 NB inputs.

26  
 27 In VMR-WB mode 2, the frame is declared as an Unvoiced frame if local VAD=0 OR if the following  
 28 condition is satisfied

$$29 \quad (\overline{R}_{xy3} < th_8) \text{ AND } (e_t < th_9) \text{ AND } ((dE < th_{10}) \text{ OR } (E_{rel} < th_{11}))$$

30  
 31 where  $th_8 = 0.695$ ,  $th_9 = 4$ ,  $th_{10} = 60$ , and  $th_{11} = -14$ ,  $th_7 = -14$ . Note that  $dE$  is increased by  
 32 10 in case of NB inputs and if  $\overline{N}_f > 21$ .

33  
 34 In VMR-WB mode 2, unvoiced frames are usually encoded as Unvoiced HR. However, they can also  
 35 be encoded with Unvoiced QR, if the following further conditions are also satisfied: If the last frame is  
 36 either unvoiced or background noise frame, and if at the end of the frame the energy is concentrated  
 37 in high frequencies and no potential voiced onset is detected in the lookahead then the frame is  
 38 encoded as Unvoiced QR. The last two conditions are detected as:

$$39 \quad (C_{norm}(d_2) < th_{12}) \text{ AND } (e_{tilt}(1) < th_{13}) \text{ where } th_{12} = 0.73, th_{13} = 3.$$

40  
 41 Note that  $R_{xy3}(2)$  is the normalized correlation in the lookahead and  $e_{tilt}(1)$  is the tilt in the second  
 42 spectral analysis, which spans the end of the frame and the lookahead.

#### 43 5.10.6 Unvoiced Speech Classification in Rate-Set I

44 In Rate-Set I (VMR-WB mode 4), unvoiced frames are encoded using RS-I Unvoiced HR. The first  
 45 condition to encode a frame as RS-I Unvoiced HR is that the last frame has been encoded using an  
 46

1 Unvoiced or CNG encoding type or the energy of the last frame is concentrated at high frequencies  
 2 and the voicing of at the end of last frame is below a certain threshold. That is

$$3 \quad (e_{old} < 2.4) \quad \text{AND} \quad (C_{norm}(d_0) + r_e < 0.66)$$

5  
 6 Further, the frame should not be detected as a voiced onset or a plosive frame. This is detected by  
 7 testing the high frequency content at the end of the frame and the energy variation within the frame.  
 8 That is

$$9 \quad (e_{tilt}(1) < 2.7) \quad \text{AND} \quad (dE0 < 250)$$

11 where is defined in Equation (5.10.2-6) and  $dE0$  is given by the maximum of

$$12 \quad E_{st}^{(1)}(i+1)/E_{st}^{(1)}(i), \quad E_{st}^{(2)}(i+1)/E_{st}^{(2)}(i), \quad i=0,\dots,7$$

14 with the test for maximum variation performed only for low energy segments ( $10 \log_{10}(E_{st}(i)) < 37$ )

15 Further, the following condition should be also satisfied

$$16 \quad ((\bar{R}_{xy3} < th_4) \quad \text{AND} \quad (e_t < th_5)) \quad \text{OR} \quad (E_{rel} < th_7)$$

18 where  $th_4 = 0.695$ ,  $th_5 = 4$ , and  $th_7 = -14$ .

20  
 21 If all above conditions are satisfied then the frame is encoded using RS-I Unvoiced HR encoding  
 22 type.

## 23 5.11 Signal Modification and HR Voiced Rate Selection

24  
 25 **Routine Name:** sig\_modification

26 **Inputs:**

- 27 •  $s(n)$ : The denoised input speech
- 28 •  $s_w(n)$ : The weighted speech signal
- 29 •  $a_i$ : The LP filter coefficients
- 30 • Mode of operation and HR-maximum signaling flag
- 31 •  $E_{rel}$ : Relative frame energy
- 32 •  $\bar{N}_f$ : Long-term average noise energy
- 33 •  $d_i, \bar{d}_{k-1}$ : The open-loop pitch lags in current and previous frames and the signal modification  
 34 pitch delay parameter at the frame end of previous frame boundary or the close-loop pitch of  
 35 the last subframe.
- 36 •  $\hat{s}_w(n)$ : The weighted synthesized signal from the previous frame

37 **Outputs:**

- 38 •  $s(n)$ : The modified speech signal
- 39 • Encoding scheme

- $\bar{d}_k$  : The signal modification pitch delay parameter at the frame end boundary

**Initialization:**

- The buffers and filter memories are reset at initialization. The previous frame pitch values are initialized to 50. The last pitch pulse position  $T_0$  in the previous frame is initialized to  $-10$  with respect to the beginning of the current frame.

The signal modification algorithm performs an inherent classification of voiced frames. It is used only if the current frame has not been classified so far, that is if it has not been classified as inactive speech frame or Unvoiced frame. The signal is actually modified only if the classification procedure selected the Voiced HR encoding type. This encoding type is generally used only in VMR-WB modes 1 and 2 and under the condition that the previous frame type is FR or Voiced HR. In VMR-WB mode 0, Voiced HR is not used in normal operation. However, in maximum half-rate operation, VMR-WB mode 0 can use the Voiced HR encoding type in case of stable voiced frames.

Signal modification is adjusting the speech signal LP residual to the determined delay contour  $\tilde{d}(n)$ .

The delay contour  $\tilde{d}(n)$  defines a long-term prediction delay for every sample of the frame. The delay contour is fully characterized over the frame  $k \in (t_{k-1}, t_k]$  by a delay parameter  $\bar{d}_k = \tilde{d}(t_k)$  and its previous value  $\bar{d}_{k-1} = \tilde{d}(t_{k-1})$  that are equal to the value of the delay contour at frame boundaries. For a frame  $k$  with size  $L=256$ , the frame boundaries are  $t_k = L-1$  and  $t_{k-1} = -1$ , corresponding to the last samples in the present frame and previous frame, respectively. The delay parameter is determined as a part of the signal modification procedure and encoded once per frame.

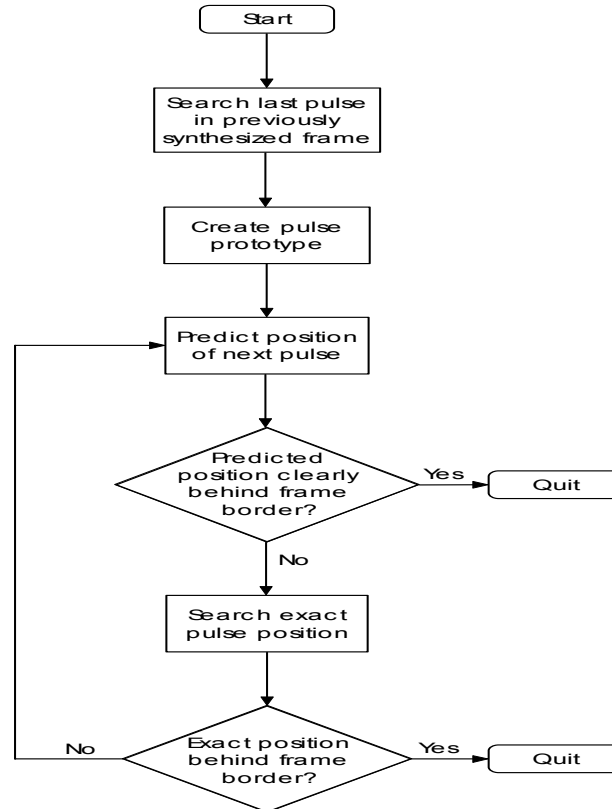
The signal modification is performed prior to the closed-loop pitch search of the adaptive codebook excitation signal that is prior to the ACELP subframe loop. The delay contour  $\tilde{d}(n)$  defining a long-term prediction delay parameter for every sample of the frame is supplied to an adaptive codebook. The delay contour  $\tilde{d}(n)$  is used to form the adaptive codebook excitation  $v(n)$  corresponding to the current subframe from the excitation  $u(n)$  using the delay contour  $\tilde{d}(n)$  as  $v(n) = u(n - \tilde{d}(n))$ . Thus the delay contour maps the past sample of the excitation signal  $u(n - \tilde{d}(n))$  to the present sample in the adaptive codebook excitation  $v(n)$ .

The signal modification procedure produces a modified residual signal  $\tilde{r}(n)$  to be used for composing a modified target signal for the closed-loop search of the fixed-codebook excitation  $c(n)$ . The modified residual signal  $\tilde{r}(n)$  is obtained by shifting the pitch cycle segments of the LP residual signal. The LP synthesis filtering of the modified residual signal with the filter  $1/A(z)$  then yields the modified speech signal. The modified target signal of the fixed-codebook excitation search is formed in accordance with the ACELP operation, but with the original speech signal replaced by its modified version. After the adaptive codebook excitation  $v(n)$  and the modified target signal have been obtained for the current subframe, the encoding can further proceed using the methods described in the next sections.

When signal modification is enabled, the speech decoder recovers the delay contour  $\tilde{d}(n)$  using the received delay parameter  $\bar{d}_k$  and its previous received value  $\bar{d}_{k-1}$  as in the encoder. This delay contour  $\tilde{d}(n)$  defines a long-term prediction delay parameter for every time instant of the current frame. The adaptive codebook excitation  $v(n) = u(n - \tilde{d}(n))$  is formed from the past excitation for the current subframe as in the encoder using the delay contour  $\tilde{d}(n)$ . The next section provides the details of the signal modification procedure as well as its use as part of the rate determination algorithm.

### 5.11.1 Search of Pitch Pulses and Pitch Cycle Segments

The signal modification scheme synchronously operates on pitch and frame by shifting each detected pitch cycle segment individually while constraining the shift at frame boundaries. This requires a mechanism for locating pitch pulses and corresponding pitch cycle segments for the current frame. Pitch cycle segments are determined based on detected pitch pulses that are searched according to Figure 5.11-1.



**Figure 5.11-1: Functional block diagram of pitch pulse search.**

Pitch pulse search operates on the weighted speech signal  $s_w(n)$  and the weighted synthesized speech signal  $\hat{s}_w(n)$ . It should be noted that the weighted speech signal  $s_w(n)$  is needed also for the lookahead in order to search the last pitch pulse in the current frame. This is done by using the weighting filter formed in the last subframe of the current frame over the lookahead portion.

The pitch pulse search procedure of Figure 5.11-1 starts by locating the last pitch pulse of the previous frame from the residual signal  $r(n)$ . A pitch pulse typically can be clearly distinguished as the maximum absolute value of the low-pass filtered residual signal in a pitch cycle having a length of approximately  $p_{k-1}$ , where  $p_{k-1}$  is the estimated pitch value at the frame end ( $p_{k-1} = d_{-1}$ ) in case of successful signal modification in the previous frame, or the closed-loop pitch estimate for the last subframe of the previous frame ( $p_{k-1} = \bar{d}_{k-1}$ ). A normalized Hamming window  $H_5(z) = (0.08z^{-2} + 0.54z^{-1} + 1 + 0.54z + 0.08z^2)/2.24$  having a length of five samples is used for the low-pass filtering in order to efficiently locate the last pitch pulse of the previous frame. This pitch pulse position is denoted by  $T_0$ . The signal modification method does not require an accurate position for this pitch pulse, but rather a rough location estimate of the high-energy segment in the pitch cycle.

After locating the last pitch pulse at  $T_0$  in the previous frame, a pitch pulse prototype of length 21 samples is extracted around this rough position estimate as

$$m_n(i) = \hat{s}_w(T_0 - 10 + i) \quad \text{for } i = 0, 1, \dots, 20. \quad (5.11.1-1)$$

This pitch pulse prototype is subsequently used in locating pitch pulses in the current frame.

The synthesized weighted speech signal  $\hat{s}_w(n)$  is used for the pulse prototype instead of the residual signal  $r(n)$ . This facilitates pitch pulse search because the periodic structure of the signal is better preserved in the weighted speech signal. The synthesized weighted speech signal  $\hat{s}_w(n)$  is obtained by filtering the synthesized speech signal  $\hat{s}(n)$  of the previous frame by the weighting filter  $W(z)$ . If the pitch pulse prototype extends over the end of the previously synthesized frame, the weighted speech signal  $s_w(n)$  of the current frame is used for this exceeding portion. The pitch pulse prototype has a high correlation with the pitch pulses of the weighted speech signal  $s_w(n)$  if the previous synthesized speech frame contains already a well-developed pitch cycle. Thus the use of the synthesized speech in extracting the prototype provides additional information for monitoring the performance of coding and for selecting an appropriate encoding method in the current frame as will be explained in more detail in the following description.

Given the position  $T_0$  of the last pulse in the previous frame, the first pitch pulse of the current frame can be predicted to occur approximately at instant  $T_0 + p(T_0)$ . Here  $p(t)$  denotes the interpolated open-loop pitch estimate at instant (position)  $t$ , that is

$$p(t) = p_{t_0} + (p_{t_1} - p_{t_0}) \frac{t}{128},$$

where

$$\begin{aligned} p_{t_0} &= p_{k-1}, p_{t_1} = d_0 \text{ for } 0 \leq t < 128 \\ p_{t_0} &= d_0, p_{t_1} = d_1 \text{ for } 128 \leq t < 256 \\ p_{t_0} &= d_1, p_{t_1} = d_2 \text{ for } 256 \leq t. \end{aligned}$$

$p_{k-1}$  denotes the estimated pitch value at the frame end in case of successful signal modification in the previous frame, or the closed-loop pitch estimate for the last subframe of the previous frame, and  $d_0, d_1, d_2$  are the open-loop pitch estimates for 2 half-frames of the current frame and the look-ahead. Note that  $t=0$  corresponds to the first sample of the current frame.

The predicted pitch pulse position  $T_0 + p(T_0)$  is then refined as

$$T_1 = T_0 + p(T_0) + \arg \max C(j), \quad (5.11.1-2)$$

where  $C(j)$  is the weighted correlation between weighted speech signal  $s_w(n)$  in the neighborhood of the predicted position and the pulse prototype

$$C(j) = \gamma(j) \sum_{k=0}^{20} m_n(k) s_w(T_0 + p(T_0) + j - 10 + k), \quad j \in [-j_{\max}, j_{\max}]. \quad (5.11.1-3)$$

Thus the refinement is the argument  $j$ , limited into  $[-j_{\max}, j_{\max}]$ , that maximizes the weighted correlation  $C(j)$ . The limit  $j_{\max}$  is proportional to the open-loop pitch estimate as  $\min\{20, \langle p(0)/4 \rangle\}$ , where the operator  $\langle \cdot \rangle$  denotes rounding to the nearest integer. The weighting function

$$\gamma(j) = 1 - |j| / p(T_0 + p(T_0)) \quad (5.11.1-4)$$

Equation (5.11.1-3) favors the pulse position predicted using the open-loop pitch estimate, since  $\gamma(j)$  attains its maximum value 1 at  $j = 0$ . The denominator  $\rho(T_0 + \rho(T_0))$  in Equation (5.11.1-4) is the interpolated open-loop pitch estimate for the predicted pitch pulse position.

After the first pitch pulse position  $T_1$  has been found using Equation (5.11.1-2), the next pitch pulse can be predicted to be at instant  $T_2 = T_1 + \rho(T_1)$  and refined as described above. This pitch pulse search comprising the prediction and refinement is repeated until either the prediction or refinement procedure yields a pitch pulse position outside the current frame. It should be noted that the pitch pulse prediction terminates the search only if a predicted pulse position is extremely far in the subsequent frame that the refinement step cannot bring it back to the current frame. This procedure yields  $c$  pitch pulse positions inside the current frame, denoted by  $T_1, T_2, \dots, T_c$ .

Pitch pulses are located with the integer resolution except the last pitch pulse of the frame denoted by  $T_c$ . Since the exact distance between the last pulses of two successive frames is needed to determine the delay parameter to be transmitted, the last pulse position is refined with a fractional resolution of  $1/4$  sample by maximizing the following correlation function for  $i, j$ .

$$C(j, i) = \sum_{i=-16}^{16} s_w(T_0 + j + i) s_w(T_c + i/4 + i), \quad j \in [-2, 1], i \in [0, 3] \quad (5.11.1-5)$$

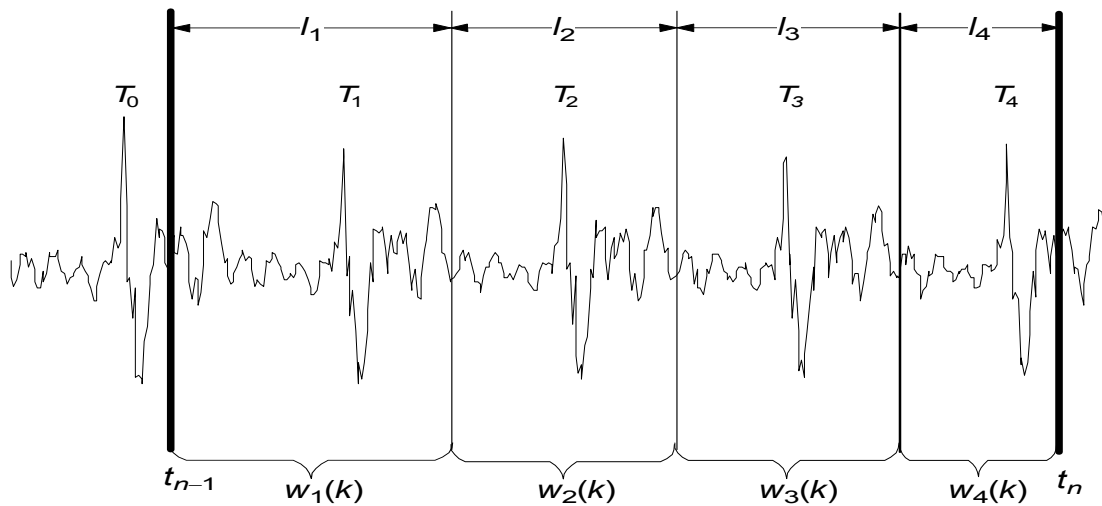
The fractional resolution is obtained by up-sampling the weighted speech signal  $s_w(n)$  in the neighborhood of the last predicted pitch pulse before evaluating the correlation function. Hamming-windowed sinc interpolation of length 33 is used for up-sampling. The fractional resolution of the last pitch pulse position helps to maintain a good performance of the long-term prediction despite the time synchrony constraint set to the frame end.

After completing pitch cycle segmentation in the current frame, an optimal shift for each segment is calculated. This operation is done using the weighted speech signal  $s_w(n)$  as will be explained in the following description. For reducing the distortion, the shifts of individual pitch cycle segments are implemented using the LP residual signal  $r(n)$ . Since shifting distorts the signal particularly around segment boundaries, it is essential to place the boundaries in low power sections of the residual signal  $r(n)$ . The segment boundaries are placed approximately in the middle of two consecutive pitch pulses, but constrained inside the current frame. Segment boundaries are always selected inside the current frame such that each segment contains exactly one pitch pulse. Segments with more than one pitch pulse or "empty" segments without any pitch pulses restrain subsequent correlation-based matching with the target signal and should be prevented in pitch cycle segmentation. The  $s$ th extracted segment of  $l_s$  samples is denoted as  $w_s(i)$  for  $i = 0, 1, \dots, l_s - 1$ . The starting instant of this segment is  $t_s$ , selected such that  $w_s(0) = s_w(t_s)$ . The number of segments in the present frame is denoted by  $c$ .

While selecting the segment boundary between two successive pitch pulses  $T_s$  and  $T_{s+1}$  inside the current frame, the following procedure is used. First the central instant between two pulses is computed as  $A = \langle (T_s + T_{s+1})/2 \rangle$ . The candidate positions for the segment boundary are located in the region  $[A - \varepsilon_{\max}, A + \varepsilon_{\max}]$ , where  $\varepsilon_{\max}$  corresponds to five samples. The energy of each candidate boundary position is computed as

$$Q(\varepsilon) = 0.75r^2(\Lambda + \varepsilon^l - 1) + r^2(\Lambda + \varepsilon^l) + 0.75r^2(\Lambda + \varepsilon^l - 1), \quad \varepsilon^l \in [-\varepsilon_{\max}, \varepsilon_{\max}] \quad (5.11.1-5)$$

The position corresponding to the smallest energy is selected because this choice typically results in the smallest distortion in the modified speech signal. The instant that minimizes Equation (5.11.1-5) is denoted as  $\varepsilon$ . The starting instant of the new segment is selected as  $t_s = A + \varepsilon + 1$ . This defines also the length of the previous segment, since the previous segment ends at instant  $A + \varepsilon$ .



1  
2 **Figure 5.11-2: Illustrative example on located pitch pulse positions and the corresponding**  
3 **pitch cycle segmentation for one frame**

#### 4 **5.11.2 Determination of the Delay Parameter**

5  
6 The main advantage of signal modification is that only one delay parameter per frame has to be  
7 encoded and transmitted to the decoder. However, special attention has to be paid to the  
8 determination of this single parameter. The delay parameter  $\bar{d}_k$  not only defines together with its  
9 previous value the evolution of the pitch cycle length over the frame, but also affects the time  
10 synchrony in the resulting modified signal.

11  
12 In the TIA/EIA/IS-127 codec [8, 21], no time synchrony is required at frame boundaries, and thus the  
13 delay parameter to be transmitted can be directly determined using an open-loop pitch estimate. This  
14 selection usually results in a time asynchrony at the frame boundary and translates into an  
15 accumulating time shift in the subsequent frame because the signal continuity has to be preserved. In  
16 the VMR-WB codec, the signal modification method preserves the time synchrony at frame  
17 boundaries. Thus, a strictly constrained shift occurs at the frame boundaries and every new frame  
18 starts in perfect time match with the original speech frame.

19  
20 To ensure time synchrony at the frame boundaries, the delay contour  $\tilde{d}(n)$  maps, with the long-term  
21 prediction, the last pitch pulse at the end of the previous synthesized speech frame to the last pitch  
22 pulse of the current frame. The delay contour defines an interpolated long-term prediction delay  
23 parameter over the current  $k$ th frame for every sample from instant  $t_{k-1} + 1$  through  $t_k$  (0 to  $L-1$ ). Only  
24 the delay parameter  $\bar{d}_k = \tilde{d}(t_k) = \tilde{d}(L-1)$  at the frame end is transmitted to the decoder implying  
25 that  $\tilde{d}(n)$  must have a form fully specified by the transmitted values. The long-term prediction delay  
26 parameter has to be selected such that the resulting delay contour fulfills the pulse mapping. In a  
27 mathematical form, this mapping can be presented as follows: Let  $\kappa_c$  be a temporary time variable  
28 and  $T_0$  and  $T_c$  the last pitch pulse positions in the previous and the current frames, respectively. The  
29 delay parameter  $\bar{d}_k$  has to be selected such that, after executing the pseudo-code presented in  
30 Table 5.11-1, the variable  $\kappa_c$  has a value very close to  $T_0$  minimizing the error  $e_n = |\kappa_c - T_0|$ . The  
31 pseudo-code starts from the value  $\kappa_0 = T_c$  and iterates backwards  $c$  times by updating  $\kappa_j := \kappa_{j-1} -$   
32  $\tilde{d}(\kappa_{j-1})$ . If  $\kappa_c$  then equals to  $T_0$ , long-term prediction can be utilized with maximum efficiency without  
33 time asynchrony at the frame end. In practice, the tolerated error  $e_n$  is limited to 1.  
34

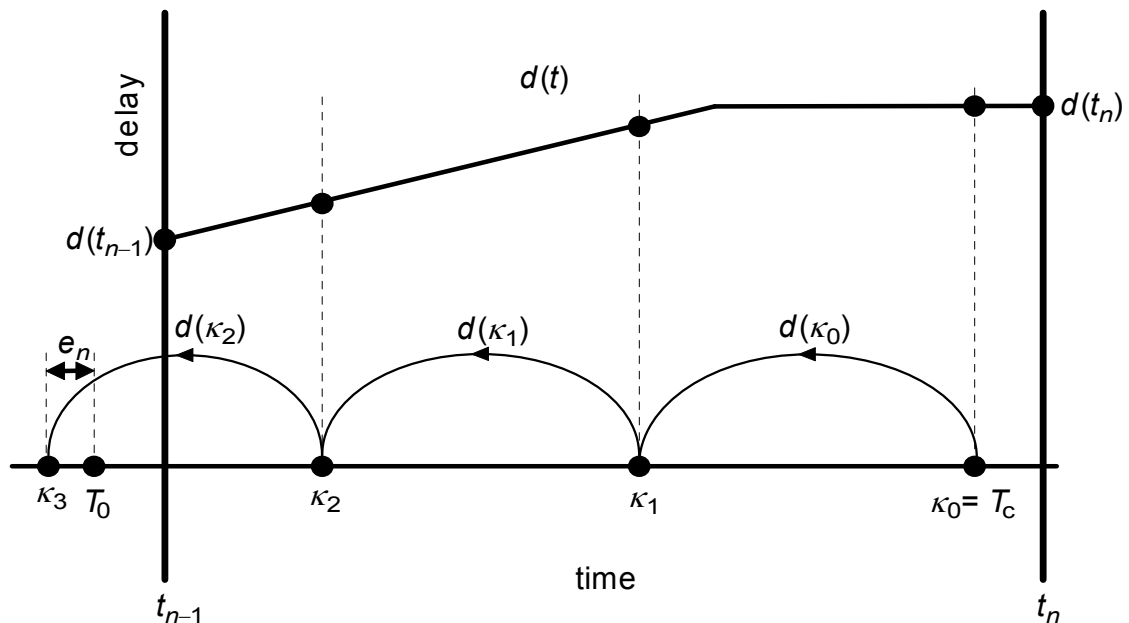
Table 5.11-1: Loop for searching the optimal delay parameter

```

1
2
3
4 % Initialization
5  $\kappa_0 := T_c$ ;
6
7 % Loop
8 for  $i = 1$  to  $c$ 
9    $\kappa_i := \kappa_{i-1} - \tilde{d}(\kappa_{i-1})$ ;
10 end;
11
12

```

13 An example of the operation of the delay selection loop in the case  $c = 3$  is illustrated in Figure  
14 5.11-3. The loop starts from the value  $\kappa_0 = T_c$  and takes the first iteration backwards as  $\kappa_1 = \kappa_0 -$   
15  $\tilde{d}(\kappa_0)$ . Iterations are continued twice more resulting in  $\kappa_2 = \kappa_1 - \tilde{d}(\kappa_1)$  and  $\kappa_3 = \kappa_2 - \tilde{d}(\kappa_2)$ . The final  
16 value  $\kappa_3$  is then compared against  $T_0$  in terms of the error  $e_n = |\kappa_3 - T_0|$ . The resulting error is a  
17 function of the delay contour that is computed in the delay selection algorithm as will be shown later  
18 in this specification.  
19



20  
21 **Figure 5.11-3: Illustrative example on the operation of the delay selection procedure when the**  
22 **number of pitch pulses is three ( $c = 3$ ).**

23  
24 The signal modification method in the TIA/EIA/IS-127 codec [8] interpolates the delay parameters  
25 linearly over the frame between  $\bar{d}_{k-1}$  and  $\bar{d}_k$ . However, when time synchrony is required at the  
26 frame end, linear interpolation tends to result in an oscillating delay contour. Thus pitch cycles in the  
27 modified speech signal contract and expand periodically causing annoying audible artifacts. The  
28 evolution and amplitude of the oscillations are related to the last pitch position. The farther the last  
29 pitch pulse is from the frame end in relation to the pitch period, the more likely the oscillations are  
30 amplified. Therefore, a piecewise linear delay contour is used as follows:  
31  
32

$$\tilde{d}(n) = \begin{cases} (1 - \alpha(n))\bar{d}_{k-1} + \alpha(n)\bar{d}_k, & t_{k-1} < n \leq t_{k-1} + \sigma_k \\ \bar{d}_k, & t_{k-1} + \sigma_k < n < t_k \end{cases} \quad (5.11.2-1)$$

where

$$\alpha(n) = (n) / \sigma_k \quad (5.11.2-2)$$

The oscillations are significantly reduced by using this delay contour. Here  $t_n$  and  $t_{n-1}$  are the end instants of the current and previous frames, respectively, and  $\bar{d}_k$  and  $\bar{d}_{k-1}$  are the corresponding delay parameter values. Note that  $t_{k-1} + \sigma_k$  ( $\sigma_k - 1$ ) is the instant after which the delay contour remains constant. The parameter  $\sigma_k$  varies as a function of  $\bar{d}_{k-1}$  as

$$\sigma_k = \begin{cases} 172 \text{ samples,} & \bar{d}_{k-1} \leq 90 \text{ samples} \\ 128 \text{ samples,} & \bar{d}_{k-1} > 90 \text{ samples} \end{cases} \quad (5.11.2-3)$$

and the frame length  $L$  is 256 samples. To avoid oscillations, it is beneficial to decrease the value of  $\sigma_k$  as the length of the pitch cycle increases. On the other hand, to avoid rapid changes in the delay contour  $\tilde{d}(n)$  in the beginning of the frame ( $n < \sigma_k$ ), the parameter  $\sigma_k$  has to be always at least a half of the frame length. Rapid changes in  $\tilde{d}(n)$  degrade easily the quality of the modified speech signal.

Note that depending on the encoding method of the previous frame  $\bar{d}_{k-1}$  can be either the delay value at the frame end (signal modification enabled) or the close-loop delay value of the last subframe (signal modification disabled). Since the past value  $\bar{d}_{k-1}$  of the delay parameter is known at the decoder, the delay contour is unambiguously defined by  $\bar{d}_k$ , and the decoder is able to form the delay contour using Equation (5.11.2-1).

The only parameter, which can be varied while searching the optimal delay contour, is  $\bar{d}_k$ , the delay parameter value at the end of the frame constrained into [34, 231] samples. There is no simple explicit method for solving the optimal  $\bar{d}_k$  in a general case. Instead, several values have to be tested to find the best solution. However, the search is straightforward. The value of  $\bar{d}_k$  is first predicted as

$$\bar{d}_k^{(0)} = 2 \frac{T_c - T_0}{c} - \bar{d}_{k-1} - 2. \quad (5.11.2-4)$$

Then, the search is conducted in three phases by increasing the resolution and focusing the search range to be examined inside [34, 231] in every phase. The delay parameters giving the smallest error  $e_n = |\kappa_c - T_0|$  in the procedure of Table 5.11-1 in these three phases are denoted by  $\bar{d}_k^{(1)}$ ,  $\bar{d}_k^{(2)}$ , and  $\bar{d}_k = \bar{d}_k^{(3)}$ , respectively. In the first phase, the search is done around the value  $\bar{d}_k^{(0)}$  predicted using Equation (5.11.2-4) with a resolution of 4 samples in the range  $[\bar{d}_k^{(0)} - 11, \bar{d}_k^{(0)} + 12]$  when  $\bar{d}_k^{(0)} < 60$ , and in the range  $[\bar{d}_k^{(0)} - 15, \bar{d}_k^{(0)} + 16]$  otherwise. The second phase constrains the range into  $[\bar{d}_k^{(1)} - 3, \bar{d}_k^{(1)} + 3]$  and uses the integer resolution. The last, third phase examines the range  $[\bar{d}_k^{(2)} - 3/4, \bar{d}_k^{(2)} + 3/4]$  with a resolution of  $1/4$  sample for  $\bar{d}_k^{(2)} < 92^{1/2}$ . Above that range  $[\bar{d}_k^{(2)} - 1/2,$

1  $\bar{d}_k^{(2)} + 1/2]$  and a resolution of  $1/2$  sample is used. This third phase yields the optimal delay parameter  
 2  $\bar{d}_k$  to be transmitted to the decoder. This procedure is a compromise between the search accuracy  
 3 and complexity.

4  
 5 The delay parameter  $\bar{d}_k \in [34, 231]$  can be coded using nine bits per frame using a resolution of  $1/4$   
 6 sample for  $\bar{d}_k < 92^{1/2}$  and  $1/2$  sample for  $\bar{d}_k > 92^{1/2}$ .

### 7 8 **5.11.3 Modification of the Signal** 9

10 After the delay parameter  $\bar{d}_k$  and the pitch cycle segmentation have been determined, the signal  
 11 modification procedure itself can be initiated. The speech signal is modified by shifting individual pitch  
 12 cycle segments one by one adjusting them to the delay contour  $\tilde{d}(n)$ . A segment shift is determined  
 13 by correlating the segment in the weighted speech domain with a target signal. The target signal is  
 14 composed using the synthesized weighted speech signal  $\hat{s}_w(n)$  of the previous frame and the  
 15 preceding, already shifted segments in the current frame, together with the delay contour  $\tilde{d}(n)$ . The  
 16 actual shift is done on the residual signal  $r(n)$ .

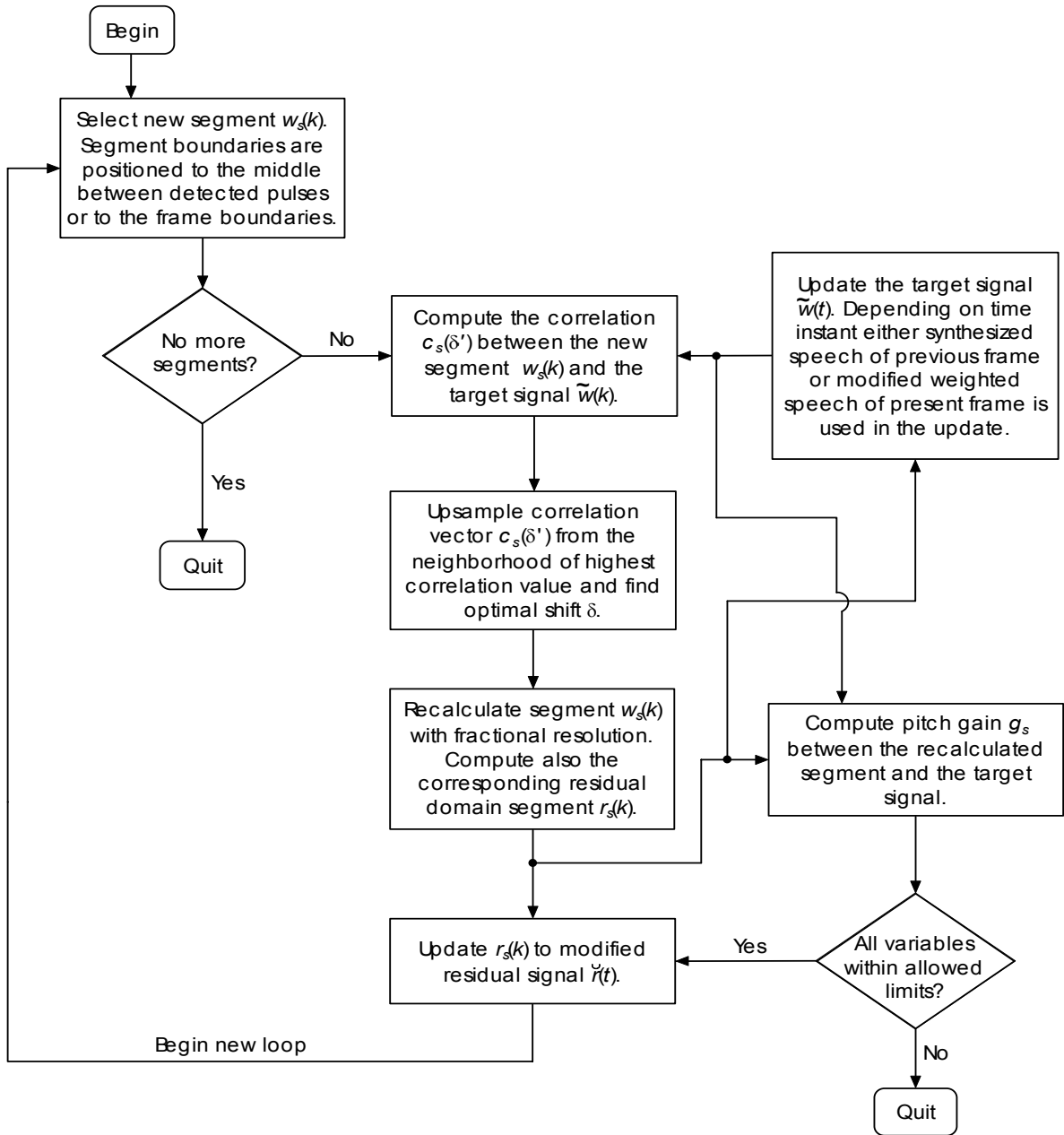
17  
 18 Signal modification has to be done carefully to both maximize the performance of long-term prediction  
 19 and simultaneously to preserve the perceptual quality of the modified speech signal. The required  
 20 time synchrony at frame boundaries has to be taken into account also during modification.

21  
 22 A block diagram of the signal modification method is shown in Figure 5.11-4. Modification starts by  
 23 extracting a new segment  $w_s(n)$  of  $l_s$  samples from the weighted speech signal  $s_w(n)$ . This segment is  
 24 defined by the segment length  $l_s$  and starting instant  $t_s$  giving  $w_s(n) = w(t_s + n)$  for  $k = 0, 1, \dots, l_s - 1$ .  
 25 The segmentation procedure shall be performed in accordance with the foregoing description.

26  
 27 For finding the optimal shift of the current segment  $w_s(n)$ , a target signal  $\tilde{w}(n)$  is created. For the first  
 28 segment  $w_1(k)$  in the current frame, this target signal is obtained by the recursion

$$\begin{aligned}
 \tilde{w}(n) &= \hat{s}_w(n), & n < 0 \\
 \tilde{w}(n) &= \tilde{w}(n - \tilde{d}(n)), & n = 0, \dots, l_1 + \delta_s - 1.
 \end{aligned}
 \tag{5.11.3-1}$$

31  
 32 Here  $\hat{s}_w(n)$  is the weighted synthesized speech signal available in the previous frame for  $n \leq t_{k-1}$ . The  
 33 parameter  $\delta_s = 5$  is the range of the shift search and  $l_1$  is the length of the first segment. Equation  
 34 (5.11.3-1) can be interpreted as simulation of long-term prediction using the delay contour over the  
 35 signal portion in which the current shifted segment may potentially be situated. The computation of  
 36 the target signal for the subsequent segments follows the same principle and will be presented later  
 37 in this section.

1  
2

3 **Figure 5.11-4: Functional block diagram on the signal modification procedure that adjusts the**  
4 **speech frame to the selected delay contour**

5 The search procedure for finding the optimal shift of the current segment can be initiated after forming  
6 the target signal. This procedure is based on the correlation  $c_s(\delta')$  between the segment  $w_s(n)$  that  
7 starts at instant  $t_s$  and the target signal  $\tilde{w}(n)$  as

8

$$9 \quad c_s(\delta') = \sum_{k=0}^{l_s-1} w_s(k) \tilde{w}(k + t_s + \delta'), \quad \delta' \in [-\delta_s, \delta_s], \quad (5.11.3-2)$$

10

11 After the integer shift  $\delta$  that maximizes the correlation  $c_s(\delta')$  in (5.11.3-2) is found, the maximum  
12 correlation value is searched with a fractional resolution in the open interval  $(\delta - 1, \delta + 1)$ , and

1 bounded into  $[-\delta_s, \delta_s]$ . The correlation  $c_s(\delta')$  is up-sampled in this interval to a resolution of 1/8  
 2 sample using Hamming-windowed sinc interpolation of a length equal to 65 samples. The shift  $\delta$   
 3 corresponding to the maximum value of the up-sampled correlation is then the optimal shift with a  
 4 fractional resolution. After finding this optimal shift, its fractional part is incorporated into the weighted  
 5 speech segment  $w_s(n)$ . That is, the precise new starting instant of the segment is updated as  $t_s := t_s -$   
 6  $\delta + \delta_l$ , where  $\delta_l = \lceil \delta \rceil$  is the upward-rounded shift. Similarly, the fractional part of the optimal shift is  
 7 also incorporated into the residual segment  $r_s(n)$  corresponding to the weighted speech segment  
 8  $w_s(n)$  and computed from the residual signal  $r(n)$  using again the sinc interpolation as described  
 9 before. Since the fractional part of the optimal shift is now incorporated into the residual and weighted  
 10 speech segments, all subsequent computations can be implemented with  $\delta_l = \lceil \delta \rceil$ .

11  
 12 Even if the optimal shift search range is  $[-5,5]$  samples, the actual maximum permitted shift varies  
 13 with the pitch period. The following values are used for  $\delta_s$ :

$$14 \quad \delta_s = \begin{cases} 4^{1/2} \text{ samples,} & \bar{d}_k < 90 \text{ samples} \\ 5 \text{ samples,} & \bar{d}_k \geq 90 \text{ samples} \end{cases} \quad (5.11.3-3)$$

16  
 17 If the correlation function is maximized for  $\delta > \delta_s$ , the signal modification is aborted and Voiced HR  
 18 encoding type is not used. As will be described later in this section, the value of  $\delta_s$  is even more  
 19 limited for the first and the last segment in the frame.

20  
 21 The final task is to update the modified residual signal  $\tilde{r}(n)$  by copying the current residual signal  
 22 segment  $r_s(n)$  into it:

$$23 \quad \tilde{r}(t_s + \delta_l + n) = r_s(n), \quad n = 0, 1, \dots, l_s - 1 \quad (5.11.3-4)$$

25  
 26 Since shifts in successive segments are independent from each other, the segments positioned to  
 27  $\tilde{r}(n)$  either overlap or have a gap in between them. Straightforward weighted averaging is used for  
 28 overlapping segments, that is for negative  $\delta_l$

$$29 \quad \tilde{r}(t_s + \delta_l + k) = \left(1 - \frac{k+1}{|\delta_l|}\right) \tilde{r}(t_s + \delta_l + k) + \left(\frac{k+1}{|\delta_l|}\right) r_s(k), \quad k=0, \dots, |\delta_l|-1 \quad (5.11.3-5)$$

31  
 32 The remaining segment samples are simply copied following (5.11.3-4). The gaps are filled with  
 33 zeros. There are only 2 exceptions. The first exception is the case of the first segment when only one  
 34 sample is missing, that is the fractional shift  $\delta$  lower than 1 has been selected. In this case the  
 35 residual signal  $r(n)$  is up-sampled with 1/8 sample resolution around the instant  $t_{k-1}$  and the gap is  
 36 filled with the sample closest to  $r(t_{k-1} + (\delta_l - \delta)/2)$ . The second exception is when some samples are  
 37 missing at the end of the last segment. In this case, the last available sample is simply repeated until  
 38 the frame ends. Since the number of overlapping or missing samples is usually small and the  
 39 segment boundaries occur at low-energy regions of the residual signal, usually no perceptual artifacts  
 40 are caused.

41  
 42 Processing of the subsequent pitch cycle segments follows the above-mentioned procedure, except  
 43 that the target signal  $\tilde{w}(t)$  is formed differently than for the first segment. The samples of  $\tilde{w}(n)$  are  
 44 first replaced with the modified weighted speech samples as

$$45 \quad \tilde{w}(t_s + \delta_l + n) = w_s(n), \quad n = 0, 1, \dots, l_s - 1 \quad (5.11.3-6)$$

47  
 48 When updating the target signal following (5.11.3-6), the segments can either overlap or have a gap  
 49 in between them similarly as explained above for the residual signal. Overlapping segments in the  
 50 target signal are processed exactly the same way as in the modified residual signal. If samples are

1 missing (positive  $\delta_l$ ), a linear interpolation is used between the last available sample in  $\tilde{w}(n)$  and the  
 2 first sample of  $w_s(n)$

$$3 \quad \tilde{w}(t_s+k) = \left(1 - \frac{k+1}{|\delta_l|}\right)\tilde{w}(t_s-1) + \left(\frac{k+1}{|\delta_l|}\right)w_s(0), \quad k = 0, \dots, |\delta_l|-1 \quad (5.11.3-7)$$

5 Again, the case of the first segment with only one missing sample is processed differently. The  
 6 missing sample is simply the average of  $\tilde{w}(t_{n-1})$  (the last sample of the previous frame) and  $w_s(0)$ .  
 7 Then the samples following the updated segment are also updated,

$$8 \quad \tilde{w}(n) = \tilde{w}(n-\tilde{d}(k)), \quad n = t_s + \delta_l + l_s, \dots, t_s + \delta_l + l_s + l_{s+1} + 4 \quad (5.11.3-8)$$

10 The update of the target signal  $\tilde{w}(n)$  ensures higher correlation between successive pitch cycle  
 11 segments in the modified speech signal following the delay contour  $\tilde{d}(n)$  and thus more accurate  
 12 long-term prediction. While processing the last segment of the frame, the target signal  $\tilde{w}(n)$  does not  
 13 need to be updated.

14 The shifts of the first and the last segments in the frame are special cases. Before shifting the first  
 15 segment, it should be ensured that no high power regions exist in the residual signal  $r(n)$  close to the  
 16 frame boundary  $t_{k-1}$ , because shifting such a segment may cause artifacts. The high power region is  
 17 searched by squaring the first pitch period of the residual signal  $r(n)$  as

$$18 \quad E_0(i) = r^2(i), \quad j = 0, \dots, \bar{d}_{k-1} - 1 \quad (5.11.3-9)$$

20 where  $\bar{d}_{k-1}$  is the pitch period at the end of the previous frame as described in Section 5.11.2. If the  
 21 maximum of  $E_0(i)$  is detected close to the frame boundary in the range  $[t_{k-1} + 1, t_{k-1} + 2]$ , the  
 22 allowed shift is limited to  $\frac{1}{4}$  samples. The allowed shift is limited to  $\frac{1}{4}$  samples also if one of the pitch  
 23 pulse positions  $T_0$  or  $T_1$  as described in Section 5.11.1 are found in the range  $[t_{k-1}, t_{k-1} + 2]$ . If the  
 24 proposed shift  $|\delta|$  for the first segment is smaller than this limit, the signal modification procedure is  
 25 enabled in the current frame, but the first segment is kept intact.

26 The last segment in the frame is processed in a similar manner. As was described in the foregoing  
 27 description, the delay contour  $\tilde{d}(n)$  is selected such that in principle no shifts are required for the  
 28 last segment. However, because the target signal is repeatedly updated during signal modification, it  
 29 is possible the last segment has to be shifted slightly. This shift is always constrained to be smaller  
 30 than  $3/2$  samples. The shift limit is further reduced to be smaller than  $0.5$  samples if the last pulse  
 31 position  $T_c$  is close to the frame boundary, that is

$$32 \quad T_c > t_{k+1} - \min\{10, 0.22 * \bar{d}_k\}$$

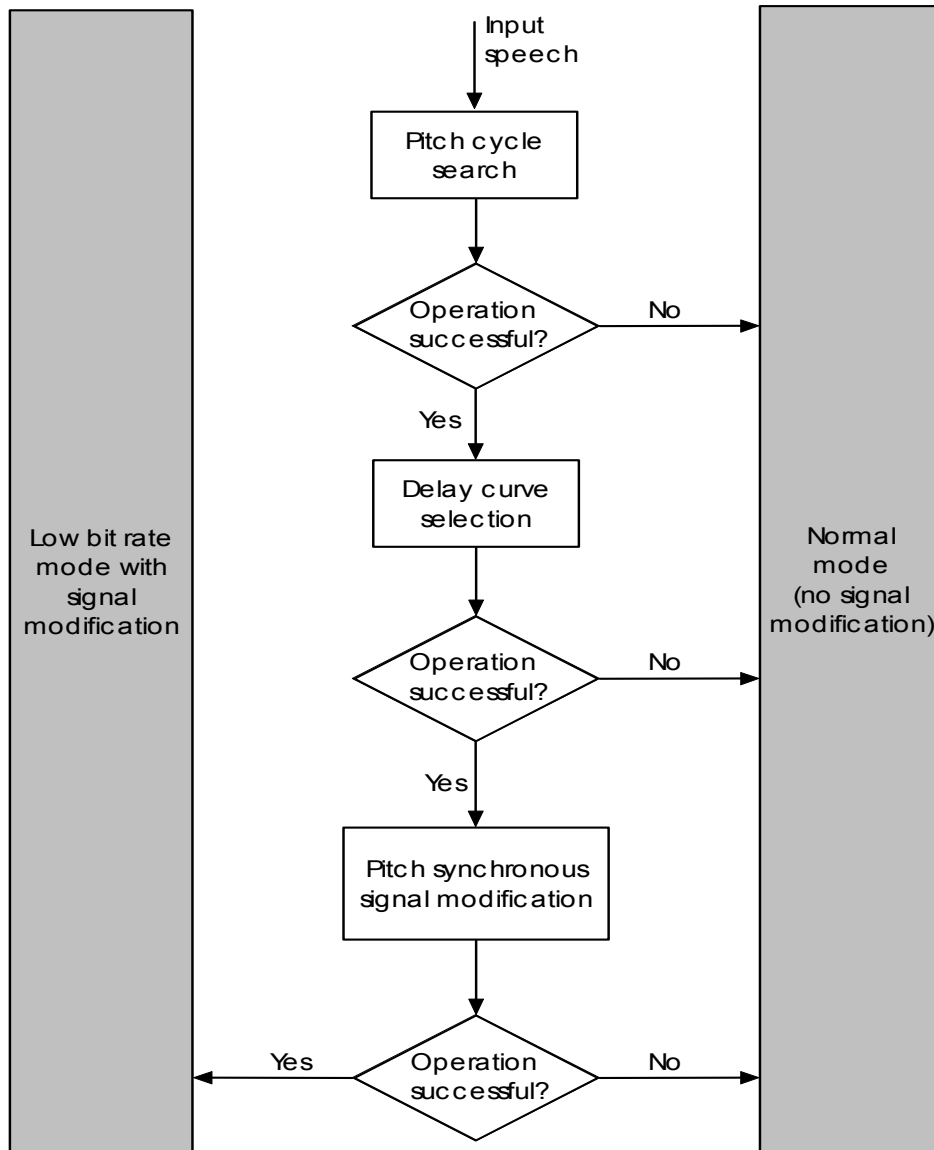
33 If the signal modification algorithm suggests larger shift than permitted, signal modification is aborted  
 34 and Generic FR encoding type is used.

35 If there is a high power region at the frame end, no shift is allowed at all. This condition is verified by  
 36 using the squared residual signal  $r^2(i)$ . If the maximum of  $r^2(i)$  evaluated for  $i \in [t_k+2-\bar{d}_k, t_k+2]$  is  
 37 attained for  $i$  larger than  $t_k$  or if the maximum of  $r^2(i)$  evaluated for  $i \in [t_k+7-\bar{d}_k, t_k+6]$  is attained for  $i$   
 38 larger than  $t_k-4$ , no shift is allowed for the last segment. Similarly as for the first segment, when the

1 proposed shift  $|\delta| < \frac{1}{4}$ , the present frame is still accepted for modification, but the last segment is  
 2 kept intact.  
 3  
 4 It should be noted that the shift does not translate into the next frame, and every new frame starts  
 5 perfectly synchronized with the original input signal.  
 6  
 7 If the signal modification procedure is successful and after determining the modified residual signal,  
 8 the modified speech signal  $s(n)$  is computed by filtering the residual signal through the synthesis filter  
 9  $1/A(z)$ . The modified signal  $s(n)$  replaces the input denoised signal in the subsequent processing for  
 10 computing the adaptive and algebraic codebook contributions.

11  
 12 **5.11.4 Voiced Classification Logic Incorporated into the Signal Modification Procedure**  
 13

14 The signal modification method incorporates an efficient classification and mode determination  
 15 mechanism as depicted in Figure 5.11-5. Every operation yields several indicators quantifying the  
 16 attainable performance of long-term prediction in the current frame. If any of these indicators are  
 17 outside their allowed limits, the signal modification procedure is terminated and the original signal is  
 18 preserved intact.  
 19



20

**Figure 5.11-5: Functional block diagram of closed loop voiced signal classification.**

The pitch pulse search procedure produces several indicators on the periodicity of the present frame. First, the signal modification is aborted immediately if the difference between the pitch at the end of the previous frame and the open-loop pitch estimate at the end of the current frame is significant, that is if

$$\begin{aligned} & \left| 4[\bar{d}_{k-1} - p(t_k)] \right| > \bar{d}_{k-1}, \quad \text{Or} \\ & \left| 4[\bar{d}_{k-1} - p(t_k)] \right| > p(t_k). \end{aligned}$$

$p(t_k)$  is the interpolated open loop pitch estimate for the last sample  $t_k$  in the current frame given by  $p(t_k) = 0.9921875 d_0 + 0.0078125 d_1$  where  $d_0$  and  $d_1$  are the open loop pitch estimates for the first and the second half-frame of the current frame, respectively (Section 5.8).

Second, if the difference between the detected pitch pulse positions and the interpolated open-loop pitch estimate is not within a permitted range, that is

$$|0.4[T_i - T_{i-1} - p(T_i)]| > 0.2 p(T_i), \quad i = 1, 2, \dots, c, \quad (5.11.4-1)$$

the signal modification procedure is aborted.

The selection of the delay contour  $\tilde{d}(n)$  provides additional information on the evolution of the pitch cycles and the periodicity of the current speech frame. The signal modification procedure is continued only if the condition  $|\bar{d}_k - \bar{d}_{k-1}| \leq \bar{\delta}_d \bar{d}_k$  is satisfied. In VMR-WB mode 2,  $\bar{\delta}_d$  varies linearly with the relative frame energy  $E_{rel}$  as

$$\bar{\delta}_d = -0.0429 * E_{rel} + 0.0714 \quad (5.11.4-2)$$

and is constrained between 0.2 and 0.5; otherwise  $\bar{\delta}_d$  equals 0.2. This condition means that only a small delay change is tolerated for classifying the current frame as purely voiced frame. The algorithm also evaluates the success of the delay selection loop of Table 5.11-1 by examining the difference  $|\kappa_c - T_0|$  for the selected delay parameter value  $\bar{d}_k$ . If this difference is greater than one sample, the signal modification procedure is terminated.

To ensure a good quality for the modified speech signal, it is advantageous to constrain the shifts performed for successive pitch cycle segments. This is achieved by imposing the criteria

$$\left| \delta^{(s)} - \delta^{(s-1)} \right| \leq \begin{cases} 4.0 & \text{samples, } \bar{d}_k \leq 90 \text{ samples} \\ 4.8 & \text{samples, } \bar{d}_k > 90 \text{ samples} \end{cases} \quad (5.11.4-3)$$

on all segments of the frame. Here  $\delta^{(s)}$  and  $\delta^{(s-1)}$  are the shifts done for the  $s$ th and  $(s-1)$ th pitch cycle segments, respectively. If the thresholds are exceeded, the signal modification procedure is interrupted and the original signal is maintained.

When the frames subjected to signal modification are encoded at a low bit rate, it is essential that the shape of pitch cycle segments remains similar over the frame. This allows faithful signal modeling by long-term prediction and thus encoding at a low bit rate without degrading the subjective quality. The similarity of successive segments can be quantified simply by the normalized correlation

$$g_s = \frac{\sum_{k=0}^{l_s-1} w_s(k) \tilde{w}(k + t_s + \delta_l)}{\sqrt{\sum_{k=0}^{l_s-1} w_s^2(k) \sum_{k=0}^{l_s-1} \tilde{w}^2(k + t_s + \delta_l)}} \quad (5.11.4-4)$$

between the current segment  $w_s(k)$  updated with the fractional part of the optimal shift, and the target signal  $\tilde{w}(n)$  at the optimal shift  $\delta_l$ . The success of the procedure is examined using the criteria  $g_s \geq \bar{\delta}_g$ . The threshold  $\bar{\delta}_g$  varies with the operational mode and the estimated pitch period. The reason is that the signal modification algorithm performs generally better for longer pitch periods.

If maximum half-rate is enforced for the current frame,  $\bar{\delta}_g = 0.5$ . In VMR-WB mode 2,  $\bar{\delta}_g$  varies linearly with  $E_{rel}$  to increase the Voiced HR usage in low energy frames: For  $d_n > 96$ ,  $\bar{\delta}_g = 0.45 * E_{rel} + 0.79$  and is upper-bounded to 0.79. For  $\bar{d}_k \leq 96$ ,  $\bar{\delta}_g = 0.05 * E_{rel} + 0.847$ , upper bounded to 0.847. In VMR-WB mode 1,  $\bar{\delta}_g = 0.91$  for  $\bar{d}_k > 100$  and  $\bar{\delta}_g = 0.948$  for  $\bar{d}_k \leq 100$ . The threshold  $\bar{\delta}_g$  has been used to adjust the selection of the Voiced HR encoding type to comply with the desired average bit rates. If the condition  $g_s \geq \bar{\delta}_g$  is not satisfied for all segments, the signal modification procedure is terminated and the original signal is kept intact.

If the signal modification is successfully performed, long-term prediction is able to model the modified speech frame efficiently, facilitating its encoding at a low bit rate without degrading subjective quality. In this case, the adaptive codebook excitation has a dominant contribution in describing the excitation signal, and thus the bit rate allocated for the fixed-codebook excitation can be reduced. Otherwise, the frame is likely to contain a non-stationary speech segment such as a voiced onset or rapidly evolving voiced speech signal. These frames typically require a high bit rate for maintaining good subjective quality.

#### 5.11.4.1 Voiced Classification in Rate-Set I Operation

In case of Rate-Set I operation, if signal modification is successful, RS-I Voiced FR is used. The goal is to improve the quality of voiced segments rather than reduce the average bit rate. Thus, encoding type RS-I Voiced HR is not used in normal operation and is only used when maximum half-rate is enforced. In case of maximum half-rate and if  $g_s \geq 0.5$  then RS-I Voiced HR is used. In Rate-Set I normal operation,  $\bar{\delta}_g = 0.75$  for  $\bar{d}_k > 100$  and  $\bar{\delta}_g = 0.85$  for  $\bar{d}_k \leq 100$ . Thus, if  $g_s \geq \bar{\delta}_g$ , RS-I Voiced FR is used otherwise no signal modification is performed.

## 5.12 Selection of FR and Generic HR, and Maximum and Minimum Rate Operation

**Routine Name:** rate\_select2

**Inputs:**

- Mode of operation and HR-maximum signaling flag
- $E_{rel}$ : Relative frame energy
- $\bar{N}_f$ : Long-term average noise energy
- Open-loop lag in the first half-frame
- Previous frame encoding type

**Outputs:**

- Encoding type

**Initialization:**

- None

If the rate selection reaches this stage, then the frame has not been declared as inactive speech frame (not encoded with CNG-ER or CNG-QR), nor declared as unvoiced frame (not encoded with Unvoiced HR or Unvoiced QR), nor declared as stable voiced frame (not encoded with Voiced HR). Thus the frame is likely to contain a non-stationary speech segment such as a voiced onset or rapidly evolving voiced speech signal. These frames typically require a general-purpose encoding model at high bit rate for maintaining good speech quality. Thus in this case an appropriate FR encoding type is mainly used. However, frames with very low energy and not detected as non-speech, unvoiced or stable voiced can be encoded using Generic HR encoding in order to reduce the average data rate. In the AMR-WB interoperable mode, Generic HR encoding type is not used and all frames at this stage are encoded with interoperable FR in normal operation. In VMR-WB mode 0, Generic HR encoding type is not used in normal operation and all frames at this stage are encoded with Generic FR. In VMR-WB modes 1 and 2, Generic HR encoding is used if the relative energy  $E_{rel}$  is lower than a certain threshold  $th_{14}$ .

In VMR-WB mode 1, the threshold is given as follows:

*If last frame type is Unvoiced HR then  $th_{14} = -14$  else  $th_{14} = -11$ .*

Furthermore, if the long-term average noise energy  $\bar{N}_f$  is larger than 21 then  $th_{14}$  is incremented by 2.7. In VMR-WB mode 2, the threshold is given as follows:

*If the pitch open-loop lag in the first half-frame is larger than 100 then  $th_{14} = -6$  else  $th_{14} = -5$ .*

### 5.12.1 Maximum and Minimum Rate Operation in Rate-Set II

Depending on the application, a maximum or a minimum bit rate for a particular frame can be forced. Most often, the maximum bit rate imposed by the system during signaling periods (e.g., dim and burst) is limited to HR. However, the system can impose also lower rates. In normal operation and by default the maximum rate is FR and the minimum rate is ER.

If minimum and maximum rates are different from the default values then the signal classification described above will take this into account while making the rate selection.

For minimum rate operation, if min-rate is FR then lower rates cannot be used and all frames are encoded with an appropriate FR encoding type.

If min-rate is HR then QR and ER cannot be used. In this case, in AMR-WB interoperable mode, inactive speech frames are encoded with Interoperable HR and in other modes inactive frames are encoded with Unvoiced HR. The rest of classification is as usual.

If min-rate is QR then in all modes, all inactive frames are encoded with CNG-QR. The rest of classification is as usual.

If min-rate is ER, then the rate selection works in normal operation. In this case in the AMR-WB interoperable mode, the first inactive (VAD\_flag=0) speech frame is encoded with CNG-QR (corresponding to SID\_UPDATE in AMR-WB) then the next  $k-1$  frames encoded with CNG-ER (those frames would be discarded at system interface with AMR-WB), then the  $k$ th frame is encoded with CNG-QR and so on, until active speech frames resume. In the VMR-WB codec,  $k$  is set to 8, that is CNG-QR is used every 8<sup>th</sup> frame. In the other modes, all inactive frames are encoded with CNG-ER.

1 In the case of maximum HR limitation, all active speech frames that would be classified as FR during  
 2 normal operation must be encoded using an appropriate HR encoding type. The classification and  
 3 rate selection mechanism are preformed as usual. That is determining inactive speech frames  
 4 (encoded with CNG-ER or CNG-QR), unvoiced frames (encoded with Unvoiced HR or QR), stable-  
 5 voiced frames (encoded with Voiced HR). All remaining frames that would be classified as FR during  
 6 normal operation are encoded using the Generic HR encoding type except in the AMR-WB  
 7 interoperable mode where an interoperable HR encoding type is used.

8  
 9 In the AMR-WB interoperable mode, the interoperable HR encoding type was designed to enable  
 10 interoperable interconnection between VMR-WB and AMR-WB in case of maximum half-rate  
 11 operation with minimal impact on the quality. Since there is no AMR-WB codec mode at 6.2 kbps or  
 12 below, the interoperable HR was designed by dropping some of the bits corresponding to the  
 13 algebraic codebook indices (see Table 4.2-1). After dropping the selected codebook indices, the  
 14 remaining encoding parameters, which represent the new encoder output, are transmitted to the  
 15 decoder along with a signal classification of I-HR. At the system interface with AMR-WB, the missing  
 16 fixed-codebook indices are randomly generated and the frame is decoded as an AMR-WB frame at  
 17 12.65, 8.85, or 6.60 kbps, depending on the AMR-WB codec mode.

18  
 19 A special feature in case of maximum HR operation is that the thresholds used to distinguish between  
 20 unvoiced and voiced frames are in general more relaxed to allow as many frames as possible to be  
 21 encoded using the Unvoiced HR and Voiced HR encoding types. In case of maximum HR operation,  
 22 in VMR-WB mode 0, the frame is declared an Unvoiced frame if  $local\_VAD=0$  or the following  
 23 condition is satisfied

$$24 \quad (\bar{R}_{xy3} < th_4) \text{ AND } (e_t < th_5) \text{ AND } (dE < th_6)$$

25  
 26  
 27  
 28 where  $th_4 = 0.695$ ,  $th_5 = 4$ ,  $th_6 = 40$  (the thresholds used for VMR-WB mode 1 classification).

29 The unvoiced classification in VMR-WB modes 1 and 2 is processed the same way as if the  
 30 maximum rate would not be limited.

31  
 32 If maximum half-rate is enforced for the current frame, the Voiced HR encoding type selection is  
 33 modified such that the normalized correlation threshold  $\delta_g = 0.5$  for all VMR-WB modes 0, 1 and 2  
 34 (Section 5.11.4).

35  
 36 If the maximum bit rate is limited to QR by the system and the signal is classified as unvoiced, then  
 37 Unvoiced QR can be used. This is however possible only in the non-interoperable modes of VMR-WB  
 38 (i.e., mode 0, 1, and 2), as the AMR-WB codec is unable to decode the QR frames. If Unvoiced QR  
 39 cannot be used, the frame is encoded as if no maximum rate limitation was imposed and it is marked  
 40 as erased frame for the bit packing. Similarly, if the imposed maximum rate is limited to ER, all active  
 41 speech frames are encoded normally, but marked as erased. Then the frame is encoded as an  
 42 Erasure ER frame or Erasure QR frame (using the lowest rate allowed) by means of an in-band  
 43 signaling. This is done using the fact that the bits corresponding to ISF indices are not permitted to be  
 44 all 0. This pattern is thus used for signaling an erasure. Further, to prevent generation of an all-zero  
 45 packet by the VMR-WB encoder, the least significant bit of the last ISF byte is set to 1. This translates  
 46 into setting the second data bit of ER frame to 0x01 and the 5<sup>th</sup> data bit of QR frame to 0x01. All other  
 47 bits are set to 0.

#### 48 49 **5.12.2 Maximum and Minimum Rate Operation in Rate-Set I**

50  
 51 Maximum and minimum rate operation in Rate-Set I is similar to that of Rate-Set II described above  
 52 with minor differences since there is no QR encoding type in Rate-Set I.

53  
 54 For minimum rate operation, if min-rate is FR then lower rates cannot be used and all frames are  
 55 encoded with an appropriate FR encoding type (RS-I Generic FR or RS-I Voiced FR).

1 If min-rate is HR then ER cannot be used. In this case, inactive frames are encoded with Unvoiced  
2 HR. The rest of classification is as usual.

3  
4 If min-rate is ER, then the rate selection works in normal operation. All inactive frames are encoded  
5 with RS-I CNG-ER.

6  
7 In the case of maximum HR limitation, all active speech frames that would be classified as FR during  
8 normal operation must be encoded using an appropriate HR encoding type. The classification and  
9 rate selection mechanism are performed as usual. That is determining inactive speech frames  
10 (encoded with RS-I CNG-ER) and unvoiced frames (encoded with RS-I Unvoiced HR), Remaining  
11 frames that would be classified as FR during normal operation are encoded using the RS-I Generic  
12 HR encoding type or RS-I Voiced HR encoding type. Note that these two HR encoding types are not  
13 used in normal operation.

14  
15 In case of maximum HR operation, the frame is declared an Unvoiced frame if local VAD=0 or the  
16 following condition is satisfied

$$17 \quad (\bar{R}_{xy3} < th_4) \text{ AND } (e_t < th_5) \text{ AND } (dE < th_6)$$

18  
19  
20 where  $th_4 = 0.695$ ,  $th_5 = 4$ ,  $th_6 = 40$ .

21  
22 These frames are encoding using RS-I Unvoiced HR. Then signal is tested whether it is stable voiced  
23 and can be encoded using RS-I Voiced HR. This is performed using a relaxed test as follows: if  $g_s \geq$   
24  $0.5$ , RS-I Voiced HR is used otherwise no signal modification is performed and the frame is encoded  
25 using RS-I Generic HR.

### 26 5.13 Quantization of the ISP Coefficients

27  
28 **Routine Name:** `isf_enc`

29 **Inputs:**

- 30 •  $q_i$ : The imittance spectral pairs for current frame

31 **Outputs:**

- 32 •  $\hat{q}_i$ : The quantized imittance spectral pairs for current frame
- 33 •  $f_i$ : The imittance spectral frequencies for current frame

34 **Initialization:**

- 35 • The memory of the quantizer predictors are set to zero at initialization.

36  
37 The LP filter coefficients are quantized using the ISP representation in the frequency domain; that is

$$38 \quad f_i = \frac{f_s}{2\pi} \arccos(q_i), \quad i = 0, \dots, 14$$

$$39 \quad = \frac{f_s}{4\pi} \arccos(q_i), \quad i = 15$$

(5.13-1)

40 where  $f_i$  are the ISFs in Hz [0,6400] and  $f_s = 12\,800$  is the sampling frequency. The ISF vector is  
41 given by  $\mathbf{f}^t = [f_0, f_1, \dots, f_{15}]$ , with  $t$  denoting transpose. Either 1<sup>st</sup> order moving-average (MA) prediction or  
42 1<sup>st</sup> order auto-regressive (AR) prediction is applied to the mean-removed ISF vector, and the  
43 prediction error vector is quantized using a combination of split vector quantization (SVQ) and  
44 multistage vector quantization (MSVQ). The prediction and quantization are performed as follows. Let  
45  $\mathbf{z}(n)$  denote the mean-removed ISF vector at frame  $n$ . The prediction residual vector  $\mathbf{r}(n)$  is given by:

$$\mathbf{r}(n) = \mathbf{z}(n) - \mathbf{p}(n) \quad (5.13-2)$$

where  $\mathbf{p}(n)$  is the predicted ISF vector at frame  $n$ . The LP filter quantization will make use of the voice classification for the current frame. In all encoding types except Voiced HR, first order MA prediction is used where:

$$\mathbf{p}(n) = \alpha_{MA} \hat{\mathbf{r}}(n-1) \quad (5.13-3)$$

where  $\hat{\mathbf{r}}(n-1)$  is the quantized residual vector at the past frame and  $\alpha_{MA} = 1/3$ .

In order to improve the ISF quantization performance in case of Voiced HR frames, AR prediction is used. Voiced HR encoding type is used to encode stable voiced signals, whereby successive ISF vectors are strongly correlated. Thus the use of AR prediction results in a prediction error with lower dynamic range. Since the predictor is switched back to MA prediction for other types of frames and the spectrum is usually stable during Voiced HR frames, the effect of error propagation in case of frame erasures is well controlled. The predicted ISF vector in case of Voiced HR frames is given by

$$\mathbf{p}(n) = \alpha_{AR} \hat{\mathbf{z}}(n-1) \quad (5.13-4)$$

where  $\hat{\mathbf{z}}(n-1)$  is the mean-removed quantized ISF vector from previous frame and  $\alpha_{AR} = 0.65$ . When AR prediction is used, the prediction error in Equation (5.13-2) has a lower dynamic range. Thus, in order to use the same first stage quantization tables for the different rates and encoding types, the error vector  $\mathbf{r}(n)$  is scaled to bring the dynamic range close to that of MA prediction. The scaling is given by

$$\mathbf{r}_s(n) = S_q \mathbf{r}(n) \quad (5.13-5)$$

where  $S_q = 1.25$  in case of AR prediction and  $S_q = 1$  in case of MA prediction (no scaling).

The ISF (scaled) prediction error vector  $\mathbf{r}_s$  is quantized using split-multistage vector quantization S-MSVQ. The vector is split into 2 subvectors  $\mathbf{r}_1(n)$  and  $\mathbf{r}_2(n)$  of dimensions 9 and 7, respectively. The 2 subvectors are quantized in two stages. In the first stage  $\mathbf{r}_1(n)$  is quantized with 8 bits and  $\mathbf{r}_2(n)$  with 8 bits. A schematic block diagram of ISF quantization using switched MA/AR prediction is shown in

Figure 5.13-1. For FR and Unvoiced HR, the quantization error vectors  $\mathbf{r}_i^{(2)} = \mathbf{r} - \hat{\mathbf{r}}_i, i=1,2$  are split in the next stage into 3 and 2 subvectors, respectively. The subvectors are quantized using the bit-rates described in Table 5.13-1:

**Table 5.13-1: Quantization of mean-removed ISF vector for the FR and Unvoiced HR.**

1. Unquantized 16-element-long ISF vector				
2. Stage 1 ( $\mathbf{r}_1$ ) 8 bits			2. Stage 1 ( $\mathbf{r}_2$ ) 8 bits	
3. Stage 2 $\mathbf{r}_1^{(2)}, 0-2$ 6 bits	3. Stage 2 $\mathbf{r}_1^{(2)}, 3-5$ 7 bits	3. Stage 2 $\mathbf{r}_1^{(2)}, 6-8$ 7 bits	3. Stage 2 $\mathbf{r}_2^{(2)}, 0-2$ 5 bits	3. Stage 2 $\mathbf{r}_2^{(2)}, 3-6$ 5 bits

For Generic HR and Voiced HR, the quantization error vectors  $\mathbf{r}_i^{(2)} = \mathbf{r} - \hat{\mathbf{r}}_i, i=1,2$  are split in the next stage into 2 and 1 subvectors, respectively. The subvectors are quantized using the bit-rates described in Table 5.13-2.

**Table 5.13-2: Quantization of ISF vector for Generic HR and Voiced HR**

1. Unquantized 16-element-long ISF vector		
2. Stage 1 ( $r_1$ ) 8 bits		2. Stage 1 ( $r_2$ ) 8 bits
3. Stage 2 $r_2^{(2)}$ , 0-4 7 bits	3. Stage 2 $r_1^{(2)}$ , 5-8 7 bits	3. Stage 2 $r_2^{(2)}$ , 0-6 6 bits

- 1 For Unvoiced QR, the quantization error vectors  $r_i^{(2)} = r_i - \hat{r}_i$ ,  $i=1,2$ , are split in the next stage into 2  
2 and 1 subvectors, respectively. The subvectors are quantized using the bit-rates described in Table  
3 5.13-3.

4 **Table 5.13-3: Quantization of ISF vector for Generic HR and Voiced HR**

1. Unquantized 16-element-long ISF vector		
2. Stage 1 ( $r_1$ ) 8 bits		2. Stage 1 ( $r_2$ ) 8 bits
3. Stage 2 $r_1^{(2)}$ , 0-4 5 bits	3. Stage 2 $r_1^{(2)}$ , 5-8 5 bits	3. Stage 2 $r_2^{(2)}$ , 0-6 6 bits

- 5  
6 The two 8-bit codebooks are shared by all ISF 46-bit, 36-bit, and 32-bit quantizers. However, in the  
7 case of the 36-bit quantizer used for Voiced HR frames, the 8-bit codebook of the 1<sup>st</sup> split has been  
8 slightly modified to suit better Voiced HR frame ISF quantization. Thus, 28 entries statistically less  
9 used by this coding type have been replaced by entries optimized for Voiced HR frames. A squared  
10 error distortion measure is used in the quantization process. In general, for an input ISF or error  
11 residual subvector  $r_i$ ,  $i = 1,2$  and a quantized vector at index  $k$ ,  $\hat{r}_i^k$ , the quantization is performed by  
12 finding the index  $k$  which minimizes

$$13 \quad E = \sum_{i=m}^n [r_i - \hat{r}_i^k]^2 \quad (5.13-6)$$

- 14 where  $m$  and  $n$  are the indices of the first and last elements of the subvector.

15

- 16 Once the quantized (scaled) prediction error vector is found, inverse scaling is applied in case of AR  
17 prediction, that is

18

$$19 \quad \hat{r}(n) = \hat{r}_s(n) / S_q, \quad (5.13-7)$$

20

- 21 and mean-removed quantized ISF vector is given by

22

$$23 \quad \hat{z}(n) = p(n) + \hat{r}(n). \quad (5.13-8)$$

24

- 25 The quantized ISF vector is then found by adding the ISF-mean to the quantized mean-removed ISF  
26 vector. The memories of both MA and AR predictors are updated for use in next frame.

27

- 28 After quantization in the frequency domain, the ISF parameters are converted into the cosine domain  
29 to obtain the ISP vector  $\hat{q}$ . Similar to the case of unquantized LP parameters, the quantized ISPs in  
30 the present and previous frames are interpolated to obtain a different quantized LP filter in every  
31 subframe (as in Section 5.6.5).

32

33

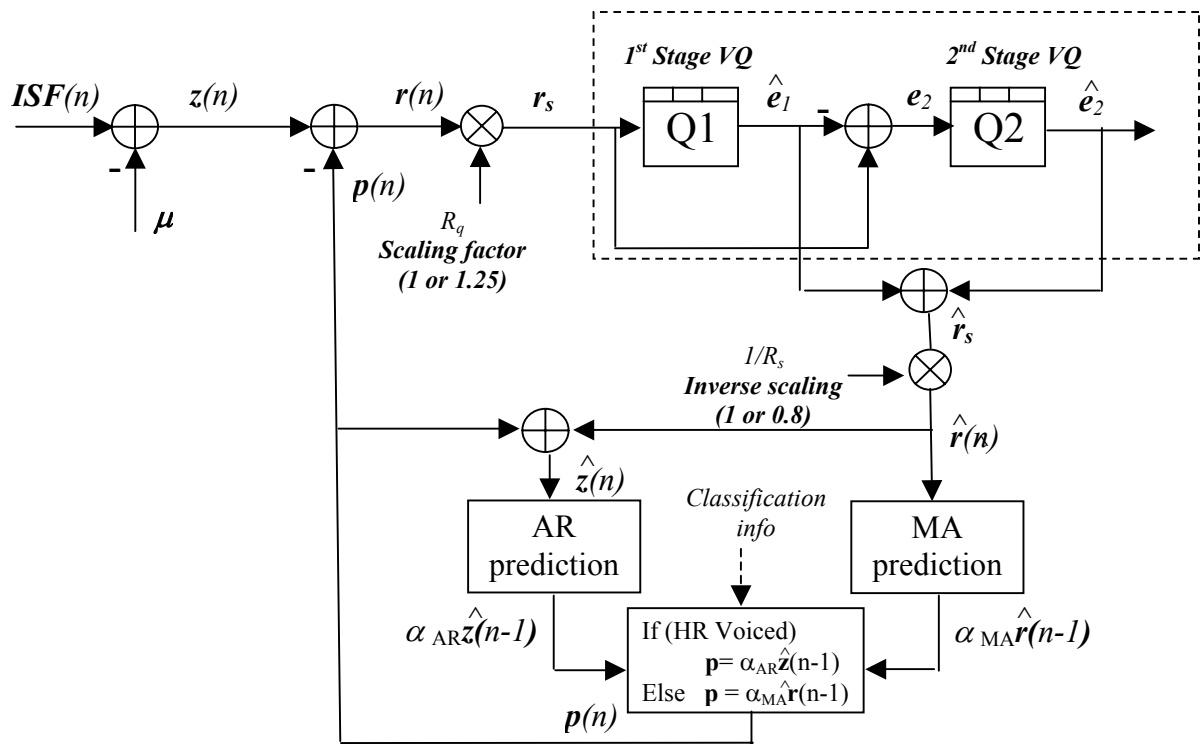


Figure 5.13-1: Block diagram of ISF quantization using switched MA/AR prediction.

### 5.13.1 Quantization of ISP Parameters in Rate-Set I Operation

In RS-I Generic FR, the same 36-bit quantizer as in RS-II Generic HR is used (with MA prediction). In RS-I Voiced FR and Signaling Voiced HR, the same 36-bit quantizer as in RS-II Voiced HR is used (with AR prediction and using the 28 entries specific to voiced signals). In RS-I Unvoiced HR, the same 46-bit quantizer as in RS-II Generic FR is used (with MA prediction).

In RS-I Generic HR, a 30 bit quantizer is used. This quantizer is derived from the 36-bit quantizer used in RS-I Generic FR by dropping the second stage quantization of the last split which saves 6 bits. This is shown in Table 5.13-4. Similarly, in RS-I Voiced HR, a 30 bit quantizer is used which is derived from the 36-bit quantizer used in RS-I Voiced FR (by dropping the last split in the second stage quantization).

Table 5.13-4: Quantization of ISF vector for RS-I Generic HR and RS-I Voiced HR

1. Unquantized 16-element-long ISF vector		
2. Stage 1 ( $r_1$ ) 8 bits		2. Stage 1 ( $r_2$ ) 8 bits
3. Stage 2 $r_1^{(2)}$ , 0-4 7 bits	3. Stage 2 $r_1^{(2)}$ , 5-8 7 bits	3. Stage 2 No further quantization

In RS-I Signaling Generic HR, a 23 bit quantizer is used which is derived from the 36-bit quantizer used in RS-I Generic FR. This is done by dropping the second stage quantization of the last two splits. This is shown in Table 5.13-5 below.

Table 5.13-5: Quantization of ISF vector for RS-I Signaling Generic HR.

1. Unquantized 16-element-long ISF vector		
2. Stage 1 ( $r_1$ ) 8 bits		2. Stage 1 ( $r_2$ ) 8 bits
3. Stage 2 $r_1^{(2)}$ , 0-4 7 bits	3. Stage 2 No further quantization	3. Stage 2 No further quantization

1

2 **5.14 Impulse Response Computation**

3

4 **Routine Name:** `vmr_encoder`5 **Inputs:**

- 6 •  $a_i$ : The unquantized interpolated LP filter coefficients
- 7 •  $\hat{a}_i$ : The quantized interpolated LP filter coefficients

8 **Outputs:**

- 9 •  $h(n)$ : The impulse response of the synthesis filter

10 **Initialization:**

- 11 • None

12

13 The impulse response,  $h(n)$ , of the weighted synthesis filter

14

$$15 \quad H(z)W(z) = A(z/\gamma_1)H_{\text{de-emph}}(z)/\hat{A}(z)$$

16 (5.14-1)

17

18 is computed for each subframe. The interpolation of the LP coefficients is described in Section 5.6.5.  
 19 This impulse response is needed for the search of adaptive and fixed-codebooks. The impulse  
 20 response  $h(n)$  is computed by filtering the vector of coefficients of the filter  $A(z/\gamma_1)$  extended by  
 21 zeros through the two filters  $1/\hat{A}(z)$  and  $H_{\text{de-emph}}(z)$ .

22 **5.15 Target Signal Computation**

23

24 **Routine Name:** `find_targets`25 **Inputs:**

- 26 •  $s(n)$ : speech signal
- 27 •  $a_i$ : The unquantized LP filter coefficients
- 28 •  $\hat{a}_i$ : The quantized LP filter coefficients

29 **Outputs:**

- 30 •  $x(n)$ : The target signal for the adaptive codebook search
- 31 •  $r(n)$ : The residual signal

32 **Initialization:**

- 33 • The filter memories are set to zero at initialization.

34

1 The target signal for adaptive codebook search is usually computed by subtracting the zero-input  
 2 response of the weighted synthesis filter  $H(z)W(z) = A(z/\gamma_1)H_{\text{de-emph}}(z)/\hat{A}(z)$  from the  
 3 weighted speech signal  $s_w(n)$ . This is performed on a subframe basis.

4  
 5 An equivalent procedure for computing the target signal, which is used in this codec, is the filtering of  
 6 the LP residual signal  $r(n)$  through the combination of synthesis filter  $1/\hat{A}(z)$  and the weighting  
 7 filter  $A(z/\gamma_1)H_{\text{de-emph}}(z)$ . After determining the excitation for the subframe, the initial states of  
 8 these filters are updated by filtering the difference between the LP residual and excitation. The  
 9 memory update of these filters is explained in Section 5.21. The residual signal  $r(n)$  which is needed  
 10 for finding the target vector is also used in the adaptive codebook search to extend the past excitation  
 11 buffer. This simplifies the adaptive codebook search procedure for delays less than the subframe size  
 12 of 64 as will be explained in the next section. The LP residual is given by

$$13 \quad r(n) = s(n) + \sum_{i=1}^{16} \hat{a}_i s(n-i), \quad n = 0, \dots, 63 \quad (5.15-1)$$

## 14 5.16 Adaptive Codebook Search

15  
 16 **Routine Name:** pit\_encode, lp\_filt\_excitation\_enc, gp\_clip\_test\_isf, gp\_clip,  
 17 gp\_clip\_test\_gain\_pit, hnw, opt\_hnw\_state\_update, select\_subframe

### 18 Inputs:

- 19 •  $r(n)$ : The residual signal
- 20 •  $u(n)$ : Past excitation signal
- 21 •  $x(n)$ : The target signal for the adaptive codebook search
- 22 •  $h(n)$ : The impulse response of the weighted synthesis filter
- 23 •  $\bar{d}_k$ : The signal modification pitch delay parameter at the frame end boundary
- 24 •  $T_{op}$ : The open-loop estimates
- 25 •  $f_i$ : The immittance spectral frequencies for current frame

### 26 Outputs:

- 27 •  $v(n)$ : The adaptive codevector
- 28 •  $y(n)$ : The filtered adaptive codevector
- 29 • Integer and fractional close-loop pitch lag
- 30 •  $g_p$ : The adaptive codebook gain

### 31 Initialization:

- 32 • The buffers are set to zero at initialization. The 3-dimensional memory of the gain clipping  
 33 procedure is initialized to 120, 0.6 and 0, respectively.

34  
 35 The adaptive codebook search is only performed in FR and Generic HR coding types. For Voiced HR  
 36 encoding type, no closed-loop pitch search is performed since the pitch delay contour for the frame is  
 37 determined as explained in Section 5.8.6. In this case, the adaptive codebook excitation is computed  
 38 based in the delay contour. In case of Unvoiced and CNG-QR/ER encoding types, the adaptive  
 39 codebook is not used.

The adaptive codebook search consists of performing closed-loop pitch search, and then computing the adaptive codevector,  $v(n)$ , by interpolating the past excitation at the selected fractional pitch lag. The adaptive codebook parameters (or pitch parameters) are the pitch delay and pitch gain  $g_p$  (adaptive codebook gain). In the search stage, the excitation is extended by the LP residual to simplify the closed-loop search.

### 5.16.1 Adaptive Codebook Search in Full Rate Encoding Type

For FR encoding types, adaptive codebook search is performed on a subframe basis. In the first and third subframes, a fractional pitch delay is used with resolutions  $\frac{1}{4}$  in the range  $[34, 127\frac{3}{4}]$ , resolutions  $\frac{1}{2}$  in the range  $[128, 159\frac{1}{2}]$ , and integers only in the range  $[160, 231]$ . For the second and fourth subframes, a pitch resolution of  $\frac{1}{4}$  is always used in the range  $[T_p-8, T_p+7\frac{3}{4}]$ , where  $T_p$  is nearest integer to the fractional pitch lag of the previous (1<sup>st</sup> or 3<sup>rd</sup>) subframe.

Closed-loop pitch analysis is performed around the open-loop pitch estimates on a subframe basis. In the first (and third) subframe the range  $T_{op} \pm 7$ , bounded by 34...231, is searched, where  $T_{op} = d_0$  or  $d_1$ . For the other subframes, closed-loop pitch analysis is performed around the integer pitch selected in the previous subframe, as described above. In FR encoding types, the pitch delay is encoded with 9 bits in the first and third subframes and the relative delay of the other subframes is encoded with 6 bits. The closed-loop pitch search is performed by minimizing the mean-squared weighted error between the original and synthesized speech. This is achieved by maximizing

$$T_k = \frac{\sum_{n=0}^{63} x(n)y_k(n)}{\sqrt{\sum_{n=0}^{63} y_k(n)y_k(n)}} \quad (5.16.1-1)$$

where  $x(n)$  is the target signal and  $y_k(n)$  is the past filtered excitation at delay  $k$  (past excitation convolved with  $h(n)$ ). Note that the search range is limited around the open-loop pitch as explained earlier. The convolution  $y_k(n)$  is computed for the first delay in the searched range, and for the other delays, it is updated using the recursive relation

$$y_k(n) = y_{k-1}(n-1) + u(-k)h(n) \quad n=0, \dots, 63 \quad (5.16.1-2)$$

where  $u(n)$ ,  $n = -(231+17), \dots, 63$ , is the excitation buffer. Note that in search stage, the samples  $u(n)$ ,  $n = 0, \dots, 63$ , are not known, and they are needed for pitch delays less than 64. To simplify the search, the LP residual is copied to  $u(n)$  in order to make the relation in Equation (5.16.1-2) valid for

all delays. Once the optimum integer pitch delay is determined, the fractions from  $-\frac{3}{4}$  to  $\frac{3}{4}$  with a

step of  $\frac{1}{4}$  around that integer are tested. The fractional pitch search is performed by interpolating the

normalized correlation in Equation (5.16.1-1) and searching for its maximum. The interpolation is performed using an FIR filter for interpolating the term in Equation (5.16.1-1) using a Hamming windowed sinc function truncated at  $\pm 17$ . The filter has its cut-off frequency ( $-3$  dB) at 5050 Hz and  $-6$  dB at 5760 Hz in the down-sampled domain, which means that the interpolation filter exhibit low-pass frequency response.

### 5.16.2 Adaptive Codebook Search in Generic HR

In the Generic HR encoding type, adaptive codebook search is performed twice per frame. That is, the adaptive codebook parameters are computed every half-frame. In the first half-frame, a fractional pitch delay is used with resolutions  $\frac{1}{2}$  in the range  $[34, 91\frac{1}{2}]$ , and integers only in the range  $[92, 231]$ .

For the second half-frame a pitch resolution of  $\frac{1}{2}$  is always used in the range  $[T_1-8, T_1+7\frac{1}{2}]$ , where

$T_1$  is nearest integer to the fractional pitch lag of the first half-frame. In Generic HR, the adaptive codebook search is similar to FR with the difference that the excitation codevector size is 128 instead of 64.

### 5.16.3 Computation of Adaptive Codebook Excitation in FR and Generic HR

Once the fractional pitch lag is determined, the initial adaptive codebook excitation  $v'(n)$  is computed by interpolating the past excitation signal  $u(n)$  at the given phase (fraction). The interpolation is performed using an FIR filter (Hamming windowed sinc function) for interpolating the past excitation with the sinc truncated at  $\pm 63$ . The filter has its cut-off frequency ( $-3$  dB) at 5840 Hz and  $-6$  dB at 6020 Hz in the down-sampled domain, which means that the interpolation filter exhibit low-pass frequency response. Thus, even when the pitch delay is an integer value, the adaptive codebook excitation consists of a low-pass filtered version of the past excitation at the given delay and not a direct copy thereof. Further, for delays smaller than the subframe size, the adaptive codebook excitation is completed based on the low-pass filtered interpolated past excitation and not by repeating the past excitation. In FR encoding types, the adaptive codebook excitation is computed for the subframe size of 64 samples. In the Generic HR, it is computed for 128 samples.

### 5.16.4 Computation of Adaptive Codebook Excitation in Voiced HR

In case of the Voiced HR encoding type, signal modification is used and a delay contour is computed for the whole frame as described in Equation (5.11.2-1). The initial adaptive codebook excitation  $v'(n)$  in a certain subframe is computed by interpolating the past excitation in the adaptive codebook buffer at the delays given by the delay contour. The delay contour is computed using a  $1/8$  sub-sample resolution. The interpolation is performed using an FIR filter for interpolating the past excitation with the sinc truncated at  $\pm 32$ . The filter has its cut-off frequency ( $-3$  dB) at 5330 Hz and  $-6$  dB at 6020 Hz in the down-sampled domain.

### 5.16.5 Frequency Dependent Pitch Prediction

In order to enhance the pitch prediction performance in wideband signals, a frequency-dependant pitch predictor is used. This is important in wideband signals since the periodicity does not necessarily extend over the whole spectrum. In this algorithm, there are two signal paths associated with respective sets of pitch codebook parameters, wherein each signal path comprises a pitch prediction error calculating device for calculating a pitch prediction error of a pitch codevector from a pitch codebook search mechanism. One of these two paths comprises a low-pass filter for filtering the pitch codevector before calculation of pitch prediction error. The pitch prediction error is then calculated for these two signal paths. The pitch prediction errors calculated for the two signal paths are compared, and the signal path having the lowest calculated pitch prediction error is selected, along with the associated pitch gain and pitch lag. The low-pass filter used in the second path is in the form  $B_{LP}(z) = 0.18 + 0.64z^{-1} + 0.18z^{-2}$ . Note that 1 bit is used to encode the chosen path.

Thus, there are two possibilities to generate the adaptive codebook  $v(n)$ ,  $v(n) = v'(n)$  in the first

path, or  $v(n) = \sum_{i=-1}^1 b_{LP}(i+1)v'(n+i)$  in the second path, where  $\mathbf{b}_{LP} = [0.18, 0.64, 0.18]$ . The path,

which results in minimum energy of the target signal  $x_2(n) = x(n) - g_p y(n)$ ,  $n = 0, \dots, 63$ , is

selected for the filtered adaptive codebook vector, where  $g_p$  is the pitch gain computed as in the next section.

In case of FR encoding types, 1 bit per subframe is used indicate the use of low-pass filtering (4 bits per frame). In case of Voiced HR encoding type, the path filtering decision in the first subframe is extended to the second subframe. Similarly, the same low-pass filtering decision is used in both third and fourth subframes. Thus only 2 bits per frame are used for low-pass filtering in case of Voiced HR. In case of Generic FR, no low-pass filtering of the adaptive codebook excitation is used.

### 5.16.6 Computation of Adaptive Codebook Gain

The adaptive codebook gain, or the pitch gain, is then found by

$$g_p = \frac{\sum_{n=0}^{N-1} x(n)y(n)}{\sum_{n=0}^{N-1} y(n)y(n)}, \quad \text{constrained by } 0 \leq g_p \leq 1.2 \quad (5.16.6-1)$$

where  $y(n) = v(n) * h(n)$  is the filtered adaptive codebook vector (zero-state response of  $H(z)W(z)$  to  $v(n)$ ), and  $N=64$ . Note that the adaptive codebook vector may have been low-pass filtered as described in the previous section.

To avoid instability in case of channel errors, the adaptive codebook gain, or pitch gain,  $g_p$  is bounded by 0.95 if the pitch gains of the previous subframes have been close to 1 and the LP filters of the previous subframes have been close to being unstable (highly resonant).

The instability elimination method tests two conditions, resonance condition using the LP spectral parameters (minimum distance between adjacent ISFs), and gain condition by testing for high-valued pitch gains in the pervious frames. The method works as follows. First, the minimum distance between adjacent ISFs is computed as

$$d_{\min} = \min(ISF(i) - ISF(i-1)), \quad i = 1, \dots, 14, \quad (5.16.6-2)$$

where the ISF frequencies are in the range [0,6400], then mean minimum distance is computed as

$$\bar{d}_{\min} = 0.8\bar{d}_{\min} + 0.2d_{\min} \quad \text{constrained by } \bar{d}_{\min} \leq 120. \quad (5.16.6-3)$$

Second, the mean pitch gain is computed as

$$\bar{g} = 0.9\bar{g} + 0.1g_p \quad \text{constrained by } \bar{g} \geq 0.6. \quad (5.16.6-4)$$

If  $\bar{g} \geq 0.9$  and  $\bar{d}_{\min} \leq 60$  then pitch gain clipping is performed by limiting pitch gain to  $g_p = 0.95$ .

These conditions correspond to an average pitch gain of more that 0.9 and an average minimum distance of less than 60 Hz in the last 8 to 9 subframes approximately. For such signals, the instability at the output may happen in case of channel errors due to mismatch between the decoder and the encoder. Limiting the pitch gain to 0.9 in such conditions avoids this problem.

The initial values of  $\bar{g}$  and  $\bar{d}_{\min}$  are 0.6 and 120, respectively. Every time ER, QR, or Unvoiced HR are used,  $\bar{g}$  and  $\bar{d}_{\min}$  are reset to their initial values. In addition to the above mentioned methods, the adaptive codebook gain  $g_p$  is bounded by 0.95 also if the subframe energy of the target signal  $x(n)$  drops by more than 6 dB relatively to the previous subframe and the mean pitch gain  $\bar{g} > 1$  at

1 the same time. This measure prevents a potential divergence between floating-point and fixed-point  
2 implementations of the VMR-WB decoder.

### 5.16.7 Adaptive Codebook in Rate-Set I Operation

7 In RS-I Generic FR, in the first and third subframes, a fractional pitch delay is used with resolutions  
8  $1/2$  in the range  $[34, 91 \frac{1}{2}]$ , and integers only in the range  $[92, 231]$ . For the second and fourth  
9 subframes, a pitch resolution of  $1/2$  is always used in the range  $[T_1-8, T_1+7 \frac{1}{2}]$ , where  $T_1$  is nearest  
10 integer to the fractional pitch lag of the previous (1st or 3rd) subframe. The adaptive codebook search  
11 is performed similar to RS-II Generic FR described in Section 5.16-1 above. The computation of the  
12 adaptive codebook excitation in RS-I Generic FR is similar to RS-II Generic FR described in Section  
13 5.16-3.

15 In RS-I Voiced FR, signal modification is used and the adaptive codebook excitation is computed  
16 similar to RS-II Voiced HR described in Section 5.16-4.

18 It should be noted that in Rate-Set I operation, since harmonic noise weighting is used in the  
19 algebraic codebook search, the adaptive codebook gain is computed after harmonic noise weighting  
20 described in Section 5.16.8 has been performed. Further, the frequency dependent pitch prediction  
21 approach described in Section 5.16.5 is modified so that a variable low pass filter is used (this will be  
22 described in Section 5.16.9).

#### 5.16.7.1 Adaptive Codebook in RS-I Generic HR and RS-I Voiced HR

25 In RS-I Voiced FR, signal modification is used and the pitch lag is encoded with 9 bits once per  
26 frame. The adaptive codebook excitation is computed similar to RS-II Voiced HR described in Section  
27 5.16-4.

29 In RS-I Generic HR, the pitch delay is determined and encoded only twice per frame. In the first half-  
30 frame, a integer pitch delay is used (resolution 1) in the range  $[34, 91]$ , and integers with resolution 2  
31 in the range  $[92, 230]$  (only even pitch lags). For the second half-frame, a pitch resolution of 1 (only  
32 integers) is always used in the range  $[T_1-8, T_1+7]$ , where  $T_1$  is integer pitch lag of the previous half-  
33 frame. The adaptive codebook search is performed similar to RS-II Generic FR described in Section  
34 5.16-1 above with the exception that the search is performed over two subframes (10 ms half-frame).  
35 That is, the mean-squared weighted error is minimized by maximizing the term

$$T_k = \frac{\sum_{n=0}^{127} x(n)y_k(n)}{\sqrt{\sum_{n=0}^{127} y_k(n)y_k(n)}}$$

37 with  $x(n)$  and  $y_k(n)$  computed over 128-sample half-frames.

40 The computation of the adaptive codebook excitation in RS-I Generic FR is similar to RS-II Generic  
41 FR described in Section 5.16-3. Although the lag is computed once for the first two subframes, the  
42 excitation is computed every subframe by forcing the pitch gain to a certain value  $g_f$  (described in  
43 Section 5.17.2.4) in either first or second subframe. For the subframe where the pitch gain is forced to  
44  $g_f$ , no fixed codebook is used (the excitation is only based on the adaptive codebook contribution).  
45 The subframe in which the pitch gain is forced to  $g_f$  is determined in closed loop by trying both  
46 combinations and selecting the one that minimizes the weighted error over the half-frame. In the first  
47 iteration, the pitch gain and adaptive codebook excitation and the fixed codebook excitation and gain  
48 are computed in the first subframe, and in the second subframe the pitch gain is forced to  $g_f$  and the  
49 adaptive codebook excitation is computed with no fixed codebook contribution. In the second  
50 iteration, in the first subframe the pitch gain is forced to  $g_f$  and the adaptive codebook excitation is

1 computed with no fixed codebook contribution, and in the second subframe the pitch gain and  
 2 adaptive codebook excitation and the fixed codebook excitation and gain are computed. The  
 3 weighted error is computed for both iterations and the one that minimizes the error is retained. 1 bits  
 4 is used per half-frame to determine the index of the subframe where fixed codebook contribution is  
 5 used. The same procedure is repeated for the second half-frame (third and fourth subframes).

6  
 7 This procedure of using the fixed codebook contribution only in one subframe in a half-frame is also  
 8 utilized in RS-I Voiced HR.

9  
 10 It should be noted that in Rate-Set I operation, since harmonic noise weighting is used in the  
 11 algebraic codebook search, the adaptive codebook gain is computed after harmonic noise weighting  
 12 described in Section 5.16.8 has been performed. Further, the frequency dependent pitch prediction  
 13 approach described in Section 5.16.5 is modified so that a variable low pass filter is used (this will be  
 14 described in Section 5.16.9).

### 15 16 **5.16.8 Harmonic Noise Weighting in Rate-Set I Operation**

17  
 18 The perceptual weighting used in the closed loop adaptive codebook search is modified in Rate-Set I  
 19 operation to include harmonic noise weighting (HNW), which overcomes some of the limitations of the  
 20 low rate coding modes [27]. HNW efficiently utilizes noise-masking properties by exploiting the  
 21 speech signal's harmonic structure. The HNW filter is given by

$$22 \quad C(z) = 1 - \varepsilon_p g'_p z^{-D} \quad (5.16.8-1)$$

23  
 24 where  $D$  is the closed loop fractional lag (pitch delay),  $g'_p$  is the HNW gain, and  $\varepsilon_p$  is the HNW  
 25 coefficient. The perceptual weighting filter is now given by

$$26 \quad W(z) = A(z/\gamma)H_{de-emph}(z)C(z) \quad (5.16.8-2)$$

27  
 28  
 29 The gain  $g'_p$  in Equation (5.16.8-1) is found by

$$30 \quad g'_p = \frac{\sum_{n=0}^{63} x(n)y(n)}{\sum_{n=0}^{63} y(n)y(n)}, \quad \text{bounded by } 0 \leq g'_p \leq 1.0, \quad (5.16.8-3)$$

31  
 32  
 33 where  $y(n) = v'(n) * h(n)$ . The HNW coefficient  $\varepsilon_p$  is found from a decreasing function of delay given  
 34 by

$$35 \quad \varepsilon_p = \begin{cases} \varepsilon_{\min}, & D \geq D_{\max} \\ \varepsilon_{\min} + \Delta \frac{(D_{\max} - D)}{D_{\max}}, & D \geq \left(1 - \frac{\varepsilon_{\max} - \varepsilon_{\min}}{\Delta}\right) D_{\max} \\ \varepsilon_{\max}, & \text{Otherwise} \end{cases} \quad (5.16.8-4)$$

36  
 37  
 38 where  $\varepsilon_{\max}$  is the maximum allowable value of the HNW coefficient,  $\varepsilon_{\min}$  is the minimum allowable  
 39 value of the HNW coefficient,  $D_{\max}$  is the maximum pitch period above which HNW coefficient is set to  
 40  $\varepsilon_{\min}$ , and  $\Delta$  is the slope for the HNW coefficient. In addition,  $\varepsilon_p$  is set to zero for large variations ( $> \pm 10$ )  
 41 in closed loop pitch lag between the current and previous subframes.  
 42

1 To generate the fractional pitch delay  $z^{-D}$  part of the HNW filter, an interpolation is performed using a  
 2 FIR low pass filter with a windowed sinc function truncated at  $\pm 31$  and a cut-off frequency at  
 3 approximately 1200 Hz.

4  
 5 The harmonic noise weighted target signal is generated by passing  $x(n)$  through the HNW filter  $C(z)$ .  
 6 The state of the harmonic noise filter used for generating the harmonic noise weighted target signal  
 7 comprises the past spectrally weighted error and is computed in the memory update Section 5.21.  
 8 The output of the HNW filter is represented by  $x_{hnw}(n)$ .

9  
 10 The harmonic noise weighted impulse response is generated by passing  $h(n)$  through the filter  $C(z)$ .  
 11 The state of the filter  $C(z)$  is set to 0 for generating the impulse response. The impulse response thus  
 12 generated is represented by  $h_{hnw}(n)$ .

### 13 5.16.9 Multi-tap Optimized Long-Term Predictor (LTP)

14  
 15  
 16  
 17 In Rate-Set I operation, instead of a fixed low-pass filter (LPF), a variable filter is used for filtering the  
 18 adaptive codebook excitation  $v(n)$ . The filter given by

$$19 \quad B_{lp}(z) = \frac{1-\alpha}{2}z + \alpha + \frac{1-\alpha}{2}z^{-1}, \quad 0.55 \leq \alpha \leq 1.0$$

20  
 21 is used to generate the adaptive codebook excitation  $v'(n)$ , where the filter parameter  $\alpha$  is determined  
 22 as follows: The excitation  $v(n)$  is decomposed into two constituents:

$$23 \quad \begin{aligned} v_1(n) &= v(n) \\ v_2(n) &= v(n+1) + v(n-1) \end{aligned}$$

24 so that

$$25 \quad v'(n) = \alpha v_1(n) + \frac{1-\alpha}{2} v_2(n).$$

26 If we define  $y_1(n) = v_1(n) * h(n)$ ,  $y_2(n) = v_2(n) * h(n)$ , and  $y(n) = v'(n) * h(n)$ , then  $\alpha$  is obtained such  
 27 that it minimizes the distortion (mean squared error) between  $x(n)$  and gain scaled  $y(n)$ , and is  
 28 computed using the following set of equations:

$$29 \quad \alpha_1 = y_1^t y_1 + \frac{y_2^t y_2}{4} - y_1^t y_2$$

$$\alpha_2 = y_1^t y_2 - \frac{y_2^t y_2}{2}$$

$$\alpha_3 = \frac{y_2^t y_2}{4}$$

$$\alpha_4 = x^t y_1 - \frac{x^t y_2}{2}$$

$$\alpha_5 = \frac{x^t y_2}{2}$$

$$\alpha = \frac{\alpha_2 \alpha_5 - 2 \alpha_4 \alpha_3}{\alpha_2 \alpha_4 - 2 \alpha_1 \alpha_5}$$

30 The parameter  $\alpha$  is bounded between 1.0 and 0.55. The adaptive codebook gain can now be found  
 31 by:

32

$$g_p = \frac{\sum_{n=0}^{63} x(n)y(n)}{\sum_{n=0}^{63} y(n)y(n)}, \quad \text{bounded by } 0 \leq g_p \leq 1.2. \quad (5.16.9-1)$$

This process is inherently different from the RS-II modes in that the filter coefficients are found without the need for explicit filtering operations. That is, the coefficients are determined directly from the available data, and are chosen to minimize the mean squared error between the filtered ACB vector  $y(n)$  and the ACB target vector  $x(n)$ . The overall result is that the LTP filter is implemented as a constrained multi-tap LTP filter, using a sub-sample resolution delay and an ACB implementation. The coefficients of this filter, however, are not directly quantized, but are re-optimized as part of the gain VQ after the algebraic codebook search has been performed, as described in section 5.20.

## 5.17 Algebraic Codebook for FR, Voiced HR, and Generic HR

**Routine Name:** `inov_encode`, `find_targets`, `fs_flag`

**Inputs:**

- $r(n)$ : The residual signal
- $x(n)$ : The target signal for the adaptive codebook search
- $v(n)$ : The adaptive codevector
- $y(n)$ : The filtered adaptive codevector
- $h(n)$ : The impulse response of the weighted synthesis filter
- Integer and fractional closed-loop pitch lag or the signal modification pitch delay parameter at the frame boundary
- $g_p$ : The adaptive codebook gain
- $\hat{a}_i$ : The quantized interpolated LP filter coefficients
- $\bar{N}_f$ : Long-term average noise energy
- $flag_{F(z)}$ : formant shaping flag

**Outputs:**

- $c(n)$ : The algebraic codevector
- $z(n)$ : The filtered algebraic codevector
- $flag_{F(z)}$ : formant shaping flag

**Initialization:**

- None

### 5.17.1 Codebook Structure

The codebook structure is based on interleaved single-pulse permutation (ISPP) design. The 64 positions in the codevector are divided into 4 tracks of interleaved positions, with 16 positions in each

1 track. The different codebooks at the different rates are constructed by placing a certain number of  
 2 signed pulses in the tracks (from 1 to 6 pulses per track). The codebook index, or codeword,  
 3 represents the pulse positions and signs in each track. Thus, no codebook storage is needed, since  
 4 the excitation vector at the decoder can be constructed through the information contained in the index  
 5 itself (no lookup tables).

6  
 7 An important feature of this codebook is that it is a dynamic codebook, whereby the algebraic  
 8 codevectors are filtered through an adaptive pre-filter  $F(z)$ . The transfer function of the adaptive pre-  
 9 filter varies in time in relation to parameters representative of spectral characteristics of the signal to  
 10 shape frequency characteristics of the excitation signal to damp frequencies perceptually annoying to  
 11 the human ear. Here, a pre-filter relevant to wideband signals is used whereby  $F(z)$  consists of two  
 12 parts: a periodicity enhancement part  $1/(1 - 0.85z^{-T})$  and a tilt part  $(1 - \beta_1 z^{-1})$ . That is,  
 13

$$14 \quad F(z) = \frac{1 - \beta_1 z^{-1}}{1 - 0.85z^{-T}} \quad (5.17.1-1)$$

15  
 16 The periodicity enhancement part of the filter colors the spectrum by damping inter-harmonic  
 17 frequencies, which are annoying to the human ear in case of voiced signals. In case of FR and  
 18 Generic HR coding types,  $T$  is the integer part of the pitch lag (representing the fine spectral structure  
 19 of the speech signal). In case of Voiced HR,  $T$  is computed on a sample basis following the pitch  
 20 contour.

21  
 22 The factor  $\beta_1$  of the tilt part of the pre-filter is related to the voicing of the previous subframe and is  
 23 bounded by  $[0.0, 0.5]$ . It is computed as  
 24

$$25 \quad \beta_1 = \frac{0.5E'_v}{E'_v + E'_c} \quad (5.17.1-2)$$

26  
 27 where  $E'_v$  and  $E'_c$  are the energies of the scaled pitch codevector and scaled innovation codevector of  
 28 the previous subframe, respectively. The role of the tilt part is to reduce the excitation energy at low  
 29 frequencies in case of voiced frames.

30  
 31 In Rate-Set I operation, the adaptive pre-filter  $F(z)$  also includes a filter based on the spectral  
 32 envelope which colors the spectrum by damping frequencies between the formant regions. In this  
 33 case the adaptive pre-filter is given by  
 34

$$35 \quad F(z) = \frac{1 - \beta_1 z^{-1} \hat{A}(z/\eta_1)}{1 - 0.85z^{-T} \hat{A}(z/\eta_2)} \quad (5.17.1-3)$$

36  
 37 where  $\eta_1=0.75$  and  $\eta_2=0.9$ . However the additional formant shaping part of the filter  $F(z)$  is  
 38 inactivated in presence of high-level noise to limit distortion. The switching between formant shaping  
 39 enabled and disabled states depends on the long-term average noise  $\bar{N}_f$ . It is signaled to the  
 40 decoder using Rate-Set I Unvoiced HR frames and it applies to all subsequent Rate-Set I Generic or  
 41 Voiced FR or HR frames until next switching is signaled. That is, a formant shaping flag  $flag_{F(z)}$  is set  
 42 to 0 (formant shaping disabled) if a Rate-Set I Unvoiced HR frame is used in the encoder and the  
 43 long-term average noise  $\bar{N}_f > 29$ . Similarly,  $flag_{F(z)}$  is set to 1 if an Unvoiced HR frame is used and  
 44  $\bar{N}_f < 20$ . This hysteresis prevents repeated switching of  $flag_{F(z)}$ .

45  
 46 The codebook search is performed in the algebraic domain by combining the filter  $F(z)$  with the  
 47 weighed synthesis filter prior to the codebook search. Thus, the impulse response  $h(n)$  must be

1 modified to include the pre-filter  $F(z)$ . That is,  $h(n) \leftarrow h(n) * f(n)$ . The codebook structures of  
 2 different encoding types are given below.

### 3 5.17.1.1 FR Encoding Types

4  
 5 In this codebook, the innovation vector contains 8 non-zero pulses. All pulses can have the  
 6 amplitudes +1 or -1. The 64 positions in a subframe are divided into 4 tracks, where each track  
 7 contains two pulses, as shown in Table 5.17-1.

12 **Table 5.17-1: Potential positions of individual pulses in the algebraic codebook in FR**

Track	Pulse	Positions
1	$i_0, i_4$	0, 4, 8, 12, 16, 20, 24, 28, 32, 36, 40, 44, 48, 52, 56, 60
2	$i_1, i_5$	1, 5, 9, 13, 17, 21, 25, 29, 33, 37, 41, 45, 49, 53, 57, 61
3	$i_2, i_6$	2, 6, 10, 14, 18, 22, 26, 30, 34, 38, 42, 46, 50, 54, 58, 62
4	$i_3, i_7$	3, 7, 11, 15, 19, 23, 27, 31, 35, 39, 43, 47, 51, 55, 59, 63

13 Each two-pulse position in one track is encoded with 8 bits (total of 32 bits, 4 bits for the position of  
 14 every pulse), and the sign of the first pulse in the track is encoded with 1 bit (total of 4 bits). This gives  
 15 a total of 36 bits for the algebraic code.

16  
 17 In the case of two pulses per track of  $K = 2^M$  potential positions (here  $M = 4$ ), each pulse needs 1 bit  
 18 for the sign and  $M$  bits for the position, which gives a total of  $2M + 2$  bits. However, some redundancy  
 19 exists due to the unimportance of the pulse ordering. For example, placing the first pulse at position  $p$   
 20 and the second pulse at position  $q$  is equivalent to placing the first pulse at position  $q$  and the second  
 21 pulse at position  $p$ . One bit can be saved by encoding only one sign and deducing the second sign  
 22 from the ordering of the positions in the index. Here the index is given by

$$I_{2p} = p_1 + p_0 \times 2^M + s \times 2^{2M} \quad (5.17.1.1-1)$$

24  
 25 where  $s$  is the sign index of the pulse at position index  $p_0$ . If the two signs are equal then the smaller  
 26 position is set to  $p_0$  and the larger position is set to  $p_1$ . On the other hand, if the two signs are not  
 27 equal, then the larger position is set to  $p_0$  and the smaller position is set to  $p_1$ . At the decoder, the sign  
 28 of the pulse at position  $p_0$  is readily available. The second sign is deduced from the pulse ordering. If  
 29  $p_0$  is larger than  $p_1$  then the sign of the pulse at position  $p_1$  is opposite to that at position  $p_0$ . If this is  
 30 not the case, then the two signs are set equal

### 31 5.17.1.2 Voiced HR and Generic HR Encoding Types

32  
 33 In this codebook, the innovation vector contains 2 non-zero pulses. All pulses can have the  
 34 amplitudes +1 or -1. The 64 positions in a subframe are divided into 2 tracks, where each track  
 35 contains one pulse, as shown in Table 5.17-2.

36 **Table 5.17-2: Potential positions of individual pulses in the algebraic codebook for Voiced HR**  
 37 **and Generic HR**

Track	Pulse	Positions
1	$i_0$	0, 2, 4, 6, 8, 10, 12, 14, 16, 18, 20, 22, 24, 26, 28, 30, 32, 34, 36, 38, 40, 42, 44, 46, 48, 50, 52, 54, 56, 58, 60, 62
2	$i_1$	1, 3, 5, 7, 9, 11, 13, 15, 17, 19, 21, 23, 25, 27, 29, 31, 33, 35, 37, 39, 41, 43, 45, 47, 49, 51, 53, 55, 57, 59, 61, 63

1 Each pulse position in one track is encoded with 5 bits and the sign of the pulse in the track is  
2 encoded with 1 bit. This gives a total of 12 bits for the algebraic code.

3

4 The position index is given by the pulse position in the subframe divided by the pulse spacing (integer  
5 division). The division remainder gives the track index. For example, a pulse at position 31 has a  
6 position index of  $31/2 = 15$  and it belongs to the track with index 1 (second track). The sign index here  
7 is set to 0 for positive signs and 1 for negative signs. The index of the signed pulse is given by

$$8 \quad I_{ip} = p + s \times 2^M \quad (5.17.1.2-1)$$

9

10 where  $p$  is the position index,  $s$  is the sign index, and  $M = 5$  is the number of bits per track.

### 11 5.17.1.3 RS-I FR Encoding Types

12

13 In this 20-bit codebook, the innovation vector contains 4 non-zero pulses. All pulses can have the  
14 amplitudes +1 or -1. The 64 positions in a subframe are divided into 4 tracks, where each track  
15 contains one pulse, as shown in Table 5.17-3.

16 **Table 5.17-3: Potential positions of individual pulses in the algebraic codebook in RS-I FR**

Track	Pulse	Positions
1	$i_0$	0, 4, 8, 12, 16, 20, 24, 28, 32, 36, 40, 44, 48, 52, 56, 60
2	$i_1$	1, 5, 9, 13, 17, 21, 25, 29, 33, 37, 41, 45, 49, 53, 57, 61
3	$i_2$	2, 6, 10, 14, 18, 22, 26, 30, 34, 38, 42, 46, 50, 54, 58, 62
4	$i_3$	3, 7, 11, 15, 19, 23, 27, 31, 35, 39, 43, 47, 51, 55, 59, 63

17 Each pulse position in one track is encoded with 4 bits and the sign of the pulse is encoded with 1 bit,  
18 this gives a total of 20 bits for the algebraic code.

19

20 The position index is given by the pulse position in the subframe divided by the pulse spacing (integer  
21 division). The division remainder gives the track index. For example, a pulse at position 31 has a  
22 position index of  $31/4 = 7$  and it belongs to the track with index 3 (fourth track). The sign index here is  
23 set to 0 for positive signs and 1 for negative signs. The index of the signed pulse is given by

$$24 \quad I_{ip} = p + s \times 2^M \quad (5.17.1.2-1)$$

25

26 where  $p$  is the position index,  $s$  is the sign index, and  $M = 4$  is the number of bits per track.

### 27 5.17.1.4 RS-I Generic HR and Voiced HR Encoding Types

28

29 In RS-I Generic and Voiced HR encoding types, the same 12-bit codebook described in section  
30 5.17.1.2 is used. However, the codebook is used only in one out of two subframes (once per half-  
31 frame)

32

### 5.17.2 Algebraic Codebook Search

The algebraic codebook is searched by minimizing the mean square error between the weighted input speech and the weighted synthesis speech. The target signal used in the closed-loop pitch search is updated by subtracting the adaptive codebook contribution. That is

$$x_2(n) = x(n) - g_p y(n), \quad n = 0, \dots, 63 \quad (5.17.2-1)$$

where  $y(n) = v(n)*h(n)$  is the filtered adaptive codebook vector and  $g_p$  is the unquantized adaptive codebook gain.

The matrix  $\mathbf{H}$  is defined as the lower triangular Toeplitz convolution matrix with diagonal  $h(0)$  and lower diagonals  $h(1), \dots, h(63)$ , and  $\mathbf{d} = \mathbf{H}^t \mathbf{x}_2$  is the correlation between the target signal  $x_2(n)$  and the impulse response  $h(n)$  (also known as the backward filtered target vector), and  $\Phi = \mathbf{H}^t \mathbf{H}$  is the matrix of correlations of  $h(n)$ . Here,  $h(n)$  is the impulses response of the combination of the synthesis filter, the weighting filter, and the pre-filter  $F(z)$  which includes a long-term filter.

The elements of the vector  $\mathbf{d}$  are computed by

$$d(n) = \sum_{i=n}^{63} x_2(i)h(i-n), \quad n = 0, \dots, 63 \quad (5.17.2-2)$$

and the elements of the symmetric matrix  $\Phi$  are computed by

$$\phi(i, j) = \sum_{n=j}^{63} h(n-i)h(n-j), \quad i = 0, \dots, 63 \quad j = i, \dots, 63 \quad (5.17.2-3)$$

If  $\mathbf{c}_k$  is the algebraic codevector at index  $k$ , then the algebraic codebook is searched by maximizing the search criterion

$$Q_k = \frac{(\mathbf{x}_2^t \mathbf{H} \mathbf{c}_k)^2}{\mathbf{c}_k^t \mathbf{H}^t \mathbf{H} \mathbf{c}_k} = \frac{(\mathbf{d}^t \mathbf{c}_k)^2}{\mathbf{c}_k^t \Phi \mathbf{c}_k} = \frac{(R_k)^2}{E_k} \quad (5.17.2-4)$$

The vector  $\mathbf{d}$  and the matrix  $\Phi$  are usually computed prior to the codebook search.

The algebraic structure of the codebooks allows for very fast search procedures since the innovation vector  $\mathbf{c}_k$  contains only a few nonzero pulses. The correlation in the numerator of Equation (5.17.2-4) is given by

$$R = \sum_{i=0}^{N_p-1} s_i d(m_i) \quad (5.17.2-5)$$

where  $m_i$  is the position of the  $i$ th pulse,  $s_i$  is its amplitude (sign), and  $N_p$  is the number of pulses. The energy in the denominator of Equation (5.17.2-4) is given by

$$E = \sum_{i=0}^{N_p-1} \phi(m_i, m_i) + 2 \sum_{i=0}^{N_p-2} \sum_{j=i+1}^{N_p-1} s_i s_j \phi(m_i, m_j) \quad (5.17.2-6)$$

#### 5.17.2.1 Codebook Search in FR Encoding Types

As explained above, a 36-bit algebraic codebook is used in Full-Rate, thereby efficient non-exhaustive search procedures are used to simplify the search. These procedures consist of signal-

1 selected pulse amplitude for pre-setting the signs and depth-first tree search for determining the pulse  
2 positions. These procedures will be described below.

3  
4 To simplify the search procedure, the pulse amplitudes are predetermined based on a certain  
5 reference signal  $b(n)$ . In this signal-selected pulse amplitude approach, the sign of a pulse at position  
6  $i$  is set equal to the sign of the reference signal at that position. Here, the reference signal  $b(n)$  is  
7 given by

$$8 \quad b(n) = \sqrt{\frac{E_d}{E_r}} r_{LTP}(n) + \alpha d(n) \quad (5.17.2.1-1)$$

9 where  $E_d = \mathbf{d}^t \mathbf{d}$  is the energy of the signal  $d(n)$  and  $E_r = \mathbf{r}_{LTP}^t \mathbf{r}_{LTP}$  is the energy of the signal  $r_{LTP}(n)$   
10 which is the residual signal after long-term prediction, and the entire amplitude information is given in  
11 terms of the sign of a pulse at position  $i$  being either +1 or -1. The scaling factor  $\alpha$  controls the  
12 amount of dependence of the reference signal on  $d(n)$ , and it is lowered as the bit rate is increased.  
13 Here  $\alpha = 1$  is used in FR encoding types.

14  
15 To simplify the search the signal  $d(n)$  and matrix  $\Phi$  are modified to incorporate the pre-selected signs.  
16 Let  $s_b(n)$  denote the vector containing the signs of  $b(n)$ . The modified signal  $d'(n)$  is given by

$$17 \quad d'(n) = s_b(n)d(n) \quad n = 0, \dots, N-1 \quad (5.17.2.1-2)$$

18  
19 and the modified autocorrelation matrix  $\Phi'$  is given by

$$21 \quad \phi'(i, j) = s_b(i)s_b(j)\phi(i, j), \quad i = 0, \dots, N-1; \quad j = i, \dots, N-1 \quad (5.17.2.1-3)$$

22  
23 The correlation at the numerator of the search criterion  $Q_k$  is now given by

$$24 \quad R = \sum_{i=0}^{N_p-1} d'(m_i) \quad (5.17.2.1-4)$$

25  
26 and the energy at the denominator of the search criterion  $Q_k$  is given by

$$28 \quad E = \sum_{i=0}^{N_p-1} \phi'(m_i, m_i) + 2 \sum_{i=0}^{N_p-2} \sum_{j=i+1}^{N_p-1} \phi'(m_i, m_j) \quad (5.17.2.1-5)$$

29  
30 Once the amplitudes have been pre-selected and incorporated into  $d'(n)$  and  $\phi'(i, j)$ , the search will  
31 be confined to a subset of position/amplitude combinations by searching only the pulse  
32 amplitude/position combinations having non-zero-amplitude pulses with respect to the pre-selected  
33 amplitudes. Therefore, only those codewords having pulse positions in which the nonzero pulses  
34 agree in sign with the corresponding positions in  $b(n)$  are used in computing  $R$  and  $E$  to obtain the  
35 search criterion  $Q = (R)^2 / E$ .

36  
37 Thus, the goal of the search now is to determine the codevector with the best set of  $N_p$  pulse  
38 positions assuming amplitudes of the pulses have been selected as described above. The basic  
39 selection criterion is the maximization of the above-mentioned ratio  $Q_k$ .

40  
41 In order to reduce the search complexity, a fast search procedure known as depth-first tree search  
42 procedure is used, whereby the pulse positions are determined  $N_m$  pulses at a time. More precisely,

1 the  $N_p$  available pulses are partitioned into  $M$  non-empty subsets of  $N_m$  pulses respectively such that  
 2  $N_1+N_2...+N_m...+N_M = N_p$ . A particular choice of positions for the first  $J = N_1+N_2...+N_{m-1}$  pulses  
 3 considered is called a level- $m$  path or a path of length  $J$ . The basic criterion for a path of  $J$  pulse  
 4 positions is the ratio  $Q_k(J)$  when only the  $J$  relevant pulses are considered.

5  
 6 The search begins with subset #1 and proceeds with subsequent subsets according to a tree  
 7 structure whereby subset  $m$  is searched at the  $m$ th level of the tree. The purpose of the search at  
 8 level 1 is to consider the  $N_1$  pulses of subset #1 and their valid positions in order to determine one, or  
 9 a number of, candidate path(s) of length  $N_1$ , which are the tree nodes at level 1. The path at each  
 10 terminating node of level  $m-1$  is extended to length  $N_1+N_2...+N_m$  at level  $m$  by considering  $N_m$  new  
 11 pulses and their valid positions. One, or a number of, candidate extended path(s) are determined to  
 12 constitute level- $m$  nodes. The best codevector corresponds to that path of length  $N_p$  which maximizes  
 13 the criterion  $Q_k(N_p)$  with respect to all level- $M$  nodes.

14  
 15 A special form of the depth-first tree search procedure is used here, in which two pulses are searched  
 16 at a time, that is,  $N_m = 2$ , and these 2 pulses belong to two consecutive tracks. Further, instead of  
 17 assuming that the matrix  $\Phi$  is pre-computed and stored, which requires a memory of  $N \times N$  words  
 18 ( $64 \times 64 = 4k$  words); a memory-efficient approach is used which reduces the memory requirement.  
 19 In this approach, the search procedure is performed in such a way that only a part of the needed  
 20 elements of the correlation matrix are pre-computed and stored. This part corresponds to the  
 21 correlations of the impulse response corresponding to potential pulse positions in consecutive tracks,  
 22 as well as the correlations corresponding to  $\phi(j,j)$ ,  $j = 0, \dots, N-1$  (that is the elements of the main  
 23 diagonal of matrix  $\Phi$ ).

24  
 25 In order to reduce the complexity, while testing possible combinations of two pulses, a limited number  
 26 of potential positions of the first pulse are tested. Further, in case of a large number of pulses, some  
 27 pulses in the higher levels of the search tree are fixed. In order to guess intelligently which potential  
 28 pulse positions are considered for the first pulse, or in order to fix some pulse positions, a  
 29 "pulse-position likelihood-estimate vector"  $\mathbf{b}$  is used, which is based on speech-related signals. The  
 30  $p$ th component  $b(p)$  of this estimate vector  $\mathbf{b}$  characterizes the probability of a pulse occupying  
 31 position  $p$  ( $p = 0, 1, \dots, N-1$ ) in the best codevector that is being searched. Here the estimate vector  $\mathbf{b}$   
 32 is the same vector used for pre-selecting the amplitudes and given in Equation (5.17.2.1-1).

33  
 34 The search procedures for all bit rate modes are similar. Two pulses are searched at a time, and  
 35 these two pulses always correspond to consecutive tracks. That is the two searched pulses are in  
 36 tracks  $T_0-T_1$ ,  $T_1-T_2$ ,  $T_2-T_3$ , or  $T_3-T_0$ . Before searching the positions, the sign of a pulse at potential  
 37 position  $n$  is set to the sign of  $b(n)$  at that position. Then the modified signal  $d'(n)$  is computed as  
 38 described above by including the predetermined signs.

39  
 40 For the first 2 pulses (1<sup>st</sup> tree level), the correlation at the numerator of the search criterion is given  
 41 by

$$42 \quad R = d'(m_0) + d'(m_1) \quad (5.17.2.1-6)$$

43  
 44 and the energy at the denominator of the search criterion  $Q_k$  is given by

$$46 \quad E = \phi'(m_0, m_0) + \phi'(m_1, m_1) + 2\phi'(m_0, m_1) \quad (5.17.2.1-7)$$

47  
 48 where the correlations  $\phi'(m_i, m_j)$  has been modified to include the pre-selected signs at positions  $m_i$   
 49 and  $m_j$ .

50  
 51 For subsequent levels, the numerator and denominator are updated by adding the contribution of two  
 52 new pulses. Assuming that two new pulses at a certain tree level with positions  $m_k$  and  $m_{k+1}$  from two  
 53 consecutive tracks are searched, then the updated value of  $R$  is given by

1

$$R = R + d'(m_k) + d'(m_{k+1}) \quad (5.17.2.1-8)$$

3

4 and the updated energy is given by

5

$$E = E + \phi'(m_k, m_k) + \phi'(m_{k+1}, m_{k+1}) + 2\phi'(m_k, m_{k+1}) + 2R_{hv}(m_k) + 2R_{hv}(m_{k+1}) \quad (5.17.2.1-9)$$

8

9 where  $R_{hv}(m)$  is the correlation between the impulse response  $h(n)$  and a vector  $v_h(n)$  containing the  
10 addition of delayed versions of impulse response at the previously determined positions. That is,

$$v_h(n) = \sum_{i=0}^{k-1} h(n - m_i) \quad (5.17.2.1-10)$$

12 and

$$R_{hv}(m) = \sum_{n=m}^{N-1} h(n)v_h(n - m) \quad (5.17.2.1-11)$$

14

15 At each tree level, the values of  $R_{hv}(m)$  are computed online for all possible positions in each of the  
16 two tracks being tested. It can be seen from Equation (5.17.2.1-9) that only the correlations  
17  $\phi'(m_k, m_{k+1})$  corresponding to pulse positions in two consecutive tracks need to be stored  
18 ( $4 \times 16 \times 16$  words), along with the correlations  $\phi'(m_k, m_k)$  corresponding to the diagonal of the  
19 matrix  $\Phi$  (64 words). Thus the memory requirement in the present algebraic structure is 1088 words  
20 instead of  $64 \times 64 = 4096$  words.

21

22 In the FR encoding types, 2 pulses are placed in each track giving a total of 8 pulses per subframe of  
23 length 64. Two pulses are searched at a time, and these two pulses always correspond to  
24 consecutive tracks. That is the two searched pulses are in tracks  $T_0-T_1$ ,  $T_1-T_2$ ,  $T_2-T_3$ , or  $T_3-T_0$ . The  
25 tree has 4 levels in this case. At the first level, pulse  $P_0$  is assigned to track  $T_0$  and pulse  $P_1$  to track  
26  $T_1$ . In this level, no search is performed and the two pulse positions are set to the maximum of  $b(n)$  in  
27 each track. In the second level, pulse  $P_2$  is assigned to track  $T_2$  and pulse  $P_3$  to track  $T_3$ . 4 positions  
28 for pulse  $P_2$  are tested against all 16 positions of pulse  $P_3$ . The 4 tested positions of  $P_2$  are  
29 determined based on the maxima of  $b(n)$  in the track. In the third level, pulse  $P_4$  is assigned to track  
30  $T_1$  and pulse  $P_5$  to track  $T_2$ . 8 positions for pulse  $P_4$  are tested against all 16 positions of pulse  $P_5$ .  
31 Similar to the previous search level, the 8 tested positions of  $P_4$  are determined based on the maxima  
32 of  $b(n)$  in the track. In the fourth level, pulse  $P_6$  is assigned to track  $T_3$  and pulse  $P_7$  to track  $T_0$ . 8  
33 positions for pulse  $P_6$  are tested against all 16 positions of pulse  $P_7$ . Thus the total number of tested  
34 combinations is  $4 \times 16 + 8 \times 16 + 8 \times 16 = 320$ . The whole process is repeated 4 times (4 iterations)  
35 by assigning the pulses to different tracks. For example, in the 2nd iteration, pulses  $P_0$  to  $P_7$  are  
36 assigned to tracks  $T_1, T_2, T_3, T_0, T_2, T_3, T_0,$  and  $T_1$ , respectively. Thus the total number of tested  
37 position combinations is  $4 \times 320 = 1280$ .

38

39 Once the pulse positions and signs are determined, the algebraic codevector is constructed then the  
40 fixed-codebook excitation vector is found by filtering the algebraic codevector through the pre-filter  
41  $F(z)$ .

#### 42 5.17.2.2 Codebook Search in Voiced HR and Generic HR

43

44 In the Voiced HR and Generic HR encoding types, as explained above a 12 bit codebook is used.  
45 Due to the reasonable codebook size, exhaustive search is used. Since there are only two pulses, the

1 computation of the correlation and energy terms in Equation (5.17.2-4) is simple. The correlation in  
2 the numerator of Equation (5.17.2-6) is given by

$$3 \quad R = s_0 d(m_0) + s_1 d(m_1) \quad (5.17.2.2-1)$$

4  
5 and the energy at the denominator of the search criterion  $Q_k$  is given by  
6

$$7 \quad E = \phi(m_0, m_0) + \phi(m_1, m_1) + 2s_0 s_1 \phi(m_0, m_1) \quad (5.17.2.2-2)$$

8  
9 The search criterion is computed for all possible position and sign combinations to find the optimum  
10 pulse positions and signs. The correlations of the impulse response corresponding to potential pulse  
11 positions in the two tracks are only computed and stored, as well as the correlations corresponding to  
12  $\phi(j,j)$ ,  $j = 0, \dots, 63$  (that is the elements of the main diagonal of matrix  $\Phi$ ). This gives a total of  
13  $32 \times 32 + 64 = 1088$  as in the FR case.

14  
15 Once the pulse positions and signs are determined, the algebraic codevector is constructed then the  
16 fixed-codebook excitation vector is found by filtering the algebraic codevector through the pre-filter  
17  $F(z)$ .  
18

### 19 **5.17.2.3 Codebook Search in RS-I Generic FR and RS-I Voiced FR**

20  
21 In Rate-Set I, a joint parameter search approach proposed in [28,34] is used for the algebraic  
22 codebook search. This method jointly optimizes the adaptive codebook and the fixed codebook gains  
23 together with the fixed codebook excitation. This is accomplished by modifying the correlation matrix  
24  $\Phi$  based on the filtered adaptive codebook excitation  $y(n)$  thereby generating a processing matrix  
25  $\tilde{\Phi}$ .

26 The steps for generating the processing matrix used in the joint parameter search are as follows:  
27

- 28 1. A vector  $\mathbf{d}_c = \mathbf{H}'\mathbf{y}$  is generated, which is defined as the correlation between the filtered  
29 adaptive codebook vector  $y(n)$  and the impulse response  $h(n)$  (or backward filtered ACB  
30 vector).
- 31 2. A joint search weighting factor  $\lambda_j$  is computed which is given by

$$32 \quad \lambda_j = \begin{cases} 0, & g_p = 0 \\ 1.0, & g_p < 1.2 \\ 0.4, & g_p = 1.2 \end{cases} \quad (5.17.2.3-1)$$

33 When  $g_p$  is clipped at 0.95 then  $\lambda_j$  is set to 0.4.

- 34 3. The energy  $M$  of  $y(n)$  given by

$$35 \quad M = \sum_{i=0}^{63} y(i)y(i) \quad (5.17.2.3-2)$$

36 is computed and if the energy  $M$  is less than a predefined threshold  $M_{th}$  then the energy  $M$  is  
37 modified using the following equation

$$38 \quad M = M + 0.2M \left(1 - \frac{M}{M_{th}}\right) \quad (5.17.2.3-3)$$

- 39 4. The matrix  $\tilde{\Phi}$  is computed by

$$\tilde{\Phi} = \Phi - \lambda_j \frac{\mathbf{d}_c \mathbf{d}_c^t}{M}. \quad (5.17.2.3-4)$$

and the elements of the symmetric matrix  $\tilde{\Phi}$  are generated by

$$\tilde{\phi}(i, j) = \phi(i, j) - \lambda_j \frac{d_c(i)d_c(j)}{M} \quad (5.17.2.3-5)$$

In Rate-Set I,  $\Phi$  represents  $\tilde{\Phi}$  i.e.,  $\Phi \leftarrow \tilde{\Phi}$  in the subsequent descriptions.

Then as described in Section 5.17.2, the search maximizes the term  $Q=R^2/E$  where the correlation  $R$  is given by

$$R = \sum_{i=0}^{N_p-1} s_i d(m_i) \quad (5.17.2.3-6)$$

and the energy  $E$  in the denominator is given by

$$E = \sum_{i=0}^{N_p-1} \phi(m_i, m_i) + 2 \sum_{i=0}^{N_p-2} \sum_{j=i+1}^{N_p-1} s_i s_j \phi(m_i, m_j) \quad (5.17.2.3-7)$$

Further, to simplify the search procedure, the pulse amplitudes are predetermined based on a certain reference signal  $b(n)$ . In this signal-selected pulse amplitude approach, the sign of a pulse at position  $i$  is set equal to the sign of the reference signal at that position. Here, the reference signal  $b(n)$  is given by

$$b(n) = \sqrt{\frac{E_d}{E_r}} r_{LTP}(n) + \alpha d(n) \quad (5.17.2.3-8)$$

where  $E_d = \mathbf{d}^t \mathbf{d}$  is the energy of the signal  $d(n)$  and  $E_r = \mathbf{r}_{LTP}^t \mathbf{r}_{LTP}$  is the energy of the signal  $r_{LTP}(n)$  which is the residual signal after long-term prediction, and the entire amplitude information is given in terms of the sign of a pulse at position  $i$  being either +1 or -1. The scaling factor  $\alpha$  controls the amount of dependence of the reference signal on  $d(n)$ , and it is lowered as the bit rate is increased. Here  $\alpha = 1$  is used in FR encoding types.

To simplify the search the signal  $d_c(n)$  and matrix  $\Phi$  are modified to incorporate the pre-selected signs. Let  $s_b(n)$  denote the vector containing the signs of  $b(n)$ . The modified signal  $d'(n)$  is given by

$$d'_c(n) = \sqrt{\frac{w_j}{M}} s_b(n) d_c(n), \quad n = 0, \dots, N-1 \quad (5.17.2.3-9)$$

The modified autocorrelation matrix  $\Phi'$  is given by

$$\phi'(i, j) = s_b(i)s_b(j)\phi(i, j), \quad i = 0, \dots, N-1; \quad j = i, \dots, N-1 \quad (5.17.2.3-10)$$

The correlation at the numerator of the search criterion  $Q_k$  is now given by

$$R = \sum_{i=0}^{N_p-1} d'(m_i) \quad (5.17.2.3-11)$$

1  
2 and the energy at the denominator of the search criterion  $Q_k$  is given by  
3

$$4 \quad E = \sum_{i=0}^{N_p-1} \phi'(m_i, m_i) + 2 \sum_{i=0}^{N_p-2} \sum_{j=i+1}^{N_p-1} \phi'(m_i, m_j) \quad (5.17.2.3-12)$$

5  
6 Once the amplitudes have been pre-selected and incorporated into  $d'(n)$  and  $\phi'(i, j)$ , the search will  
7 be confined to a subset of position/amplitude combinations by searching only the pulse  
8 amplitude/position combinations having non-zero-amplitude pulses with respect to the pre-selected  
9 amplitudes. Therefore, only those codewords having pulse positions in which the nonzero pulses  
10 agree in sign with the corresponding positions in  $b(n)$  are used in computing  $R$  and  $E$  to obtain the  
11 search criterion  $Q = (R)^2 / E$ .

12  
13 Thus, the goal of the search now is to determine the codevector with the best set of  $N_p$  pulse  
14 positions assuming amplitudes of the pulses have been selected as described above. The basic  
15 selection criterion is the maximization of the above-mentioned ratio  $Q_k$ .

16  
17 In order to reduce the search complexity, the fast search procedure known as depth-first tree search  
18 procedure is used, as described in Section 5.17.1.1.

19  
20 For the first 2 pulses (1<sup>st</sup> tree level), the correlation at the numerator of the search criterion is given  
21 by

$$22 \quad R = d'(m_0) + d'(m_1) \quad (5.17.2.3-13)$$

23  
24 Further, a denominator modifier is also computed which is given by

$$25 \quad S = d'_c(m_0) + d'_c(m_1)$$

26 and the energy at the denominator of the search criterion  $Q_k$  is given by  
27

$$28 \quad E = \phi'(m_0, m_0) + \phi'(m_1, m_1) + 2\phi'(m_0, m_1) \quad (5.17.2.3-14)$$

29  
30 where the correlations  $\phi'(m_i, m_j)$  has been modified to include the pre-selected signs at positions  $m_i$   
31 and  $m_j$ .

32  
33 For subsequent levels, the numerator and denominator are updated by adding the contribution of two  
34 new pulses. Assuming that two new pulses at a certain tree level with positions  $m_k$  and  $m_{k+1}$  from two  
35 consecutive tracks are searched, then the updated value of  $R$  is given by  
36

$$37 \quad R = R + d'(m_k) + d'(m_{k+1}) \quad (5.17.2.3-15)$$

38  
39 and the updated value of  $S$  is given by  
40

$$41 \quad S = S + d'_c(m_0) + d'_c(m_1)$$

42 and the updated energy is given by  
43

$$E = E + \phi'(m_k, m_k) + \phi'(m_{k+1}, m_{k+1}) + 2\phi'(m_k, m_{k+1}) + 2R_{hv}(m_k) + 2R_{hv}(m_{k+1}) \quad (5.17.2.3-16)$$

where  $R_{hv}(m)$  is the correlation between the impulse response  $h(n)$  and a vector  $v_h(n)$  containing the addition of delayed versions of impulse response at the previously determined positions. That is,

$$v_h(n) = \sum_{i=0}^{k-1} h(n - m_i) \quad (5.17.2.3-17)$$

and

$$R_{hv}(m) = \sum_{n=m}^{N-1} h(n)v_h(n - m) \quad (5.17.2.3-18)$$

A correction term is added to  $R_{hv}(m)$  to get

$$R_{hv}(m) = R_{hv}(m) - Sd'_c(m)$$

At each tree level, the values of  $R_{hv}(m)$  are computed online for all possible positions in each of the two tracks being tested. It can be seen from Equation (5.17.2.3-16) that only the correlations  $\phi'(m_k, m_{k+1})$  corresponding to pulse positions in two consecutive tracks need to be stored ( $4 \times 16 \times 16$  words), along with the correlations  $\phi'(m_k, m_k)$  corresponding to the diagonal of the matrix  $\Phi$  (64 words). Thus the memory requirement in the present algebraic structure is 1088 words instead of  $64 \times 64 = 4096$  words.

In the RS-I Generic FR and RS-I Voiced FR encoding types, a 20-bit codebook is used whereby 1 pulse is placed in each track giving a total of 4 pulses per subframe of length 64. Two pulses are searched at a time, and these two pulses always correspond to consecutive tracks. That is the two searched pulses are in tracks  $T_0-T_1$ ,  $T_1-T_2$ ,  $T_2-T_3$ , or  $T_3-T_0$ . The tree has 2 levels in this case. At the first level, pulse  $P_0$  is assigned to track  $T_0$  and pulse  $P_1$  to track  $T_1$ . In this level, 4 positions for pulse  $P_0$  are tested against all 16 positions of pulse  $P_1$ . The 4 tested positions of  $P_0$  are determined based on the maxima of  $b(n)$  in the track. In the second level, pulse  $P_2$  is assigned to track  $T_2$  and pulse  $P_3$  to track  $T_3$ . 8 positions for pulse  $P_2$  are tested against all 16 positions of pulse  $P_3$ . Similar to the previous search level, the 8 tested positions of  $P_2$  are determined based on the maxima of  $b(n)$  in the track. Thus the total number of tested combinations is  $4 \times 16 + 8 \times 16 = 192$ . The whole process is repeated 4 times (4 iterations) by assigning the pulses to different tracks. For example, in the 2nd iteration, pulses  $P_0$  to  $P_3$  are assigned to tracks  $T_1$ ,  $T_2$ ,  $T_3$ , and  $T_0$ , respectively. Thus the total number of tested position combinations is  $4 \times 192 = 768$ .

Once the pulse positions and signs are determined, the algebraic codevector is constructed then the fixed-codebook excitation vector is found by filtering the algebraic codevector through the pre-filter  $F(z)$ .

#### 5.17.2.4 Codebook Search in RS-I Generic HR and RS-I Voiced HR

The search of the 12-bit codebook is the same as described in Section 5.17.2.2. However, the fixed-codebook contribution is used in one out of two subframes. In the subframes with no fixed-codebook contribution, the pitch gain is forced to a certain value  $g_f$ . The value is determined as the ratio between the energies of the LP synthesis filters in the previous and present frames, constrained to be less or equal to 1. The value of  $g_f$  is given by

$$g_f = \frac{\sum_{n=0}^{127} h_{LPold}^2(n)}{\sum_{n=0}^{127} h_{LPnew}^2(n)} \quad \text{Constrained by } g_f \leq 1. \quad (5.17.2.4-1)$$

where  $h_{LPold}(n)$  and  $h_{LPnew}(n)$  denote the impulse responses of the previous and present frames, respectively. For stable voiced segments, the value of  $g_f$  is close to 1. Determining  $g_f$  using the ratio in Equation (5.17.2.4-1) forces the pitch gain to a low value when the present frame becomes resonant.

As described in Section 5.16.7.1, the subframe in which the pitch gain is forced to  $g_f$  is determined in closed loop by trying both combinations and selecting the one that minimizes the weighted error over the half-frame. This will be described in more details below.

Determining the excitation each two subframes is performed in two iterations. In the first iteration, the excitation is determined in the first subframe as usual. The adaptive codebook excitation and the pitch gain are determined. Then the target signal for fixed codebook search is updated and the fixed codebook excitation and gain are computed, and the adaptive and fixed codebook gains are jointly quantized. In the second subframe, the adaptive codebook memory is updated using the total excitation from the first subframe, then the pitch gain is forced to  $g_f$  and the adaptive codebook excitation is computed with no fixed codebook contribution. Thus, the total excitation from the first iteration in the first subframe is given by:

$$u_{sf1}^{(1)}(n) = \hat{g}_p^{(1)} v_{sf1}^{(1)}(n) + \hat{g}_c^{(1)} c_{sf1}^{(1)}(n), \quad n = 0, \dots, 63 \quad (5.17.2.4-2)$$

and the total excitation in the second subframe is given by

$$u_{sf2}^{(1)}(n) = g_f^{(1)} v_{sf2}^{(1)}(n) \quad n = 0, \dots, 63 \quad (5.17.2.4-3)$$

Before starting the second iteration, the memories of the synthesis and weighting filters and the adaptive codebook memories are saved for the two subframes.

In the second iteration, in the first subframe the pitch gain is forced to  $g_f$  and the adaptive codebook excitation is computed with no fixed codebook contribution. The total excitation in the first subframe is then given by

$$u_{sf1}^{(2)}(n) = g_f^{(2)} v_{sf1}^{(2)}(n) \quad n = 0, \dots, 63 \quad (5.17.2.4-4)$$

Then, the memory of the adaptive codebook and the filter's memories are updated based on the excitation from the first subframe.

In the second subframe, the target signal is computed, and adaptive codebook excitation and pitch gain are determined. Then the target signal is updated and the fixed codebook excitation and gain are computed. The adaptive and fixed codebook gains are jointly quantized. The total excitation in the second subframe is thus given by

$$u_{sf2}^{(2)}(n) = \hat{g}_p^{(2)} v_{sf2}^{(2)}(n) + \hat{g}_c^{(2)} c_{sf2}^{(2)}(n), \quad n = 0, \dots, 63 \quad (5.17.2.4-5)$$

Finally, to decide which iteration to choose, the weighted error is computed for both iterations over the two subframes, and the total excitation corresponding to the iteration resulting in smaller mean-squared weighted error is retained. One bit is used per half-frame to indicate the index of the subframe where fixed codebook contribution is used.

The weighted error for two subframes in the first iteration is given by

$$e_{sf1}^{(1)}(n) = \hat{g}_p^{(1)} y_{sf1}^{(1)}(n) + \hat{g}_c^{(1)} z_{sf1}^{(1)}(n), \quad n = 0, \dots, 63$$

$$e_{sf2}^{(1)}(n) = g_f^{(1)} y_{sf2}^{(1)}(n), \quad n = 0, \dots, 63$$

and the weighted error for two subframes in the second iteration is given by

$$e_{sf1}^{(2)}(n) = g_f^{(2)} y_{sf2}^{(2)}(n), \quad n = 0, \dots, 63$$

$$e_{sf2}^{(2)}(n) = \hat{g}_p^{(2)} y_{sf2}^{(2)}(n) + \hat{g}_c^{(2)} z_{sf2}^{(2)}(n), \quad n = 0, \dots, 63$$

Where  $y(n)$  and  $z(n)$  are the filtered adaptive codebook and filtered fixed codebook, respectively.

In case the first iteration is retained, the saved memories are copied back into the filter memories and adaptive codebook buffer for use in the next half-frame (since after both iterations are performed the filter memories and adaptive codebook buffer correspond to the second iteration).

The same procedure is repeated for the second half-frame (third and fourth subframes). As explained above, before starting the second half-frame excitation search, the proper memories corresponding to the selected iteration are used.

## 5.18 Gaussian Codebook Structure and Search in Unvoiced HR

**Routine Name:** gaus\_encode

**Inputs:**

- $x(n)$ : The target signal
- $h(n)$ : The impulse response of the weighted synthesis filter

**Outputs:**

- $c(n)$ : The Gaussian codevector
- $z(n)$ : The filtered Gaussian codevector
- $g_c$ : The Gaussian codebook gain

**Initialization:**

- None

### 5.18.1 Structure of the random codebook

In Unvoiced HR, a Gaussian codebook is used for representing the excitation. To simplify the search and reduce the codebook memory requirement, an efficient structure is used whereby the excitation codevector is derived by the addition of 2 signed vectors taken from a table containing 64 random vectors of dimension  $N$  (here  $N$  is the subframe size 64). Let  $\mathbf{v}_i$  denote the  $i$ th  $N$ -dimensional random vector in the random table, then a codevector is constructed by

$$\mathbf{c} = s_1 \mathbf{v}_{p_1} + s_2 \mathbf{v}_{p_2} \quad (5.18.1-1)$$

where the signs  $s_1$  and  $s_2$  are signs equal to -1 or 1, and  $p_1$  and  $p_2$  are the indices of the random vectors from the random table. In order to reduce the table memory, a shift-by-2 table is used, thus only  $64+63 \times 2 = 190$  values are needed to represent the 64 vectors of dimension  $N=64$ .

1 To encode the codebook index, one has to encode 2 signs,  $s_1$  and  $s_2$ , and two indices,  $p_1$  and  $p_2$ . The  
 2 values of  $p_1$  and  $p_2$  are in the range 0 to 63, so they need 6 bits each, and the signs need 1 bit each.  
 3 However, 1 bit can be saved since the order of the vectors  $\mathbf{v}_i$  and  $\mathbf{v}_j$  is not important. For example,  
 4 choosing  $\mathbf{v}_{16}$  as the first vector and  $\mathbf{v}_{25}$  as the second vector is equivalent to choosing  $\mathbf{v}_{25}$  as the first  
 5 vector and  $\mathbf{v}_{16}$  as the second vector. Thus, similar to the case of encoding two pulses in a track, only one  
 6 bit can be used for both signs while ordering the vector indices in a way such that the other sign  
 7 information can be easily deduced. This gives a total of 13 bits. To better explain this procedure,  
 8 assume that the two vectors have the indices  $p_1$  and  $p_2$  with sign indices  $\sigma_1$  and  $\sigma_2$ , respectively ( $\sigma=0$   
 9 if the sign is positive and  $\sigma=1$  if the sign is negative). The codevector index is given by

$$10 \quad I = \sigma_1 + 2 \times (p_1 \times 6 + p_2) \quad (5.18.1-2)$$

11  
 12 If  $p_1 \leq p_2$  then  $\sigma_2 = \sigma_1$ ; otherwise  $\sigma_2$  is different from  $\sigma_1$ . Thus, when constructing the codeword (index  
 13 of codevector), if the two signs are equal then the smaller index is assigned to  $p_1$  and the larger index  
 14 to  $p_2$ , otherwise the larger index is assigned to  $p_1$  and the smaller index to  $p_2$ .

### 15 5.18.2 Search of the Random Codebook

16  
 17 The goal of the search procedure is to find the indices  $p_1$  and  $p_2$  of the best 2 vectors and their  
 18 corresponding signs  $s_1$  and  $s_2$ , which maximize the search criterion

$$19 \quad Q_k = \frac{(\mathbf{x}^t \mathbf{z}_k)^2}{\mathbf{z}_k^t \mathbf{z}_k} = \frac{(\mathbf{x}^t \mathbf{H} \mathbf{c}_k)^2}{\mathbf{z}_k^t \mathbf{z}_k} = \frac{(\mathbf{d}^t \mathbf{c}_k)^2}{\mathbf{z}_k^t \mathbf{z}_k} \quad (5.18.1-3)$$

20  
 21 where  $\mathbf{x}$  is the target vector and  $\mathbf{z}_k = \mathbf{H} \mathbf{c}_k$  is the filtered codevector at index  $k$ . Note that in the  
 22 numerator of the search criterion, the dot product between  $\mathbf{x}$  and  $\mathbf{z}_k$  is equivalent to the dot product  
 23 between  $\mathbf{d}$  and  $\mathbf{c}_k$ , where  $\mathbf{d} = \mathbf{H}^t \mathbf{x}$  is the backward filtered target vector which is also the correlation  
 24 between  $\mathbf{d}$  and the impulse response  $\mathbf{h}$ . The elements of the vector  $\mathbf{d}$  are found by

$$25 \quad d(n) = x(n) * h(-n) = \sum_{i=n}^{N-1} x(i) h(i-n) \quad (5.18.1-4)$$

26  
 27 Since  $\mathbf{d}$  is independent of the codevector index  $k$ , it is computed only once, which simplifies the  
 28 computation of the numerator for the different codevectors.

29  
 30 After computing the vector  $\mathbf{d}$ , a pre-determination process is used to identify  $K$  out of the 64 random  
 31 vectors in the random table, so that the search process is then confined to those  $K$  vectors. The pre-  
 32 determination is performed by testing the numerator of the search criterion  $Q_k$  for the  $K$  vectors which  
 33 have the largest absolute dot product (or squared dot product) between  $\mathbf{d}$  and  $\mathbf{v}_i$ ,  $i=0, \dots, 63$ . That is,  
 34 the dot products  $\chi_i$  given by

$$35 \quad \chi_i = \sum_{n=0}^{N-1} d(n) v_i(n) \quad (5.18.1-5)$$

36  
 37 are computed for all random vectors  $\mathbf{v}_i$  and the indices of the  $K$  vectors which result in the  $K$  largest  
 38 values of  $|\chi_i|$  are retained. These indices are stored in the index vector  $m_i$ ,  $i=0, \dots, K-1$ . To further  
 39 simplify the search, the sign information corresponding to each pre-determined vector is also pre-set.  
 40 The sign corresponding to each pre-determined vector is given by the sign of  $\chi_i$  for that vector. These  
 41 pre-set signs are stored in the sign vector  $s_i$ ,  $i=0, \dots, K-1$ .

42  
 43 The codebook search is now confined to the pre-determined  $K$  vectors with their corresponding signs.  
 44 Here, the value  $K=8$  is used, thus the search is reduced to finding the best 2 vectors among 8 random

1 vectors instead of finding them among 64 random vectors. This reduces the number of tested vector  
2 combinations from  $64 \times 65/2 = 2080$  to  $8 \times 9/2 = 36$ .

3  
4 Once the best promising  $K$  vectors and their corresponding signs are pre-determined, the search  
5 proceeds for selecting 2 vectors among those  $K$  vectors which maximize the search criterion  $Q_k$ .

6  
7 We first start by computing and storing the filtered vectors  $\mathbf{w}_j$ ,  $j=0, \dots, K-1$  corresponding to the  $K$  pre-  
8 determined vectors. This can be performed by convolving the pre-determined vectors with the  
9 impulse response of the weighted synthesis filter  $h(n)$ . The sign information is also included in the  
10 filtered vectors; that is

$$11 \quad w_j(n) = s_j \sum_{i=0}^n v_{m_j}(i) h(n-i), \quad n=0, \dots, 63, \quad j=0, \dots, K-1. \quad (5.18.1-6)$$

12  
13 We then compute the energy of each filtered pre-determined vector

$$14 \quad \varepsilon_j = \mathbf{w}_j^t \mathbf{w}_j = \sum_{n=0}^{63} w_j^2(n), \quad j=0, \dots, K-1 \quad (5.18.1-7)$$

15  
16 and its dot product with the target vector

$$17 \quad \rho_j = \mathbf{x}^t \mathbf{w}_j = \sum_{n=0}^{63} w_j(n) x(n), \quad j=0, \dots, K-1. \quad (5.18.1-8)$$

18  
19 Note that  $\rho_j$  and  $\varepsilon_j$  correspond to the numerator and denominator of the search criterion due to each  
20 predetermined vector. The search proceeds now with the selection of 2 vectors among the  $K$  pre-  
21 determined vectors by maximizing the search criterion  $Q_k$ . Note that the codevector is given by

$$22 \quad \mathbf{c} = s_1 \mathbf{v}_{p_1} + s_2 \mathbf{v}_{p_2} \quad (5.18.1-9)$$

23  
24 The filtered codevector  $\mathbf{z}$  is given by

$$25 \quad \mathbf{z} = \mathbf{Hc} = s_1 \mathbf{Hv}_{p_1} + s_2 \mathbf{Hv}_{p_2} = \mathbf{w}_{p_1} + \mathbf{w}_{p_2} \quad (5.18.1-10)$$

26  
27 Note that the pre-determined signs are included in the filtered pre-determined vectors  $\mathbf{w}_i$ . The search  
28 criterion is given by (the codevector index  $k$  is dropped for simplicity)

$$29 \quad Q = \frac{(\mathbf{x}^t \mathbf{z})^2}{\mathbf{z}^t \mathbf{z}} = \frac{(\mathbf{x}^t \mathbf{w}_{p_1} + \mathbf{x}^t \mathbf{w}_{p_2})^2}{(\mathbf{w}_{p_1} + \mathbf{w}_{p_2})^t (\mathbf{w}_{p_1} + \mathbf{w}_{p_2})} = \frac{(\rho_{p_1} + \rho_{p_2})^2}{\varepsilon_{p_1} + \varepsilon_{p_2} + 2\mathbf{w}_{p_1}^t \mathbf{w}_{p_2}} \quad (5.18.1-11)$$

30  
31 The vectors  $\mathbf{w}_j$  and the values of  $\rho_j$  and  $\varepsilon_j$  are computed before starting the codebook search. The  
32 search is performed in two nested loops for all possible positions  $p_1$  and  $p_2$  that maximize the search  
33 criterion  $Q$ . Only the dot products between the different vectors  $\mathbf{w}_j$  need to be computed inside the loop.

34  
35 At the end of the two nested loops, the optimum vector indices  $p_1$  and  $p_2$  will be known. The two  
36 indices and the corresponding signs are then encoded as described above. The gain of the excitation  
37 vector is computed based on a combination of waveform matching and energy matching. The gain is  
38 given by

$$39 \quad g_c = 0.6g_w + 0.4g_e \quad (5.18.1-12)$$

1 where  $g_w$  is the gain that matches the waveforms of the vectors  $\mathbf{x}$  and  $\mathbf{z}$  and given by  $g_w = \frac{\mathbf{x}^t \mathbf{z}}{\mathbf{z}^t \mathbf{z}}$  and  
 2  $g_e$  is the gain that matches the energies of the vectors  $\mathbf{x}$  and  $\mathbf{z}$  and given by  $g_e = \sqrt{\frac{\mathbf{x}^t \mathbf{x}}{\mathbf{z}^t \mathbf{z}}}$ . Here,  $\mathbf{x}$  is  
 3 the target vector and  $\mathbf{z}$  is the filtered selected excitation vector  $\mathbf{c}$  given in Equation (5.18.1-1).

## 4 5.19 Random Excitation in RS-II Unvoiced QR and in RS-I Unvoiced HR

### 5 5.19.1 RS-II Unvoiced QR Random Codebook

6 **Routine Name:** `gaus_encode`

7 **Inputs:**

- 8 •  $x(n)$ : The target signal
- 9 •  $h(n)$ : The impulse response of the weighted synthesis filter

10 **Outputs:**

- 11 •  $c(n)$ : The Gaussian codevector
- 12 •  $z(n)$ : The filtered Gaussian codevector
- 13 •  $g_c$ : The Gaussian codebook gain

14 **Initialization:**

- 15 • None

16 In Unvoiced QR encoding type, no bits are used to encode the excitation signal. The excitation signal  
 17 is derived from the same random table used in the Unvoiced HR encoding type and in the same  
 18 manner. That is, the excitation vector is given by

$$19 \quad \mathbf{c} = s_1 \mathbf{v}_{p_1} + s_2 \mathbf{v}_{p_2} . \quad (5.19-1)$$

20 However, the signs and indices of the two vectors are randomly generated. Since in Unvoiced HR a 13-  
 21 bit random codebook is used, in Unvoiced QR, a 13-bit integer is randomly generated from which the  
 22 indices and signs of the two vectors are derived.

23 To avoid a low-frequency artifact in case of misclassification of a voiced offset as an unvoiced frame, the  
 24 following measure is applied to the randomly generated excitation vector  $\mathbf{c}$ . First the distance between  
 25 the first two ISFs is computed. If this distance is less than 100 Hz, the tilt  $c_{tilt}$  of the excitation vector is  
 26 estimated as

$$27 \quad c_{tilt} = \sum_{i=1}^{N-1} c_{i-1} c_i$$

28 where  $N$  is the subframe length. If  $c_{tilt} > 0$ , the sign of every other sample of the excitation vector is  
 29 reversed ( $c_{2i}$  is multiplied by  $-1$ ) in order to prevent vector  $\mathbf{c}$  of having a low-frequency characteristic.

30 Since no waveform matching is used to determine the excitation vector, the excitation gain is determined  
 31 based only on matching the energies of the target vector  $\mathbf{x}$  and the filtered excitation vector  $\mathbf{z} = \mathbf{c}^* \mathbf{h}$ . The  
 32 excitation gain is given by

$$g_c = \frac{1}{\sqrt{1.84}} \sqrt{\frac{\mathbf{x}'\mathbf{x}}{\mathbf{z}'\mathbf{z}}} . \quad (5.19-2)$$

The scaling factor is introduced for the following reason. In Unvoiced HR the gain is computed as a weighted sum of waveform matching and energy matching. Since in the Unvoiced QR case energy matching alone is performed, the energy of the Unvoiced QR portions of the signal were found to be higher than the case where Unvoiced HR encoding was used for the same portions. The value 1.84 has been found experimentally by comparing the energy of the Unvoiced QR portions with the energy of the same portions coded by Unvoiced HR, using a database consisting of nominal, low and high level signals.

### 5.19.2 Unvoiced Excitation Coding at RS-I Unvoiced HR

Noise Excited Linear Prediction (NELP) is used for RS-I Unvoiced HR frames. A pseudo-random noise is colored using a set of filters, and scaled appropriately by gains in 8 subframes of 32 samples each to model the excitation. The frame is divided into two halves where the 4 gains in each half-frame are jointly quantized.

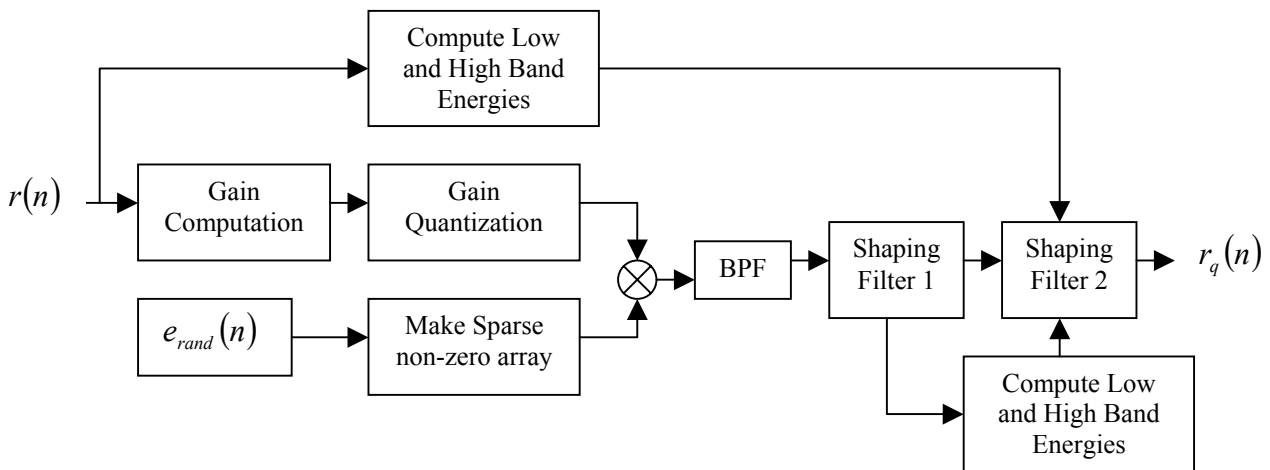


Figure 5.19.2-1: Excitation Coding for RS-I Unvoiced HR Frames

#### 5.19.2.1 Computation and Quantization of Gains

**Routine name:** nelp\_encoder

**Input:**

- Residual,  $r(n)$
- Quantized ISFs,  $isf_1^q(i)$

**Output:**

- Quantized excitation,  $r_q(n)$

The pre-processed speech,  $\tilde{s}(n)$ , is converted to the residual signal  $r(n)$  using the quantized ISFs,  $isf_1^q(i)$  (by converting the ISFs to LPCs, and zero filtering the pre-processed speech with the LPCs).

1 The energy  $E_i$  of the residual signal is computed for each of the 4 subframes ( $0 \leq i < 4$ ) in each  
2 half-frame as

$$3 \quad E_i = \frac{1}{32} \sum_{n=32i}^{32i+31} r^2(n) \quad (5.19.2.1-1)$$

4 The 4 subframe gains  $G_i$  are computed from the above energies as

$$5 \quad G_i = \sqrt{E_i} \quad (5.19.2.1-2)$$

6 A normalization factor for the above 4 gains is determined by

$$7 \quad G' = \sqrt{\frac{1}{4} \sum_{i=0}^4 G_i^2} \quad (5.19.2.1-3)$$

8 The above normalization factor is quantized using a one-dimensional codebook  $UVG1CB[16][1]$  to  
9 provide an index  $i\hat{G}'$  as

$$10 \quad i\hat{G}' = \arg \min_k \left( (\log_{10}(G') - UVG1CB[k])^2 \right), \quad 0 \leq k < 16 \quad (5.19.2.1-4)$$

11 The quantized normalization factors are then computed using

$$12 \quad \hat{G}' = 10^{UVG1CB[i\hat{G}']} \quad (5.19.2.1-5)$$

13 The 4 subframe gains are then normalized using the quantized normalization factors according to

$$14 \quad G''[i] = \frac{G[i]}{\hat{G}'} \quad , \quad 0 \leq i < 4 \quad (5.19.2.1-6)$$

15 The 4 normalized gains are then vector quantized using a 4-dimensional codebook  
16  $UVG2CB[128][4]$  to provide an index  $i\hat{G}''$ , such that

$$17 \quad i\hat{G}'' = \arg \min_k \left( \sum_{i=0}^3 (G''[i] - UVG2CB[k][i])^2 \right), \quad 0 \leq k < 128; \quad (5.19.2.1-7)$$

21 The quantized, normalized gains are then given by

$$22 \quad \hat{G}''[i] = UVG2CB[i\hat{G}''][i] \quad , \quad 0 \leq i < 4 \quad (5.19.2.1-8)$$

23 The quantized gains used to scale the NELP excitation then are

$$24 \quad \hat{G}[i] = \hat{G}' \hat{G}''[i] \quad , \quad 0 \leq i < 4 \quad (5.19.2.1-9)$$

25  
26 This procedure is repeated twice per frame (once for each half-frame).

## 5.19.2.2 Random Number Generation

A pseudo-random vector  $e_{rand}(n)$  is created using zero mean, unit variance Gaussian pseudo-random numbers that are obtained by generating uniform pseudo-random numbers and appropriately transforming them. The following two sub-sections describe the uniform pseudo-random number generator and the algorithm for effecting the uniform to Gaussian transformation.

### 5.19.2.2.1 Uniform Pseudo-Random Number Generation Algorithm

The algorithm for generating the uniform pseudo-random numbers is initialized with a seed value, and produces a new seed value with each successive invocation, as well as producing the desired pseudo-random number (uniformly distributed between -32768 and 32767). The state of the uniform random number generator is captured by the current value of the seed. Different modules making use of the random number generator should maintain their own seeds. Uniform pseudo-random numbers are generated as described in the following pseudo-code:

```
*seed = (short) (*seed * 31821L + 13849L);
```

### 5.19.2.2.2 Gaussian Pseudo-Random Number Generator

The Gaussian pseudo-random number generator makes use of the uniform pseudo-random number generator described in 5.19.2.2.1. Gaussian pseudo-random numbers are generated by means of the addition of several uniform pseudo-random numbers. That is

$$g_{rand} = \frac{1}{scale} \sum_{i=0}^{K-1} u_i$$

where  $u_i$  are uniform pseudo-random variables and  $scale$  is a scaling factor. The value  $scale$  is chosen such that the resulting Gaussian pseudo-random variables have variance equal to 1. As the value of  $K$  is increased the pseudo-random variables  $g_{rand}$  become closer to a Gaussian distribution. To maintain a low complexity, the value  $K=4$  is used. With  $u_i$  uniformly distributed between -32768 and 32767 and with  $K=4$ , the value  $scale = 2^{16} / \sqrt{3}$  results in unit variance distribution. Thus, the Gaussian pseudo-random numbers are generated by

$$g_{rand} = \frac{\sqrt{3}}{2^{16}} \sum_{i=0}^3 u_i$$

### 5.19.2.3 Creation of Sparse Non-Zero Excitation

The above pseudo-random vector  $e_{rand}(n)$  generated in a 128-sample half-frame is divided into 4 subframes, and each subframe of size 32 is made sparse by retaining the 8 largest absolute magnitude samples, and zeroing out the remaining 24. Each of these 4 subframes of the sparse vectors is denoted by  $e_{sparse}(i, n)$ ,  $0 \leq i < 4$ ;  $0 \leq n < 32$ . These sparse vectors are then scaled by their appropriate gains to create 128 samples of a sparse, excitation signal

$$r_2(32i + n) = \hat{G}(i) e_{sparse}(i, n), \quad 0 \leq i < 4; 0 \leq n < 32 \quad (5.19.2.3-1)$$

This procedure is repeated twice per frame (once for each half-frame).

#### 5.19.2.4 Shaping the Excitation

$r_2(n)$  is then filtered by the 10<sup>th</sup> order shaping filter  $SF1(z)$  to give the signal  $r_3(n)$ . This shaping filter is given by the following transfer function

$$SF1(z) = \frac{SF1_{n0} + SF1_{n1}z^{-1} + \dots + SF1_{n12}z^{-10}}{1 + SF1_{d1}z^{-1} + \dots + SF1_{d12}z^{-10}} \quad (5.19.2.4-1)$$

where the coefficients  $SF1_{ni}$ , and  $SF1_{di}$  are given in the tables **shape1\_num\_coef**, and **shape1\_den\_coef** respectively in the C-code.

$r_3(n)$  is then scaled according to the following equation to yield

$$r_4(n) = \left( \sqrt{\frac{\sum_{n=0}^{127} [r(n)]^2}{\sum_{n=0}^{127} [r_3(n)]^2}} \right) r_3(n) \quad (5.19.2.4-2)$$

$r(n)$  and  $r_4(n)$  are filtered by the 10<sup>th</sup> order low-pass filter  $LP(z)$  to give  $r_L(n)$  and  $r_{4L}(n)$ . Similarly,  $r(n)$  and  $r_4(n)$  are filtered by the 10<sup>th</sup> order high-pass filter  $HP(z)$  to give  $r_H(n)$  and  $r_{4H}(n)$ .

$$LP(z) = \frac{LP_{n0} + LP_{n1}z^{-1} + \dots + LP_{n12}z^{-10}}{1 + LP_{d1}z^{-1} + \dots + LP_{d12}z^{-10}} \quad (5.19.2.4-3)$$

$$HP(z) = \frac{HP_{n0} + HP_{n1}z^{-1} + \dots + HP_{n12}z^{-10}}{1 + HP_{d1}z^{-1} + \dots + HP_{d12}z^{-10}} \quad (5.19.2.4-4)$$

where the coefficients  $LP_{ni}$ , and  $LP_{di}$  are given in the tables **txlpf1\_num\_coef**, and **txlpf1\_den\_coef** respectively in the C-code, and the coefficients  $HP_{ni}$ , and  $HP_{di}$  are given in the tables **txhpf1\_num\_coef**, and **txhpf1\_den\_coef** respectively in the C-code.

Two factors  $R_L$ , and  $R_H$  are computed as follows

$$R_L = 10 \log_{10} \left( \frac{\sum_{n=0}^{127} [r_L(n)]^2}{\sum_{n=0}^{127} [r_{4L}(n)]^2} \right) \quad (5.19.2.4-5)$$

and

$$R_H = 10 \log_{10} \left( \frac{\sum_{n=0}^{n=127} [r_H(n)]^2}{\sum_{n=0}^{n=127} [r_{4H}(n)]^2} \right) \quad (5.19.2.4-6)$$

A filter index  $F_{idx}$  is computed as follows from  $R_L$ , and  $R_H$ , and sent to the decoder.

$$F_{idx} = \begin{cases} 1 & ; R_L < -3 \\ 2 & ; R_L \geq -3 \text{ and } R_H < -3 \\ 0 & ; \textit{otherwise} \end{cases} \quad (5.19.2.4-7)$$

The shaping filter  $SF(z)$  is determined from  $F_{idx}$  as follows

$$SF(z) = \begin{cases} SF1(z) & ; F_{idx} = 0 \\ SF1(z)SF2(z) & ; F_{idx} = 1 \\ SF1(z)SF3(z) & ; F_{idx} = 2 \end{cases} \quad (5.19.2.4-8)$$

where,  $SF2(z)$ ,  $SF3(z)$  are

$$SF2(z) = \frac{SF2_{n0} + SF2_{n1}z^{-1} + \dots + SF2_{n12}z^{-10}}{1 + SF2_{d1}z^{-1} + \dots + SF2_{d12}z^{-10}} \quad (5.19.2.4-9)$$

$$SF3(z) = \frac{SF3_{n0} + SF3_{n1}z^{-1} + \dots + SF3_{n12}z^{-10}}{1 + SF3_{d1}z^{-1} + \dots + SF3_{d12}z^{-10}} \quad (5.19.2.4-10)$$

where the coefficients  $SF2_{ni}$ ,  $SF2_{di}$ ,  $SF3_{ni}$ , and  $SF3_{di}$  are given in the tables **shape2\_num\_coef**, **shape2\_den\_coef**, **shape3\_num\_coef**, and **shape3\_den\_coef** respectively in the C-code.

Finally,  $r_2(n)$  is filtered by  $SF(z)$  to provide the quantized formant residual  $r_q(n)$  for the Rate  $\frac{1}{2}$  unvoiced frame.

Note that a shaping filter index is determined and transmitted once every half-frame. Thus the procedure described in this section is repeated twice per frame.

## 5.20 Quantization of the Adaptive and Fixed-Codebook Gains

**Routine Name:** gain\_enc\_2, gain\_enc\_3, gain\_enc\_gaus

**Inputs:**

- $x(n)$ : The target signal
- $y(n)$ : The filtered adaptive codevector
- $c(n)$ : The algebraic codevector

1 •  $z(n)$ : The filtered algebraic codevector

2 •  $g_p$ : The adaptive codebook gain

3 **Outputs:**

4 •  $\hat{g}_p$ : The quantized adaptive codebook gain

5 •  $\hat{g}_c$ : The quantized algebraic codebook gain

6 **Initialization:**

7 • The quantized energy prediction errors  $\hat{R}(k)$  are set to  $-14$ .

8

9 In FR, Generic HR, and Voiced HR encoding types, the adaptive codebook gain (pitch gain) and the  
10 fixed (algebraic) codebook gain are vector quantized using a 128-element codebook. However, in  
11 Generic HR and Voiced HR, in every two subframes, either the lower or higher half of the codebook is  
12 used.

13  
14 The fixed-codebook gain quantization is performed using MA prediction with fixed coefficients. The 4<sup>th</sup>  
15 order MA prediction is performed on the innovation energy as follows. Let  $E(n)$  be the mean-removed  
16 innovation energy (in dB) at subframe  $n$ , and given by

$$17 \quad E(n) = 10 \log \left( \frac{1}{N} g_c^2 \sum_{i=0}^{N-1} c^2(i) \right) - \bar{E} \quad (5.20-1)$$

18

19 where  $N = 64$  is the subframe size,  $c(i)$  is the fixed-codebook excitation, and  $\bar{E} = 30$  dB is the mean  
20 of the innovation energy. Equation (5.20-1) can be expressed by

21

$$22 \quad E(n) = E_i + G_c - \bar{E} \quad (5.20-2)$$

23 where

$$24 \quad E_i = 10 \log \left( \frac{1}{N} \sum_{i=0}^{N-1} c^2(i) \right) \quad (5.20-3)$$

25

26 is the mean innovation energy in dB and

27

$$28 \quad G_c = 20 \log(g_c) \quad (5.20-4)$$

29

30 is the innovation gain in dB. The predicted energy is given by

$$31 \quad \tilde{E}(n) = \sum_{i=1}^4 b_i \hat{R}(n-i) \quad (5.20-5)$$

32 where  $[b_1 \ b_2 \ b_3 \ b_4] = [0.5, 0.4, 0.3, 0.2]$  are the MA prediction coefficients, and  $\hat{R}(k)$  is the quantized  
33 energy prediction error at subframe  $k$ . The predicted energy is used to compute a predicted fixed-  
34 codebook gain  $g'_c$  as in Equation (5.20-1) (by substituting  $E(n)$  by  $\tilde{E}(n)$  and  $g_c$  by  $g'_c$ ). This is done  
35 as follows. First, the mean innovation energy  $E_i$  is found as in Equation (5.20-3) and then the  
36 predicted gain  $G'_c$  in dB is found by

37

$$38 \quad G'_c = \tilde{E}(n) + \bar{E} - E_i \quad (5.20-6)$$

39

1 The prediction gain in the linear domain is given by

2

$$3 \quad g'_c = 10^{0.05G_c} = 10^{0.05(\tilde{E}(n)+\bar{E}-E_i)} \quad (5.20-7)$$

4 A correction factor between the gain  $g_c$  and the estimated one  $g'_c$  is given by

$$5 \quad \gamma = g_c / g'_c \quad (5.20-8)$$

6 Note that the prediction error is given by

$$7 \quad R(n) = E(n) - \tilde{E}(n) = 20 \log(\gamma) \quad (5.20-9)$$

8

9 The pitch gain,  $g_p$ , and correction factor  $\gamma$  are jointly vector quantized using a 7-bit codebook. The  
10 gain codebook search is performed by minimizing the mean-squared of the weighted error between  
11 original and reconstructed speech, which is given

$$12 \quad E = \mathbf{x}^t \mathbf{x} + g_p^2 \mathbf{y}^t \mathbf{y} + g_c^2 \mathbf{z}^t \mathbf{z} - 2g_p \mathbf{x}^t \mathbf{y} - 2g_c \mathbf{x}^t \mathbf{z} + 2g_p g_c \mathbf{y}^t \mathbf{z} \quad (5.20-10)$$

13

14 where the  $x$  is the target vector,  $y$  is the filtered adaptive codebook vector, and  $z$  is the filtered fixed-  
15 codebook vector. (Each gain vector in the codebook also has an element representing the quantized  
16 energy prediction error.) The quantized energy prediction error associated with the chosen gains is  
17 used to update  $\hat{R}(n)$ .

18

19 In the FR encoding types, 7 bits are used in every subframe; however, in the search stage, only the  
20 64 codevectors that are closest to the unquantized pitch gain,  $g_p$ , are taken into account. Since the  
21 two dimensional vectors in the codebook are ordered according to the pitch gain value, the initial  
22 index closest to the unquantized pitch gain is determined and the search is confined to the 64 vectors  
23 around that index.

24

25 In the Voiced HR and Generic HR cases, only half of the 7-bit codebook is used. Thus, 6 bits are  
26 needed in every subframe for encoding the index in the codebook half, but 1 bit is needed once every  
27 two subframes to indicate which codebook half is used. This gives a total of  $6 \times 4 + 2 = 26$  bits for  
28 encoding the gains in the 4 subframes.

29

30 To decide which codebook half is used in the first two subframes, an initial pitch gain is computed  
31 based on two subframes as

32

$$33 \quad g_i = \frac{\sum_{n=0}^{2N-1} x(n)y(n)}{\sum_{n=0}^{2N-1} y(n)y(n)} \quad (5.20-11)$$

34

35 This is similar to Equation (5.16.6-1) but with the summation performed over two subframes. The  
36 computation of the target signal  $x(n)$  and the filtered pitch codebook signal  $y(n)$  is also performed over  
37 a period of two subframes. Computing the target signal  $x(n)$  over a period longer than one subframe  
38 is performed by extending the computation of the weighted speech signal  $s_w(n)$  and the zero input  
39 response  $\mathbf{s}_0$  over a longer period while using the same LP filter in first subframe for all the extended  
40 period. The target signal  $x(n)$  is computed as the weighted speech signal  $s_w(n)$  after subtracting the  
41 zero-input response  $\mathbf{s}_0$  of the weighted synthesis filter  $W(z)/\hat{A}(z)$ . Similarly, computation of the  
42 filtered adaptive codebook signal  $y(n)$  is performed by extending the computation of the adaptive  
43 codebook vector  $v(n)$  and the impulse response  $h(n)$  of the weighted synthesis filter  $W(z)/\hat{A}(z)$  of  
44 the first subframe over a period longer than the subframe length. The filtered adaptive codebook  
45 signal is the convolution between the adaptive codebook vector  $v(n)$  and the impulse response  $h(n)$ ,  
46 where the convolution in this case is computed over two subframes.

The pitch gain value at index 64 in the codebook is 0.768606. In the first subframe, if the initial pitch gain calculated over two subframes is larger than or equal to 0.768606 then the upper half of the codebook is used in the first two subframes; otherwise the lower half is used. The same procedure described above is used in the gain quantization of the third and fourth subframes.

Furthermore, if pitch gain clipping is detected (as described above), the last 27 entries in the codebook are skipped in the quantization procedure since the pitch gain in these entries is higher than 1.

### 5.20.1 Gain Quantization in Unvoiced HR and Unvoiced QR

In Unvoiced HR and QR encoding types, the adaptive codebook is not used and only the innovation gain needs to be quantized. Gain prediction is done in the same manner as described above. First, a predicted gain  $G'_c$  is computed as described in Equation (5.20-6). Then the gain prediction error in subframe  $n$  is computed as

$$\Gamma = R(n) = 20 \log(g_c) - G'_c \quad (5.20.1-1)$$

Note that this is equivalent to Equation (5.20-8) but in dB. The gain prediction error in dB is uniformly quantized between  $\Gamma_{\min}$  and  $\Gamma_{\max}$  with step size given by

$$\delta = (\Gamma_{\min} - \Gamma_{\max}) / L \quad (5.20.1-2)$$

where  $L$  is the number of quantization levels. The quantization index  $k$  is given by the integer part of

$$\frac{\Gamma - \Gamma_{\min}}{\delta} + 0.5 \quad (5.20.1-3)$$

and the quantized prediction error is given by

$$\hat{\Gamma} = \hat{R}(n) = k \times \delta + \Gamma_{\min} \quad (5.20.1-4)$$

Finally the quantized gain is given by

$$\hat{g}_c = 10^{0.05(\hat{\Gamma} + G'_c)} \quad (5.20.1-5)$$

In Unvoiced HR, 6 bits are used to quantize the gain, thus  $L_v = 64$ , and the quantization boundaries are  $\Gamma_{\min} = -30$  and  $\Gamma_{\max} = 34$  (the quantization step is 1 dB).

In Unvoiced QR, 5 bits are used to quantized the gain, thus  $L_v = 32$ , and the quantization boundaries are  $\Gamma_{\min} = -22$  and  $\Gamma_{\max} = 20$  (the quantization step is also 1.3125 dB).

### 5.20.2 Quantization of the Adaptive and Fixed-Codebook Gains In RS-I Generic and Voiced FR

In RS-I Generic FR and RS-I Voiced FR, a three-dimensional gain codebook is used in which there are two dimensions for the adaptive codebook gains and one dimension for the fixed-codebook gain correction factor [33]. The codebook size is 128 (7 bits) for RS-I Generic FR and 256 (8 bits) for RS-I Voiced FR. The prediction performed on the fixed-codebook gain is the same as in RS-II described above. The two adaptive codebook gains are  $g_1$  and  $g_2$  representing gains for excitations  $u_1(n)$  and

1  $u_2(n)$ , respectively. The gain codebook search is performed by minimizing the mean square of the  
2 weighted error between original and reconstructed speech, which is given by  
3

$$4 \quad E = x^t x + g_1^2 y_1^t y_1 + g_2^2 y_2^t y_2 + g_c^2 z^t z - 2g_1 x^t y_1 - 2g_2 x^t y_2 - 2g_c x^t z + 2g_1 g_2 y_1^t y_2 + 2g_1 g_c y_1^t z + 2g_2 g_c y_2^t z$$

### 5 5.20.3 Quantization of the Adaptive and Fixed-Codebook Gains In RS-I Generic and Voiced 6 HR 7

8  
9 The adaptive codebook and fixed-codebook gains are quantized and transmitted in one out of two  
10 subframes, as described in section. The fixed-codebook gain prediction and joint gain quantization is  
11 the same as described above in the beginning of Section 5.20 (similar to what is used in RS-II  
12 Generic FR). The only difference is that a 6-bit gain codebook is used.

## 13 5.21 Memory Update

14  
15 An update of the states of the synthesis and weighting filters is needed in order to compute the target  
16 signal in the next subframe.

17 After the two gains have been quantized, the excitation signal,  $u(n)$ , in the present subframe is found  
18 by

$$19 \quad u(n) = \hat{g}_p v(n) + \hat{g}_c c(n), \quad n = 0, \dots, 63 \quad (5.21-1)$$

20  
21 where  $\hat{g}_p$  and  $\hat{g}_c$  are the quantized adaptive and fixed-codebook gains, respectively,  $v_i(n)$  the  
22 adaptive codebook vector (interpolated past excitation), and  $c(n)$  is the fixed-codebook vector  
23 (algebraic code including pre-filtering). The states of the filters can be updated by filtering the signal  
24  $r(n) - u(n)$  (difference between residual and excitation) through the filters  $1/\hat{A}(z)$  and  
25  $A(z/\gamma_1)H_{\text{de-emph}}(z)$  for the 64 sample subframe and saving the states of the filters. This would  
26 require 3 stages of filtering. A simpler approach, which requires only one filtering, is as follows. The  
27 local synthesis speech,  $\hat{s}(n)$ , is computed by filtering the excitation signal through  $1/\hat{A}(z)$ . The  
28 output of the filter due to the input  $r(n) - u(n)$  is equivalent to  $e(n) = s(n) - \hat{s}(n)$ . So the states of the  
29 synthesis filter  $1/\hat{A}(z)$  are given by  $e(n)$ ,  $n = 48, \dots, 63$ . Updating the states of the filter  
30  $A(z/\gamma_1)H_{\text{de-emph}}(z)$  can be done by filtering the error signal  $e(n)$  through this filter to find the  
31 perceptually weighted error  $e_w(n)$ . However, the signal  $e_w(n)$  can be equivalently found by  
32

$$33 \quad e_w(n) = x(n) - \hat{g}_p y(n) - \hat{g}_c z(n) \quad (5.21-2)$$

34  
35 Since the signals  $x(n)$ ,  $y(n)$ , and  $z(n)$  are available, the states of the weighting filter are updated by  
36 computing  $e_w(n)$  as in Equation (5.21-2) for  $n = 48, \dots, 63$ . This saves two stages of filtering.  
37

### 38 5.21.1 Memory Update in RS-I Voiced FR and Generic FR

39  
40 In RS-I Voiced FR and RS-I Generic FR, since three gain parameters have been quantized, the  
41 excitation signal,  $u(n)$ , in the present subframe is found by  
42

$$43 \quad u(n) = \hat{g}_1 v_1(n) + \hat{g}_2 v_2(n) + \hat{g}_c c(n), \quad n = 0, \dots, 63, \quad (5.21.1-1)$$

44 where  $\hat{g}_1$  and  $\hat{g}_2$  are the two adaptive codebook gains. Now the signal  $e_w(n)$  is found by

1

2

$$e_w(n) = x'(n) - (\hat{g}_1 y_1'(n) + \hat{g}_2 y_2'(n) + \hat{g}_c z'(n)) \quad (5.21.1-2)$$

3

4 The states of the spectral weighting filter are updated by computing  $e_w(n)$  as in Equation (5.21.1-1) for  
 5  $n = 30, \dots, 39$ . The signals  $x'(n)$ ,  $y_1'(n)$ ,  $y_2'(n)$ , and  $z'(n)$  used in Equation (5.21.1-1) are non-harmonic  
 6 noise weighted or spectrally weighted signals. The state of the harmonic noise weighting signal is  
 7 comprised of past values of  $e_w(n)$ . The size of the state of the filter  $C(z)$  is equal to maximum pitch lag  
 8 plus the number of taps used for interpolation. The state is updated by discarding the first 64  
 9 samples of the state and appending the state with  $e_w(n)$  for  $n=0, \dots, 63$ .

## 10 5.22 Supplementary Information for Frame Error Concealment in RS-II 11 Generic FR and RS-I Voiced FR

12

13 **Routine Name:** `fer_encode`

14 **Inputs:**

- 15 •  $s(n)$ : The speech signal
- 16 •  $\hat{s}(n)$ : The synthesized speech signal
- 17 •  $s_w(n)$ : The weighted speech signal
- 18 •  $\hat{s}_w(n)$ : The weighted synthesis signal
- 19 • Encoding type
- 20 •  $\bar{d}_k$ : The signal modification pitch delay parameter at the frame boundary
- 21 •  $d_0$ ,  $d_1$ , and  $d_2$ : The pitch lags in each half-frame
- 22 •  $C_{norm}(d)$ : The normalized correlation at pitch lags  $d_1$  and  $d_2$
- 23 • The local VAD flag
- 24 •  $e_{tilt}(l)$ : Spectral tilt
- 25 • The closed-loop pitch lags
- 26 •  $E_{rel}$ : Relative frame energy
- 27 • Classification decision of previous frame

28 **Outputs:**

- 29 • Frame classification decision
- 30 •  $E$ : The synthesized speech energy
- 31 •  $T_0$ : First glottal pulse position with respect to frame beginning

32 **Initialization:**

- 33 • None

34

35 In the Generic FR encoding type, 14 bits are used to send supplementary information, which  
 36 improves frame erasure concealment and the convergence and recovery of the decoder after erased  
 37 frames. These parameters include energy information (6 bits), signal classification information (2

bits), and phase information (the estimated position of the first glottal pulse in a frame) (6 bits). In the next sections, computation and quantization of these additional parameters will be described in detail.

### 5.22.1 Signal Classification for Frame Error Concealment and Recovery

The basic idea behind using a classification of the speech for a signal reconstruction in the presence of erased frames consists of the fact that the ideal concealment strategy is different for quasi-stationary speech segments and for speech segments with rapidly changing characteristics. While the best processing of erased frames in non-stationary speech segments can be summarized as a rapid convergence of speech-encoding parameters to the ambient noise characteristics, in the case of quasi-stationary signal, the speech-encoding parameters do not vary significantly and can be kept practically unchanged during several adjacent erased frames before being damped. Also, the optimal method for a signal recovery following an erased block of frames varies with the classification of the speech signal.

The speech signal can be roughly classified as voiced, unvoiced and pauses. Voiced speech contains an important amount of periodic components and can be further divided in the following categories: voiced onsets, voiced segments, voiced transitions and voiced offsets. A voiced onset is defined as a beginning of a voiced speech segment after a pause or an unvoiced segment. During voiced segments, the speech signal parameters (spectral envelope, pitch period, ratio of periodic and non-periodic components, energy) vary slowly from frame to frame. A voiced transition is characterized by rapid variations of a voiced speech, such as a transition between vowels. Voiced offsets are characterized by a gradual decrease of energy and voicing at the end of voiced segments.

The unvoiced parts of the signal are characterized by missing the periodic component and can be further divided into unstable frames, where the energy and the spectrum changes rapidly, and stable frames where these characteristics remain relatively stable. Remaining frames are classified as silence. Silence frames comprise all frames without active speech, i.e. also noise-only frames if a background noise is present.

Not all of the above mentioned classes need a separate processing. Hence, for the purposes of error concealment techniques, some of the signal classes are grouped together.

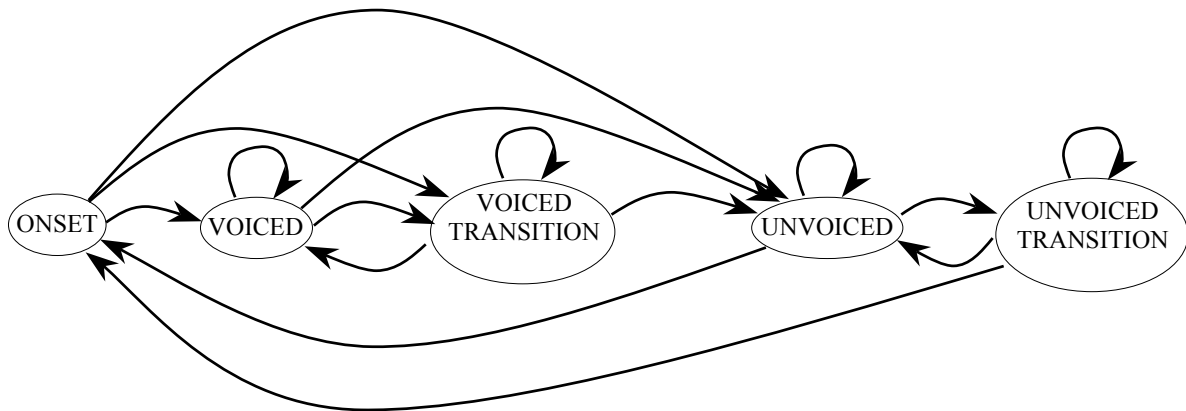
In determining the supplementary parameters, the lookahead at the encoder is used. The lookahead allows estimate of the evolution of the signal in the following frame and consequently the classification can be done by taking into account the future signal behavior.

The frame classification is done with the consideration of the concealment and recovery strategy. In other words, any frame is classified in such a way that the concealment can be optimal if the following frame is missing, or that the recovery can be optimal if the previous frame was lost. Some of the classes used for the FER processing need not be transmitted, as they can be deduced without ambiguity at the decoder. Five distinct classes are used, and defined as follows:

- UNVOICED class comprises all unvoiced speech frames and all frames without active speech. A voiced offset frame can be also classified as UNVOICED if its end tends to be unvoiced and the concealment designed for unvoiced frames can be used for the following frame in case it is lost.
- UNVOICED TRANSITION class comprises unvoiced frames with a possible voiced onset at the end. The onset is however still too short or not built well enough to use the concealment designed for voiced frames. The UNVOICED TRANSITION class can follow only a frame classified as UNVOICED or UNVOICED TRANSITION.
- VOICED TRANSITION class comprises voiced frames with relatively weak voiced characteristics. Those are typically voiced frames with rapidly changing characteristics (transitions between vowels) or voiced offsets lasting the whole frame. The VOICED TRANSITION class can follow only a frame classified as VOICED TRANSITION, VOICED or ONSET.

- VOICED class comprises voiced frames with stable characteristics. This class can follow only a frame classified as VOICED TRANSITION, VOICED or ONSET.
- ONSET class comprises all voiced frames with stable characteristics following a frame classified as UNVOICED or UNVOICED TRANSITION. Frames classified as ONSET correspond to voiced onset frames where the onset is already sufficiently well built for the use of the concealment designed for lost voiced frames. The concealment techniques used for a frame erasure following the ONSET class are the same as following the VOICED class. The difference is in the recovery strategy. If an ONSET class frame is lost (i.e. a VOICED good frame arrives after an erasure, but the last good frame before the erasure was UNVOICED), a special technique can be used to artificially reconstruct the lost onset. The artificial onset reconstruction techniques will be described in more detail in the decoder description. On the other hand if a good ONSET frame arrives after an erasure and the last good frame before the erasure was UNVOICED, this special processing is not needed, as the onset has not been lost (has not been in the lost frame).

The classification state diagram is outlined in Figure 5.22-1. The classification information is transmitted using 2 bits. As it can be seen from Figure 5.22-1, UNVOICED TRANSITION class and VOICED TRANSITION class can be grouped together as they can be unambiguously differentiated at the decoder (UNVOICED TRANSITION can follow only UNVOICED or UNVOICED TRANSITION frames, VOICED TRANSITION can follow only ONSET, VOICED or VOICED TRANSITION frames).



**Figure 5.22-1: Block diagram of a frame classification state machine for the erasure concealment.**

The following parameters are used for the classification: a normalized correlation  $\bar{R}'_{xy}$ , a spectral tilt measure  $e'_t$ , a signal to noise ratio  $snr$ , a pitch stability counter  $pc$ , a relative frame energy of the signal at the end of the current frame  $E_{rel}$  and a zero-crossing counter  $zc$ . As can be seen in the following detailed analysis, the computation of these parameters uses the available look-ahead as much as possible to take into account the behavior of the speech signal also in the following frame.

The average normalized correlation  $\bar{R}'_{xy}$  is computed as part of the open-loop pitch search module. It consists of averaging the normalized correlations of the second half-frame and the lookahead. That is

$$\bar{R}'_{xy} = 0.5(C_{norm}(d_1) + C_{norm}(d_2)). \quad (5.22.1-1)$$

The spectral tilt parameter  $e'_t$  contains the information about the frequency distribution of energy. As described above, the spectral tilt for one spectral analysis is estimated as a ratio between the energy concentrated in low frequencies and the energy concentrated in high frequencies. Here, the tilt

measure used is the average in the logarithmic domain of the spectral tilt measures  $e_{iilt}(0)$  and  $e_{iilt}(1)$  defined in Equation (5.10.2-6). That is,

$$e'_t = 10 \log(e_{iilt}(0)e_{iilt}(1)) \quad (5.22.1-2)$$

The signal to noise ratio (SNR) measure exploits the fact that for a general waveform matching encoder, the SNR is much higher for voiced sounds. The  $snr$  parameter estimation must be done at the end of the encoder subframe loop and is computed using the relation

$$snr = \frac{E_{sw}}{E_e} \quad (5.22.1-3)$$

where  $E_{sw}$  is the energy of the weighted speech signal  $s_w(n)$  of the current frame and  $E_e$  is the energy of the error between this weighted speech signal and the weighted synthesis signal of the current frame.

The pitch stability counter  $pc$  assesses the variation of the pitch period. It is computed as follows:

$$pc = |d_1 - d_0| + |d_2 - d_1| \quad (5.22.1-4)$$

The values  $d_0$ ,  $d_1$ , and  $d_2$  correspond to the open-loop pitch estimates from the first half of the current frame, the second half of the current frame and the lookahead, respectively.

The last parameter is the zero-crossing parameter  $zc$  computed on a 20 ms segment of the speech signal. The segment starts in the middle of the current frame and uses two subframes of the lookahead. Here, the zero-crossing counter  $zc$  counts the number of times the signal sign changes from positive to negative during that interval.

To make the classification more robust, the classification parameters are considered together forming a function of merit  $fm$ . For that purpose, the classification parameters are first scaled between 0 and 1 so that each parameter's value typical for unvoiced signal translates in 0 and each parameter's value typical for voiced signal translates into 1. A linear function is used between them. The scaled version  $p^s$  of a certain parameter  $p_x$  is obtained using

$$p^s = k_p p_x + c_p \quad \text{constrained by } 0 \leq p^s \leq 1 \quad (5.22.1-5)$$

The function coefficients  $k_p$  and  $c_p$  have been found experimentally for each of the parameters so that the signal distortion due to the concealment and recovery techniques used in presence of frame errors is minimal. The values used are summarized in Table 5.22-1.

**Table 5.22-1: Signal Classification Parameters and the coefficients of their respective scaling functions**

Parameter	Meaning	$k_p$	$c_p$
$\bar{R}'_{xy}$	Normalized Correlation	2.857	-1.286
$e'_t$	Spectral Tilt	0.04167	0
$snr$	Signal to Noise Ratio	0.1111	-0.3333
$pc$	Pitch Stability counter	-0.07143	1.857
$E_{rel}$	Relative Frame Energy	0.05	0.45
$zc$	Zero Crossing Counter	-0.04	2.4

The merit function has been defined as

$$f = \frac{1}{7}(2\bar{R}_{xy}^s + e_t^s + snr^s + pc^s + E_{rel}^s + zc_s) \quad (5.22.1-6)$$

where the superscript s indicates the scaled version of the parameters.

A first classification decision is made for UNVOICED classes as follows:

*If (local\_VAD=0) OR ( $E_{rel} < -8$ ) then class = UNVOICED.*

If the above condition is not satisfied, then the classification proceeds using the merit function  $f_m$  and following the rules summarized in Table 5.22-2.

**Table 5.22-2: Signal Classification Rules at the Encoder**

Previous Frame Class	Rule	Current Frame Class
ONSET	$f_m \geq 0.66$	VOICED
VOICED	$0.66 > f_m \geq 0.49$	VOICED TRANSITION
VOICED TRANSITION	$f_m < 0.49$	UNVOICED
UNVOICED TRANSITION	$f_m > 0.63$	ONSET
UNVOICED	$0.63 \geq f_m > 0.585$	UNVOICED TRANSITION
	$f_m \leq 0.585$	UNVOICED

The class information is encoded with two bits as explained above. Despite the fact that the supplementary information, which improves frame erasure concealment, is transmitted only in Generic FR frames, the classification is done for each frame. This is needed to maintain the classification state machine up to date as it uses the information about the previous frame class. The classification is however straightforward for encoding types dedicated to unvoiced or voiced frames. Hence, a VOICED HR frame is always classified as voiced frame and UNVOICED HR and QR frames are always classified as unvoiced frames.

### 5.22.2 Other Speech Parameters for Frame Error Processing

In addition to the signal classification information, the other transmitted parameters are energy information and phase information. A precise control of the speech energy is very important in frame error concealment. The importance of the energy control becomes more evident when a normal operation is resumed after an erased block of frames. Since all encoding schemes of active speech make use of a prediction, the actual energy cannot be properly estimated at the decoder. In voiced speech segments, the incorrect energy can persist for several consecutive frames, which can be very annoying especially when this incorrect-valued energy increases.

Although the energy control is very important for voiced speech because of the long-term prediction (pitch prediction), it is also important for unvoiced speech. The reason is the prediction of the innovation gain quantizer. An incorrect value of energy during unvoiced segments can cause an annoying high frequency fluctuation. The phase control can be done in several ways, mainly depending on the available bandwidth. Here, a simple phase control is achieved during lost voiced onsets by searching the approximate information about the glottal pulse position. Hence, apart from the signal classification information discussed in the previous section, the most important information to send is the information about the signal energy and the position of the first glottal pulse in a frame (phase information).

#### 5.22.2.1 Energy Information

The energy information is estimated and sent in the speech signal domain using 6 bits. The energy information is the maximum of the signal energy for frames classified as VOICED or ONSET, or the

1 average energy per sample for other frames. For VOICED or ONSET frames, the maximum signal  
2 energy is computed pitch synchronously at the end of the frame as follow:  
3

$$4 \quad E = \max(\hat{s}^2(i)) \quad i = L - t_E, \dots, L - 1 \quad (5.22.2.1-1)$$

5  
6 where  $L=256$  is the frame length and signal  $\hat{s}(i)$  is the local synthesis signal. If the pitch delay is  
7 greater than 63 samples,  $t_E$  equals the rounded closed-loop pitch lag of the last subframe. If the pitch  
8 delay is less than 64 samples, then  $t_E$  is set to twice the rounded close-loop pitch lag of the last  
9 subframe.

10  
11 For other classes,  $E$  is the average energy per sample of the second half of the current frame, i.e.  $t_E$   
12 is set to  $L/2$  and the  $E$  is computed as:  
13

$$14 \quad E = \frac{1}{t_E} \sum_{i=L-t_E}^{L-1} \hat{s}^2(i) \quad (5.22.2.1-2)$$

15  
16 The energy information is quantized using a 6 bit uniform quantizer in the range of 0 dB to 96 dB with  
17 a step of 1.55 dB. The quantization index is given by the integer part of  
18

$$19 \quad i = \frac{10 \log(E + 0.001)}{1.55} \quad (5.22.2.1-3)$$

20  
21 The index  $i$  is then limited to the range  $0 \leq i \leq 63$  and  $i \neq 62$ . Thus for values of  $i < 0$ ,  $i$  is set to 0, and for  
22 values  $i \geq 62$ ,  $i$  is set to 63. The index pattern  $i=62$  (binary '111110') is reserved for the identification of  
23 the interoperable encoding types.

### 24 **5.22.2.2 Phase Control Information**

25  
26 The phase control is particularly important while recovering after a lost segment of voiced speech for  
27 similar reasons as described in the previous section. After a block of erased frames, the decoder  
28 memories become unsynchronized with the encoder memories. Sending some phase information  
29 helps in re-synchronizing the decoder. The rough position of the first glottal pulse in the frame is sent.  
30 This information is then used for the recovery after lost voiced onsets as will be described later.

31  
32 Let  $T_{sf1}$  be the rounded closed-loop pitch lag for the first subframe. The position of the first glottal  
33 pulse is searched among the  $T_{sf1}$  first samples of the frame by looking for the sample with the  
34 maximum amplitude. Best results are obtained when the position of the first glottal pulse is measured  
35 on the low-pass filtered residual signal. A simple FIR low-pass filter with coefficients 0.25, 0.5 and  
36 0.25 is used.  
37

38 The position of the first glottal pulse  $\tau$  is encoded using 6 bits in the following manner. The precision  
39 used to encode the position of the first glottal pulse depends on the closed-loop pitch value for the  
40 first subframe  $T_{sf1}$ . This is possible because this value is known both by the encoder and the decoder,  
41 and is not subject to error propagation after one or several frame losses. When  $T_{sf1}$  is less than 64,  
42 the position of the first glottal pulse relative to the beginning of the frame is encoded directly with a  
43 precision of one sample. When  $64 \leq T_{sf1} < 128$ , the position of the first glottal pulse relative to the  
44 beginning of the frame is encoded with a precision of two samples by using a simple integer division,  
45 i.e.,  $\tau/2$ . When  $T_{sf1} \geq 128$ , the position of the first glottal pulse relative to the beginning of the frame is  
46 encoded with a precision of four samples by further dividing  $\tau$  by 2. The inverse procedure is done at  
47 the decoder. If  $T_{sf1} < 64$ , the received quantized position is used as is. If  $64 \leq T_{sf1} < 128$ , the received  
48 quantized position is multiplied by 2 and incremented by 1. If  $T_{sf1} \geq 128$ , the received quantized  
49 position is multiplied by 4 and incremented by 2 (incrementing by 2 results in uniformly distributed  
50 quantization error).

### 5.22.3 Parameters for Frame Error Processing in RS-I

In Rate-Set I operation, information is transmitted only in RS-I Voiced FR. Only the energy information is transmitted using 6 bits as described in Section 5.22.2.1. The classification is performed at the decoder as will be described later.

## 5.23 Encoding of Inactive Speech Frames (CNG-ER and CNG-QR)

**Routine Name:** CNG\_enc, CNG\_synthesis

**Inputs:**

- $E(16)$ : LP residual energy
- $\bar{N}_f$ : Long-term average noise energy
- $\hat{q}_i$ : The quantized immitance spectral pairs for current frame

**Outputs:**

- $\hat{s}(n)$ : The synthesized speech signal
- $\bar{q}_i$ : The smoothed immitance spectral pairs
- $\bar{E}_s$ : The smoothed excitation energy

**Initialization:**

- Initially,  $\bar{E}_s$  is set to  $\hat{E}_s$  if not stated otherwise in the following section.  $\bar{q}_i$  are initially set to  $\hat{q}_i$ . The seed value of the excitation random generator is initially set to 21845. Otherwise, buffers and filter memories are set to zero at initialization.

In VMR-WB modes 0, 1, and 2, inactive speech frames are encoded using CNG-ER encoding type. Comfort noise generation (CNG) is used at the decoder to regenerate inactive speech frames. This encoding type is referred to as CNG-ER. In VMR-WB mode 3, due to the interoperability with AMR-WB, encoding of inactive speech frames is performed using the same quantization procedure as in AMR-WB. This requires 35 bits/frame, which does not fit into an ER frame. Note that the last CNG bit is not used in VMR-WB and it is set to 0. Thus a Quarter-Rate frame is used. Encoding of inactive speech frames in VMR-WB mode 3 is referred to as CNG-QR encoding type. In either case, the background noise encoding parameters consist of spectral shape and energy information (LP parameters and excitation energy). At the decoder, CNG is performed by generating a random excitation scaled by a proper gain to drive the LP synthesis filter. The synthesis model uses smoothed LP filter and gain parameters.

### 5.23.1 LP Parameter Quantization in CNG-ER and CNG-QR

The mean-removed ISF vector at frame  $n$ ,  $\mathbf{z}(n)$ , is directly quantized without prediction using single-stage split-VQ. In case of CNG-ER, the vector is split into 3 subvectors of dimension 3, 5, and 8, and quantized with 5, 5, and 4 bits, respectively. In the case of CNG-QR, the AMR-WB SID\_UPDATE frame quantizer is used whereby the ISF vector is split into 5 subvectors of dimension 2, 3, 3, 4, and 4, and quantized with 6, 6, 6, 5 and 5 bits, respectively. This is shown in Table 5.23-1 and Table 5.23-2.

**Table 5.23-1: Quantization of ISF vector for CNG-ER**

Unquantized 16-element-long ISF vector		
Split 1 (0-2) 5 bits	Split 2 (3-7) 5 bits	Split 3 (8-15) 4 bits

1 **Table 5.23-2: Quantization of ISF vector for CNG-QR**

Unquantized 16-element-long ISF vector				
Split 1 (0-1) 6 bits	Split 2 (2-4) 6 bits	Split 3 (5-7) 6 bits	Split 4 (8-11) 5 bits	Split 4 (12-15) 5 bits

2  
3 The cdma2000 system does not allow generation of all-zeros and all-ones frames. To comply with  
4 this requirement in CNG-ER encoding type, all-zero and all-one patterns for the three subvectors are  
5 prevented. If the quantizer chooses an all-one index for the first two subvectors (index 31), then in the  
6 quantization of the third subvector, the last index is not permitted. Similarly, if the quantizer chooses  
7 an all-zero index for the first two subvectors (index 0), then in the quantization of the third subvector,  
8 the first two indices (indices 0 and 1) are not permitted. Index 0 is not permitted to prevent all zero  
9 frames, and index 1 is not permitted since the pattern '00000000000001' is reserved to signal  
10 NO\_DATA frames. This is useful for interoperability between AMR-WB and VMR-WB when AMR-WB  
11 is operating in DTX mode (see Section 7).

12  
13 After quantization in the frequency domain, the ISF parameters are converted into the cosine domain  
14 to obtain the ISP vector  $\hat{\mathbf{q}}$ .

### 15 5.23.2 Energy Quantization in CNG-ER and CNG-QR

16  
17 The energy per sample is computed from the LP residual energy. The LP residual energy is a  
18 byproduct of the LP analysis procedure and it is given by the value  $E(16)$  in the Levinson-Durbin  
19 recursion of Equation (5.6.2-2). The energy per sample is computed as

$$20 \quad E_s = 0.0059322 E(16) \quad (5.23.2-1)$$

21  
22 where the scaling of  $E(16)$  takes into account the LP analysis window size and shape. In case of the  
23 first inactive speech frame after an active speech frame and very low background noise ( $\bar{N}_f < 16$ ),  $E_s$   
24 is multiplied by 0.1 to prevent smearing of the respiration noise. The energy per sample is then  
25 converted to the  $\log_2$  domain for quantization purposes as

$$26 \quad E_{s,2} = \log_{10}(E_s) / \log_{10}(2) \quad (5.23.2-2)$$

27  
28 For narrowband signals,  $E_{s,2}$  is increased by 0.5 to compensate for limited bandwidth of the input  
29 signal. The computed energy in the  $\log_2$  domain is then offset by a certain negative value  $E_{offset}$  in  
30 the range  $[E_{offset_{min}}, 0]$  where  $E_{offset_{min}} = -3$  for narrowband inputs and  $E_{offset_{min}} = -2.2032$  for  
31 wideband inputs (corresponding approximately to -9 dB and -6.6 dB, respectively).

32  
33 For the first inactive frame after an active speech frame, the offset value is computed as

$$34 \quad E_{offset} = 0.1102\bar{N}_f - 3.8556 \text{ constrained by } E_{offset_{min}} \leq E_{offset} \leq 0 \quad (5.23.2-3)$$

35  
36 where  $\bar{N}_f$  is the long-term average noise energy given in Equation (5.4.3-2). In subsequent inactive  
37 speech frames,  $E_{offset}$  converges to  $E_{offset_{min}}$  using the relation

$$E_{offset} = 0.9E_{offset} + 0.1E_{offset_{min}} \quad (5.23.2-4)$$

The offset energy in the  $\log_2$  domain is given by

$$E_{s,2} = E_{s,2} + E_{offset} \quad (5.23.2-5)$$

and it is quantized using 6 bits. A uniform quantizer is used with level 0 at -2 and level 63 at 22, and with quantization step 24/63. The quantization index is found using the relation

$$index = \left\lfloor \left( (E_{s,2} + 2) \frac{63}{24} \right) \right\rfloor \quad (5.23.2-6)$$

and the quantized value is found by

$$\hat{E}_{s,2} = \frac{24}{63} index - 2 \quad (5.23.2-7)$$

The quantized energy per sample in the linear domain is then found as

$$\hat{E}_s = 2^{\hat{E}_{s,2}} \quad (5.23.2-8)$$

### 5.23.3 Local CNG Synthesis

Local CNG synthesis is performed at the encoder in order to update the filters and adaptive codebook memories. CNG is performed by generating 20 ms random values, scaling by a gain computed from the smoothed quantized energy, and filtering the scaled excitation through a smoothed LP synthesis filter. Random short integer values are generated using the relation

$$seed = short(seed \times 31821 + 13849) \quad (5.23.3-1)$$

with the seed value initially set to 21845. The normalized random excitation sequence (with energy per sample equal to 1) is computed from the random integer sequence  $r_n(n)$  for the 256-sample frame using the relation

$$c(n) = r_n(n) \sqrt{256 / \sum_{i=0}^{255} r_n(i)} \quad (5.23.3-2)$$

For the first inactive speech frame after an active speech frame, the smoothed energy used for synthesis is updated as

$$\bar{E}_s = 0.5\bar{E}_s + 0.5\hat{E}_s \quad (5.23.3-3)$$

with the exception of incoming NO\_DATA frames from AMR-WB encoder in the AMR-WB interoperable mode (Section 7). The first NO\_DATA frame is usually preceded by an inactive speech frame encoded at Full-Rate (i.e., AMR-WB VAD bit set to 0). In this case, the mean energy of the adaptive codebook memory is used for the update of  $\bar{E}_s$  with the weighting of 0.2 and 0.8 for the mean energy of the adaptive codebook and  $\bar{E}_s$  respectively. If the AMR-WB NO\_DATA frame is not

1 preceded by an inactive Full-Rate frame, no  $\bar{E}_s$  update is performed. For consequent frames, the  
 2 smoothed energy is updated as

$$3 \quad \bar{E}_s = 0.7\bar{E}_s + 0.3\hat{E}_s \quad (5.23.3-4)$$

5  
 6 Initially,  $\bar{E}_s$  is set equal to  $\hat{E}_s$ . In the AMR-WB interoperable mode (VMR-WB mode 3),  $\bar{E}_s$  is initially  
 7 set to the mean energy of the adaptive codebook memory in case of a NO\_DATA frame. The energy  
 8 scaled excitation signal is given by

$$9 \quad u(n) = \sqrt{\bar{E}_s} c(n), n=0, \dots, 255 \quad (5.23.3-5)$$

11 The smoothed ISP vectors are updated as

$$12 \quad \bar{\mathbf{q}} = 0.9\bar{\mathbf{q}} + 0.1\hat{\mathbf{q}} \quad (5.23.3-6)$$

13  
 14 where initially  $\bar{\mathbf{q}}$  is set equal to  $\hat{\mathbf{q}}$ . The processing is once again different for the first inactive ER or  
 15 QR speech frame after an active speech. If such a frame is a NO\_DATA frame preceded by an  
 16 inactive Full-Rate frame, the update procedure described above is used with weighting of 0.8 and 0.2.  
 17 For other first ER or QR inactive frames  $\bar{\mathbf{q}}$  is not updated. The smoothed ISP parameter vector is  
 18 converted to the LP coefficient domain to obtain the LP synthesis filter. The comfort noise is  
 19 generated by filtering the scaled excitation  $u(n)$  through the smoothed synthesis filter.  
 20  
 21

#### 22 **5.23.4 Memory Update in CNG-ER and CNG-QR**

23  
 24 After synthesizing inactive speech frames using CNG as described in the previous section, the  
 25 memory of the synthesis filter and adaptive codebook are updated. In the AMR-WB interoperable  
 26 mode, the adaptive codebook is reset. The buffers used in signal modification are also updated. The  
 27 previous frame ISPs  $\hat{\mathbf{q}}_4^{(n-1)}$  are updated with  $\bar{\mathbf{q}}$ . Other memory parameters are reset to their initial  
 28 states (e.g., ISP and gain quantizers, weighting filter, etc.).  
 29  
 30

31 Note that encoding of CNG-ER and CNG-QR types is performed directly after the rate selection  
 32 described in Section 5.10. If either CNG-ER or CNG-QR encoding type is chosen then the processing  
 33 described in this section is performed and no further processing related to the other encoding types is  
 34 further performed.  
 35

#### 36 **5.23.5 Encoding of Inactive Speech Frames in RS-I CNG ER**

37  
 38 The CNG operation is similar to RS-II described above. The only difference is that the LP parameters  
 39 are quantized using 11 bits and the energy is quantized with 5 bits. The LP parameters are quantized  
 40 in the ISF domain using the same codebooks as in RS-I CNG ER. However, only the first half of each  
 41 codebook is used. This will give 4, 4, and 3 bits for the 3 splits as shown in Table 5.23-3.  
 42

43 **Table 5.23-3: Quantization of ISF vector for RS-I CNG-ER**

Unquantized 16-element-long ISF vector		
Split 1 (0-2) 4 bits	Split 2 (3-7) 4 bits	Split 3 (8-15) 3 bits

44 The energy quantization is also similar to RS-II described above in Section 5.23.2. However, only 5  
 45 bits are used. A uniform quantizer is used with level 0 at -2 and level 31 at 22, and with quantization  
 46 step 24/31. The quantization index is found using the relation  
 47

$$1 \quad index = \left\lfloor \left( (E_{s,2} + 2) \frac{31}{24} \right) \right\rfloor$$

2

3 The index is then limited not to increase by more than one from the value of the previous frame if the  
 4 previous frame was also a CNG frame. If the previous frame was an active speech frame, the index is  
 5 not allowed to increase by more than 3 from the index of the last CNG frame. This measure provides  
 6 more protection against frames with corrupted rate information in RS-I.

7

8 The quantized value is found by

9

$$10 \quad \hat{E}_{s,2} = \frac{24}{31} index - 2$$

11 The quantized energy per sample in the linear domain is then found as  $\hat{E}_s = 2^{\hat{E}_{s,2}}$

12

1

## 2 6 FUNCTIONAL DESCRIPTION OF THE DECODER

3

4 The function of the decoder consists of decoding the transmitted parameters (LP parameters,  
5 adaptive codebook vector, adaptive codebook gain, fixed-codebook vector, fixed-codebook gain) and  
6 performing synthesis to obtain the reconstructed speech. The signal flow at the decoder is shown in  
7 Figure 4.1-3.

8

9 The decoding process starts with decoding the LP filter parameters. The received indices of ISP  
10 quantization are used to reconstruct the quantized ISP vector. The interpolation described in Section  
11 5.6.5 is performed to obtain 4 interpolated ISP vectors (corresponding to 4 subframes). Then the  
12 excitation signal is reconstructed and post-processed before performing LP synthesis filtering to  
13 obtain the reconstructed speech. The reconstructed speech is then de-emphasized (inverse of pre-  
14 emphasis applied at the encoder). Finally, a post-processing is applied for enhancing the periodicity  
15 in the low frequency region of the signal, and then the signal is up-sampled to 16 kHz. Finally high-  
16 band signal is generated to the frequency band from 6 to 7 kHz. Both parts of the signal are added to  
17 obtain the full-band reconstructed speech.

18

19 More details about the excitation reconstruction and excitation post-processing, synthesis and post-  
20 processing of synthesis signal, and high frequency generation will be provided in the following  
21 sections.

### 22 6.1 Reconstruction of the Excitation

23

24 The decoding process is performed in the following order: For each subframe, the interpolated ISP  
25 vector is converted to LP filter coefficient domain  $a_k$ , which is used for synthesizing the reconstructed  
26 speech in the subframe.

27

28 The following steps are repeated for each subframe:

29

30 1) **Decoding of the adaptive codebook vector:** In case of FR and Generic HR encoding  
31 types, the received pitch index (adaptive codebook index) is used to find the integer and  
32 fractional parts of the pitch lag. The initial adaptive codebook excitation vector  $v'(n)$  is found  
33 by interpolating the past excitation  $u(n)$  (at the pitch delay) using the FIR filter described in  
34 Section 5.16. In case of Generic HR no low-pass filtering is used and the adaptive codebook  
35 excitation is  $v(n) = v'(n)$ . In case of FR, the received adaptive filter index is used to decide  
36 whether the filtered adaptive codebook is  $v(n) = v'(n)$  or  $v(n) = 0.18v'(n) + 0.64v'(n-1) +$   
37  $0.18v'(n-2)$ . In case of Voiced HR, the delay contour is first computed for the whole frame  
38 as described in Equation (5.11.2-1). The initial adaptive codebook excitation  $v'(n)$  in certain  
39 subframes is computed by interpolating the past excitation in the adaptive codebook buffer at  
40 the delays given by the delay contour as described in Section 5.16.4. The received adaptive  
41 filter index is then used to find out whether the filtered adaptive codebook is  $v(n) = v'(n)$  or  
42  $v(n) = 0.18v'(n) + 0.64v'(n-1) + 0.18v'(n-2)$ . In case of Unvoiced encoding types and CNG,  
there is no adaptive codebook contribution.

43

In RS-I Generic FR and RS-I Voiced FR, the adaptive codebook vector is decomposed into  
44 two constituents as described in Section 5.16.9.

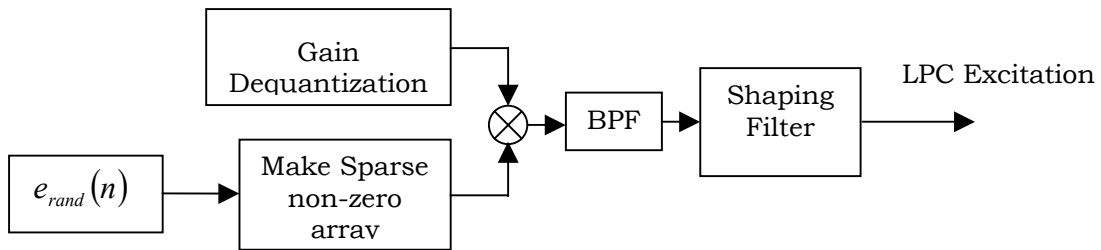
45

46 2) **Decoding of the innovative vector:** In case of FR, Voiced HR, and Generic HR, the  
47 received algebraic codebook index is used to extract the positions and amplitudes (signs) of  
48 the excitation pulses and to find the algebraic codevector  $c(n)$ . If the integer part of the pitch  
49 lag is less than the subframe size 64, the pitch sharpening procedure is applied, which  
50 translates into modifying  $c(n)$  by filtering it through the adaptive pre-filter  $F(z) = (1 - \beta_1 z^{-1}) / (1 -$   
51  $0.85z^{-T})$  which further consists of two parts: a periodicity enhancement part  $1/(1-0.85z^{-T})$ ,  
52 where  $T$  is the integer part of the pitch lag representing the fine spectral structure of the  
53 speech signal, and a tilt part  $(1 - \beta_1 z^{-1})$ , where  $\beta_1$  is related to the voicing of the previous

subframe and is bounded by [0.0,0.5]. The periodicity enhancement part of the filter colors the spectrum by damping inter-harmonic frequencies, which are annoying to the human ear in case of voiced signals. In case of FR and Generic HR encoding types,  $T$  is the integer part of the pitch lag. In case of Voiced HR,  $T$  is computed on a sample basis following the pitch contour.

In case of Unvoiced HR, the signs and indices of the two random vectors are decoded and the excitation is reconstructed as in Equation (5.18.1-1). In case of Unvoiced QR, the excitation is reconstructed as in Equation (5.18.1-1) but with randomly chosen indices and signs.

In RS-I Unvoiced HR frames, the decoding is depicted in Figure 6.1-1.



**Figure 6.1-1: Excitation Decoding for RS-I Unvoiced HR**

The gain indices  $i\hat{G}'$ , and  $i\hat{G}''[j]$ ,  $0 \leq j < 4$ , and the shaping filter index  $F_{idx}$  are decoded from the bitstream input to the decoder. The gains are unquantized using Equations (5.19.2.1-5), (5.19.2.1-8) and (5.19.2.1-9). A pseudo-random vector,  $e_{rand}(n)$ , is generated as described in Section 5.19.2.2. As described in Section 5.19.2.3, this pseudo-random vector is used to generate a gain scaled sparse signal,  $r_1(n)$ , and the shaping filter  $SF(z)$  to yield the RS-I Unvoiced HR frame excitation. As shown in Equation (5.19.2.4-8), the index of the filter,  $F_{idx}$  is used to establish  $SF(z)$ . This procedure is repeated twice per frame (the gains and shaping filter index are encoded once every half-frame). The RS-I Unvoiced HR frame excitation is then filtered by the LP synthesis filter, and the post-processing filter to generate speech.

- 3) **Decoding of the adaptive and innovative codebook gains:** In FR, Voiced HR, and Generic HR encoding types, the received index provides the adaptive codebook gain  $\hat{g}_p$  and the fixed-codebook gain correction factor  $\hat{\gamma}$ . Note that in case of Voiced HR and Generic HR, only half of the gain codebook is used. The estimated fixed-codebook gain  $g'_c$  is found as described in Section 5.20. First, the predicted energy for every subframe  $n$  is found by

$$\tilde{E}(n) = \sum_{i=1}^4 b_i \hat{R}(n-i) \quad (6.1-1)$$

and then the average innovation energy is found by

$$E_i = 10 \log \left( \frac{1}{N} \sum_{i=0}^{N-1} c^2(i) \right) \quad (6.1-2)$$

1  
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38

the predicted gain  $G'_c$  in dB is found by

$$G'_c = \tilde{E}(n) + \bar{E} - E_i \quad (6.1-3)$$

The prediction gain in the linear domain is given by

$$g'_c = 10^{0.05G'_c} = 10^{0.05(\tilde{E}(n) + \bar{E} - E_i)} \quad (6.1-4)$$

The quantized fixed-codebook gain is given by

$$\hat{g}_c = \hat{\gamma}g'_c \quad (6.1-5)$$

In case of Unvoiced HR and Unvoiced QR, only the fixed-codebook gain is transmitted. The received index  $k$  gives the gain prediction error in dB,  $\hat{\Gamma}$ , using the relation

$$\hat{\Gamma} = k \times \delta + \Gamma_{\min} \quad (6.1-6)$$

with the quantization step defined in Equation (5.20.1-2) and with the values  $\Gamma_{\min}$  and  $\delta$  for Unvoiced HR and Unvoiced QR given in Section 5.20.1. The predicted gain is then computed as in (6.1-3) and the quantized gain in dB is found by

$$\hat{G}_c = \hat{\Gamma} + G'_c \quad (6.1-7)$$

and the quantized gain is given by

$$\hat{g}_c = 10^{0.05\hat{G}_c} \quad (6.1-8)$$

For RS-I Generic FR and RS-I Voiced FR, the received index also provides the two adaptive codebook gains  $\hat{g}_1$  and  $\hat{g}_2$ .

- 4) **Computing the reconstructed excitation:** The following steps are for  $n = 0, \dots, 63$ . The total excitation is constructed by:

$$u'(n) = \hat{g}_p v(n) + \hat{g}_c c(n) \quad (6.1-9)$$

where  $c(n)$  is the codevector from the fixed-codebook after filtering it through the adaptive pre-filter  $F(z)$ . The excitation signal  $u'(n)$  is used to update the content of the adaptive codebook. The excitation signal  $u'(n)$  is then post-processed as described in the next section to obtain the post-processed excitation signal  $u(n)$  used at the input of the synthesis filter  $1/\hat{A}(z)$ .

For RS-I Generic FR and RS-I Voiced FR, the total excitation is computed by

$$u(n) = \hat{g}_1 v_1(n) + \hat{g}_2 v_2(n) + \hat{g}_c c(n).$$

The adaptive gain parameter used for post processing is now set equal to the sum of two adaptive codebook gains, i.e.,

$$\hat{g}_p = \hat{g}_1 + \hat{g}_2$$

and the adaptive codebook excitation is given by

$$v(n) = \frac{\hat{g}_1}{\hat{g}_p} v_1(n) + \frac{\hat{g}_2}{\hat{g}_p} v_2(n)$$

## 6.2 Excitation Post-processing

**Routine Name:** `enhancer`, `est_tilt`, `isf_stab`

**Inputs:**

- $v(n)$ : The adaptive codevector
- $c(n)$ : The algebraic codevector
- $\hat{g}_p$ : The quantized adaptive codebook gain
- $\hat{g}_c$ : The quantized algebraic codebook gain
- $f_i$ : The immitance spectral frequencies for current frame
- $f_i^{(p)}$ : The immitance spectral frequencies for previous frame

**Outputs:**

- $u(n)$ : The enhanced excitation signal
- $\theta$ : The ISF stability factor
- $r_v$ : The tilt of the excitation

**Initialization:**

- Memories and buffers are initialized to zero.

Before speech synthesis, a post-processing of excitation elements is performed.

### 6.2.1 Anti-Sparseness Processing in Generic HR

An adaptive anti-sparseness post-processing procedure is applied to the fixed-codebook vector  $c(n)$  in the Generic HR encoding type in order to reduce perceptual artifacts arising from the sparseness of the algebraic fixed-codebook vectors with only a few non-zero samples per subframe. The anti-sparseness processing consists of circular convolution of the fixed-codebook vector with an impulse response. Three pre-stored impulse responses are used and a number  $impNr = 0, 1, \text{ and } 2$  is set to select one of them. A value of 2 corresponds to no modification; a value of 1 corresponds to medium modification, while a value of 0 corresponds to strong modification. The selection of the impulse response is performed adaptively from the adaptive and fixed-codebook gains. The following procedure is employed:

if  $\hat{g}_p < 0.6$  then  
      $impNr = 0$ ;  
 else if  $\hat{g}_p < 0.9$  then  
      $impNr = 1$ ;  
 else  
      $impNr = 2$ ;

In other words, the onset is detected by comparing the fixed-codebook gain to the previous fixed-codebook gain. If the current value is more than three times the previous value an onset is detected.

If the onset is not detected and  $impNr = 0$ , the median filtered value of the current and the previous 4 adaptive codebook gains are computed. If this value is less than 0.6,  $impNr = 0$ .

Also, if the onset is not detected, the  $impNr$  value is restricted to increase by one step from the previous subframe.

If an onset is detected, the  $impNr$  value is increased by one, if it is less than 2.

## 6.2.2 Gain Smoothing for Noise Enhancement

A nonlinear gain smoothing technique is applied to the fixed-codebook gain  $\hat{g}_c$  in order to enhance excitation in noise. Based on the stability and voicing of the speech segment, the gain of the fixed-codebook vector is smoothed in order to reduce fluctuation in the energy of the excitation in case of stationary signals. This improves the performance in case of stationary background noise. The voicing factor is given by

$$\lambda = 0.5(1-r_v) \quad (6.2.2-1)$$

with

$$r_v = (E_v - E_c)/(E_v + E_c), \quad (6.2.2-2)$$

where  $E_v$  and  $E_c$  are the energies of the scaled pitch codevector and scaled innovation codevector, respectively ( $r_v$  gives a measure of signal periodicity). Note that since the value of  $r_v$  is between  $-1$  and  $1$ , the value of  $\lambda$  is between  $0$  and  $1$ . Note that the factor  $\lambda$  is related to the amount of unvoicing with a value of  $0$  for purely voiced segments and a value of  $1$  for purely unvoiced segments.

A stability factor  $\theta$  is computed based on a distance measure between the adjacent LP filters. Here, the factor  $\theta$  is related to the ISF distance measure. The ISF distance is given by

$$ISF_{dist} = \sum_{i=0}^{14} (f_i - f_i^{(p)})^2 \quad (6.2.2-3)$$

where  $f_i$  are the ISFs in the present frame, as defined in Equation (5.13-1), and  $f_i^{(p)}$  are the ISFs in the past frame. The stability factor  $\theta$  is given by

$$\theta = 1.25 - ISF_{dist} / 400000 \quad \text{Constrained by } 0 \leq \theta \leq 1 \quad (6.2.2-4)$$

The ISF distance measure is smaller in case of stable signals. As the value of  $\theta$  is inversely related to the ISF distance measure, then larger values of  $\theta$  correspond to more stable signals. The gain-smoothing factor  $S_m$  is given by

$$S_m = \lambda\theta \quad (6.2.2-5)$$

The value of  $S_m$  approaches 1 for unvoiced and stable signals, which is the case of stationary background noise signals. For purely voiced signals or for unstable signals, the value of  $S_m$  approaches 0. An initial modified gain  $g_0$  is computed by comparing the fixed-codebook gain  $\hat{g}_c$  to a threshold given by the initial modified gain from the previous subframe,  $g_{-1}$ . If  $\hat{g}_c$  is larger or equal to  $g_{-1}$ , then  $g_0$  is computed by decrementing  $\hat{g}_c$  by 1.5 dB bounded by  $g_0 \geq g_{-1}$ . If  $\hat{g}_c$  is smaller than  $g_{-1}$ , then  $g_0$  is computed by incrementing  $\hat{g}_c$  by 1.5 dB constrained by  $g_0 \leq g_{-1}$ .

Finally, the gain is updated with the value of the smoothed gain as follows

$$\hat{g}_c = S_m g_0 + (1 - S_m) \hat{g}_c \quad (6.2.2-6)$$

### 6.2.3 Pitch Enhancer for Generic and Voiced Encoding Types in Rate-Set I and Rate-Set II

A pitch enhancer scheme modifies the total excitation  $u'(n)$  by filtering the fixed-codebook excitation through an innovation filter whose frequency response emphasizes the higher frequencies and reduces the energy of the low frequency portion of the innovative codevector, and whose coefficients are related to the periodicity in the signal. A filter of the form

$$F_{inno}(z) = -c_{pe}z + 1 - c_{pe}z^{-1} \quad (6.2.3-1)$$

is used where  $c_{pe} = 0.125(1 - r_v)$ , with  $r_v$  being a periodicity factor given by  $r_v = (E_v - E_c)/(E_v + E_c)$  as described above. The filtered fixed-codebook codevector is given by

$$c'(n) = c(n) - c_{pe}(c(n+1) + c(n-1)) \quad (6.2.3-2)$$

and the updated post-processed excitation is given by

$$u(n) = \hat{g}_p v(n) + \hat{g}_c c'(n) \quad (6.2.3-3)$$

The above procedure can be done in one step by updating the excitation as follows

$$u(n) = u'(n) - \hat{g}_c c_{pe}(c(n+1) + c(n-1)) \quad (6.2.3-4)$$

## 6.3 Synthesis, Post-processing and Up-sampling

**Routine Name:** deemph, bass\_postfilter

**Inputs:**

- $u'(n)$ : The excitation signal
- $\hat{g}_p$ : The quantized adaptive codebook gain
- $\hat{a}_i$ : The quantized LP filter coefficients
- $T$ : Transmitted pitch lag values

**Outputs:**

- $\hat{s}(n)$  : The lower band speech synthesis at 12.8 kHz sampling rate
- The 16 or 8 kHz sampled synthesized speech.

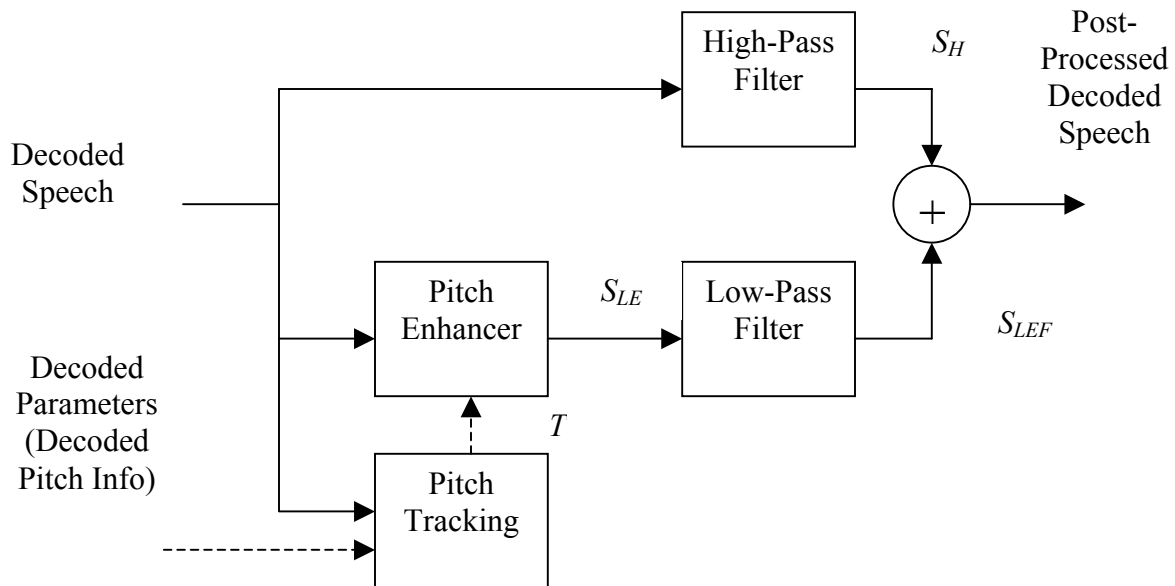
**Initialization:**

- All memories and buffers are set to zero at initialization except the mean prediction error energy  $\bar{E}_{pp}$ , which is set to 40 at initialization.

The synthesis is performed by filtering the post-processed excitation signal  $u(n)$  through the LP synthesis filter  $1/\hat{A}(z)$ . The synthesized signal is then de-emphasized by filtering through the filter  $1/(1-0.68z^{-1})$  (inverse of the pre-emphasis filter applied at the input). The synthesis and de-emphasis modules reproduce the reconstructed signal in the down-sampled domain of 12.8 kHz. The signal needs to be up-sampled to 16 kHz and then added to the high frequency signal generated from 6.4 to 8 kHz as will be described below. Before up-sampling to 16 kHz, low frequency pitch enhancement is applied to the synthesized signal and this is combined with the up-sampling procedure as will be described in the next section.

**6.3.1 Low-Frequency Pitch Enhancement Post-processing**

In the low-frequency pitch enhancement, two-band decomposition is used and adaptive filtering is applied only to the lower band. This results in a total post-processing that is mostly targeted at frequencies near the first harmonics of the synthesized speech signal.



**Figure 6.3-1: Block diagram of the low frequency pitch enhancer.**

Figure 6.3-1 shows the block diagram of the two-band pitch enhancer. In the higher branch the decoded speech signal is filtered by a high-pass filter to produce the higher band signal ( $s_H$ ). In the lower branch, the decoded speech signal is first processed through an adaptive pitch enhancer, and then filtered through a low-pass filter to obtain the lower band, post-processed signal ( $s_{LEF}$ ). The post-processed decoded speech signal is obtained by adding the lower band post-processed signal and the higher band signal. The object of the pitch enhancer is to reduce the inter-harmonic noise in the

1 decoded speech signal, which is achieved here by a time-varying linear filter described by the  
2 following equation:

$$3 \hat{s}_f(n) = (1 - \alpha)\hat{s}(n) + \alpha s_p(n) \quad (6.3.1-1)$$

5 where  $\alpha$  is a coefficient that controls the inter-harmonic attenuation,  $T$  is the pitch period of the input  
6 signal  $\hat{s}(n)$  and  $\hat{s}_f(n)$  is the output signal of the pitch enhancer.  $s_p(n)$  is the two-sided long-term  
7 prediction signal that is computed in each subframe as

$$8 s_p(n) = 0.5\hat{s}(n-T) + 0.5\hat{s}(n+T) \quad (6.3.1-2)$$

11 Parameters  $T$  and  $\alpha$  vary with time and are given by the pitch tracking module. With a value of  $\alpha = 1$ ,  
12 the gain of the filter described by Equation (6.3.1-1) is exactly 0 at frequencies  $1/(2T), 3/(2T), 5/(2T)$ ,  
13 etc.; i.e. at the mid-point between the harmonic frequencies  $1/T, 3/T, 5/T$ , etc. When  $\alpha$  approaches 0,  
14 the attenuation between the harmonics produced by the filter of Equation (6.3.1-1) decreases.

16 In case of FR and Generic HR encoding types, the received closed-loop pitch lag in each subframe is  
17 directly used (the fractional pitch lag rounded to the nearest integer). In case of Voiced HR, the pitch  
18 lag in the middle of the subframe is used (based on the delay contour).

20 The factor  $\alpha$  is computed as follows. The correlation between the signal and the predicted signal is  
21 given by

$$22 C_p = \sum_{n=0}^{N-1} \hat{s}(n)s_p(n) \quad (6.3.1-3)$$

24 and the energy of the predicted signal is given by

$$25 E_p = \sum_{n=0}^{N-1} s_p(n)s_p(n) \quad (6.3.1-4)$$

28 The factor  $\alpha$  is given by

$$29 \alpha = \frac{C_p}{0.5(E_p + 10^{0.1\bar{E}_{pp}})} \quad \text{constrained by } 0 \leq \alpha \leq 1 \quad (6.3.1-5)$$

32 where  $\bar{E}_{pp}$  is the mean prediction error energy in dB in the present subframe. The mean prediction  
33 error energy  $\bar{E}_{pp}$  is updated for the next subframe as follows. The long-term prediction error is first  
34 computed by

$$35 e_p(n) = \hat{s}(n) - \frac{C_p}{E_p} s_p(n) \quad (6.3.1-6)$$

37 and then pre-emphasized using the relation

$$38 e_{pp}(n) = e_p(n) - 0.95e_p(n-1) \quad (6.3.1-7)$$

41 The energy of the pre-emphasized error signal is then computed in dB as

$$E_{pp} = 10 \log \left[ \sum_{n=0}^{N-1} e_{pp}(n) e_{pp}(n) \right] \quad (6.3.1-8)$$

The mean error energy is then updated in every subframe by

$$\bar{E}_{pp} = 0.99 \bar{E}_{pp} + 0.01 E_{pp} \quad (6.3.1-9)$$

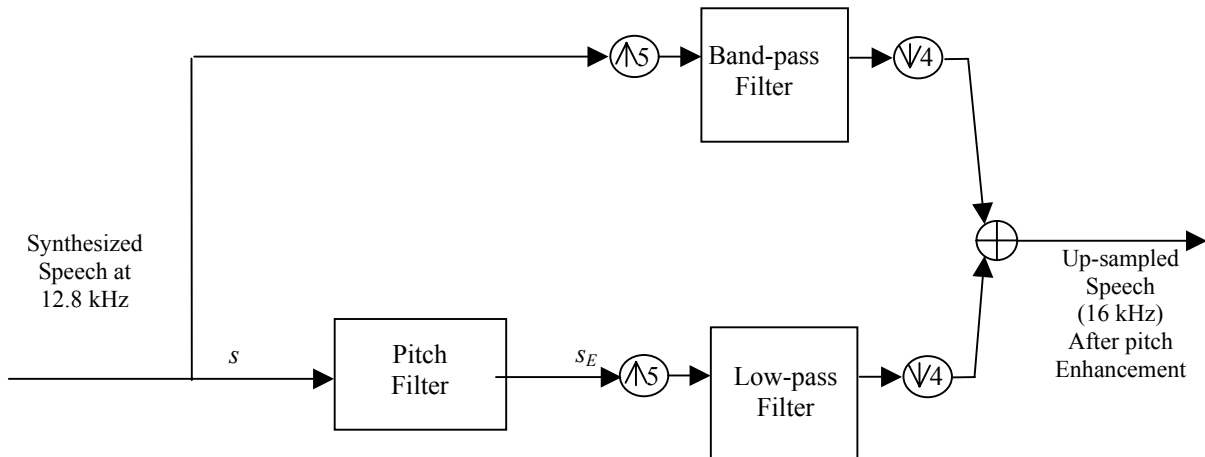
with initial value  $\bar{E}_{pp} = 40$

Note that in order to perform forward pitch prediction in Equation (6.3.1-2) (to compute  $\hat{s}(n+T)$ ), the synthesis should be extended after the end of the frame by 231 samples (the maximum pitch lag). This is done by first extending the excitation signal using the relation

$$u(n+L) = \hat{g}_p u(n+L-T), \quad n = 0, \dots, 230 \quad (6.3.1-10)$$

where  $L=256$  is the frame size, and then computing the extended synthesis signal by applying synthesis filtering and de-emphasis.

In the VMR-WB C simulation, since the signal need to be re-sampled to 16 kHz using an interpolation filter, the high-pass and low-pass filters in Figure 6.3-1 are combined with the interpolation filter to directly compute the pitch enhanced re-sampled signal. This is shown in Figure 6.3-2. The low frequency pitch enhancement is applied to the first 500 Hz of the frequency band. In case of up-sampling to 16 kHz, the band-pass filter in Figure 6.3-2 has a bandwidth from 500 Hz to 6.4 kHz in the 32 kHz up-sampled bandwidth. The Low pass filter has a bandwidth of 500 Hz.



**Figure 6.3-2: Block diagram of low frequency pitch enhancement combined with up-sampling.**

### 6.3.2 High-Pass Filtering

After low frequency pitch enhancement and re-sampling, the synthesis signal is high pass filtered as a precaution against undesired low frequency components. At 16 kHz output, a 50-Hz high pass filter is used, in the form

$$H_{hp16k}(z) = \frac{0.986211925 - 1.972423850z^{-1} + 0.986211925z^{-2}}{1 + 1.972233729z^{-1} - 0.972613969z^{-2}} \quad (6.3.2-1)$$

At 8 kHz output, a 100-Hz high pass filter is used, in the form

$$H_{hp8k}(z) = \frac{0.945976856 - 1.891953712z^{-1} + 0.945976856z^{-2}}{1 + 1.889033079z^{-1} - 0.894874345z^{-2}} \quad (6.3.2-2)$$

## 6.4 Reconstruction of High-Frequency Band

**Routine Name:** hf\_synth

**Inputs:**

- $\hat{s}(n)$ : The lower band speech synthesis at 12.8 kHz sampling rate
- $u(n)$ : The enhanced excitation signal
- $\hat{a}_i$ : The quantized LP filter coefficients

**Outputs:**

- $\hat{s}_{16k}(n)$ : The 16 kHz sampled synthesized speech with reconstructed high-frequency band

**Initialization:**

- All buffers and filter memories are set to zero at initialization. The seed of high-frequency random regeneration is initialized to 21845.

For the higher frequency band (6.4-7.0 kHz), excitation is generated to model the highest frequencies. The high frequency content is generated by filling the upper part of the spectrum with a white noise properly scaled in the excitation domain, then converted to the speech domain by spectrally shaping it with a filter derived from the same LP synthesis filter used for synthesizing the down-sampled signal.

### 6.4.1 Generation of High-Band Excitation

The high-band excitation is obtained by first generating white noise  $u_{HB1}(n)$ . The power of the high-band excitation is set equal to the power of the lower band excitation  $u_2(n)$ , which means that

$$u_{HB2}(n) = u_{HB1}(n) \sqrt{\frac{\sum_{k=0}^{63} u_2^2(k)}{\sum_{k=0}^{63} u_{HB1}^2(k)}} \quad (6.4.1-1)$$

Finally the high-band excitation at 16 kHz sampling is found by

$$u_{HB}(n) = \hat{g}_{HB} u_{HB2}(n) \quad (6.4.1-2)$$

where  $\hat{g}_{HB}$  is a scaling gain factor estimated using voicing information bounded by [0.1,1.0]. First, tilt of synthesis  $e_{\text{tilt}}$  is found

1

$$e_{\text{tilt}} = \frac{\sum_{n=1}^{63} \hat{s}_{hp}(n) \hat{s}_{hp}(n-1)}{\sum_{n=0}^{63} \hat{s}_{hp}^2(n)} \quad (6.4.1-3)$$

3

4 where  $\hat{s}_{hp}(n)$  is high-pass filtered lower band speech synthesis  $\hat{s}(n)$  with cut-off frequency of 400  
5 Hz. The scaling gain  $\hat{g}_{HB}$  is then found by

6

$$g_{HB} = w_{SP} g_{SP} + (1 - w_{SP}) g_{BG} \quad (6.4.1-4)$$

8

9 where  $g_{SP} = 1 - e_{\text{tilt}}$  is the gain for speech signal,  $g_{BG} = 0$  is the gain for background noise signal, and  
10  $w_{SP}$  is a weighting function set to 1, when VAD is ON, and 0 when VAD is OFF. In other words,  $g_{HB} =$   
11  $g_{SP}$  in case of speech signal and  $g_{HB} = 0$  in case of background noise signal (no high frequency  
12 generation). The VAD information is derived from the encoding type (VAD is OFF in case of CNG-ER  
13 or CNG-QR and VAD is ON otherwise). The encoding rate is signaled to the decoder by the receiving  
14 site multiplex sublayer.  $g_{HB}$  is bounded between [0.1, 1.0]. In case of voiced segments where less  
15 energy is present at high frequencies,  $e_{\text{tilt}}$  approaches 1 resulting in a lower gain  $g_{HB}$ . This reduces the  
16 energy of the generated noise in case of voiced segments.

17

#### 18 6.4.2 LP Filter for the High-Frequency Band

19

20 The high-band LP synthesis filter  $A_{HB}(z)$  is used for spectral shaping of the high frequency noise  
21 signal. The filter  $A_{HB}(z)$  is given by the weighted low-band LP synthesis filter as

$$A_{HB}(z) = \hat{A}(z/0.8) \quad (6.4.2-1)$$

23

24 where  $\hat{A}(z)$  is the interpolated LP synthesis filter.  $\hat{A}(z)$  has been computed analyzing signal with the  
25 sampling rate of 12.8 kHz but it is now used for a 16 kHz signal. Effectively, this means that the  
26 frequency response  $FR_{16}(f)$  of  $A_{HB}(z)$  is obtained by

$$FR_{16}(f) = FR_{12.8}\left(\frac{12.8}{16}f\right) \quad (6.4.2-2)$$

28 where  $FR_{12.8}(f)$  is the frequency response of  $A(z)$ . This means that the band 5.1-5.6 kHz in 12.8 kHz  
29 domain will be mapped to 6.4-7.0 kHz in 16 kHz domain.

30

#### 31 6.4.3 High-Band Synthesis

32

33  $u_{HB}(n)$  is filtered through  $A_{HB}(z)$ . The output of this high-band synthesis  $s_{HB}(n)$  is filtered through a  
34 band-pass FIR filter  $H_{HB}(z)$  which has the pass-band from 6 to 7 kHz. Finally,  $s_{HB}$  is added to the  
35 synthesized speech  $\hat{s}_{16k}(n)$  to produce the synthesized output speech signal  $\hat{s}_{\text{output}}(n)$ .

### 36 6.5 Frame Error Concealment

37

38 **Routine Name:** isf\_dec\_bfi, syn\_bfi

39 **Inputs:**

- 40 •  $u'(n)$ : The excitation signal from the previous frame

- 1 • Classification decision of previous frame
- 2 •  $\theta$ : The ISF stability factor
- 3 •  $b(i)$ : The adaptive codebook gains for each subframe of last good frame
- 4 •  $g(i)$ : The algebraic codebook gains for each subframe of last good frame
- 5 •  $T_c$ : Last reliable pitch lag
- 6 •  $r_v$ : The tilt of the excitation
- 7 •  $\bar{E}_s$ : The smoothed CNG excitation energy

8 **Outputs:**

- 9 •  $u'(n)$ : The excitation signal with controlled energy
- 10 •  $u(n)$ : The enhanced excitation signal with controlled energy
- 11 •  $E$ : The synthesized speech energy

12 **Initialization:**

- 13 • All buffers and filter memories are set to zero at initialization. The seed used to generate the random part of the excitation is initialized to 21845.

15  
 16 In case of frame erasures, the concealment strategy can be summarized as a convergence of the  
 17 signal energy and the spectral envelope to the estimated parameters of the background noise. The  
 18 periodicity of the signal is converged to zero. The speed of the convergence is dependent on the  
 19 parameters of the last correctly received frame class and the number of consecutive erased frames  
 20 and is controlled by an attenuation factor  $\alpha$ . The factor  $\alpha$  is further dependent on the stability of the  
 21 LP filter for UNVOICED frames. In general, the convergence is slow if the last good received frame is  
 22 in a stable segment and is rapid if the frame is in a transition segment. The values of  $\alpha$  are  
 23 summarized in Table 6.5-1.

24 **Table 6.5-1: Values of the FER concealment attenuation factor  $\alpha$**

Last Good Received Frame	Number of successive erased frames	$\alpha$
ARTIFICIAL ONSET		0.6
ONSET, VOICED	$\leq 3$	1.0
	$> 3$	0.4
VOICED TRANSITION		0.4
UNVOICED TRANSITION		0.8
UNVOICED	= 1	$0.6 \theta + 0.4$
	$> 1$	0.4

25  
 26 A stability factor  $\theta$  is computed based on a distance measure between the adjacent LP filters. Here,  
 27 the factor  $\theta$  is related to the ISF distance measure and it is bounded by  $0 \leq \theta \leq 1$ , with larger values of  $\theta$   
 28 corresponding to more stable signals. This results in decreasing energy and spectral envelope  
 29 fluctuations when an isolated frame erasure occurs inside a stable unvoiced segment. The signal  
 30 class remains unchanged during the processing of erased frames, i.e. the class remains the same as  
 31 in the last correctly received frame.

32  
 33 All the supplementary information necessary for better erasure concealment is transmitted only in  
 34 Generic FR. In Signaling HR encoding scheme, the class information and the energy information is  
 35 also preserved. In addition to these encoding schemes, the frame class is implicitly transmitted also  
 36 for VOICED HR frame and UNVOICED HR and QR frames. For the remaining RS-II active speech

frames, the class is estimated using the measure of signal periodicity  $r_v$  (6.2.2-2) averaged over all subframes. The classification is done following the rules in Table 6.5-2.

**Table 6.5-2: Signal classification rules at the decoder if not transmitted**

Previous Frame Class	Rule	Current Frame Class
ONSET	$r_v > -0.1$	VOICED
VOICED	$-0.1 \geq r_v \geq -0.5$	VOICED TRANSITION
VOICED TRANSITION	$r_v < -0.5$	UNVOICED
UNVOICED TRANSITION	$r_v > -0.1$	ONSET
UNVOICED	$-0.1 \geq r_v \geq -0.5$	UNVOICED TRANSITION
	$r_v < -0.5$	UNVOICED

### 6.5.1 Construction of the Periodic Part of the Excitation

When a frame is found to be in error by the receiving site multiplex sublayer, an Erasure packet type is provided to the VMR-WB decoder. For a concealment of erased frames following a correctly received UNVOICED frame, no periodic part of the excitation is generated. For a concealment of erased frames following a correctly received frame other than UNVOICED, the periodic part of the excitation is constructed by repeating the last pitch period of the previous frame. If this is the case of the 1<sup>st</sup> erased frame after a good frame, this pitch pulse is first low-pass filtered. The filter used is a simple 3-tap linear phase FIR filter with the coefficients equal to 0.18, 0.64 and 0.18. The pitch period  $T_c$  used to select the last pitch pulse and hence used during the concealment is defined so that pitch multiples or submultiples can be avoided, or reduced. The following logic is used in determining the pitch period  $T_c$

*if (  $(T_3 < 1.8 T_s)$  AND  $(T_3 > 0.6 T_s)$  ) OR  $(T_{cnt} \geq 30)$ , then  $T_c = T_3$ , else  $T_c = T_s$ .*

Here,  $T_3$  is the rounded pitch period of the 4<sup>th</sup> subframe of the last good received frame and  $T_s$  is the rounded pitch period of the 4<sup>th</sup> subframe of the last good stable voiced frame with coherent pitch estimates. A stable voiced frame is defined here as a VOICED frame preceded by a frame of voiced type (VOICED TRANSITION, VOICED, ONSET). The coherence of pitch is verified in this implementation by examining whether the closed-loop pitch estimates are reasonably close; i.e. whether the ratios between the last subframe pitch, the 2<sup>nd</sup> subframe pitch and the last subframe pitch of the previous frame are within the interval (0.7, 1.4).

This determination of the pitch period  $T_c$  implies that if the pitch at the end of the last good frame and the pitch of the last stable frame are close, the pitch of the last good frame is used. Otherwise, this pitch is considered unreliable and the pitch of the last stable frame is used instead to avoid the impact of erroneous pitch estimates at voiced onsets. This logic is valid only if the last stable segment is not too far in the past. Hence a counter  $T_{cnt}$  is defined that limits the effect of the last stable segment. If  $T_{cnt}$  is greater or equal to 30; i.e. if there are at least 30 frames since the last  $T_s$  update, the last good frame pitch is used systematically.  $T_{cnt}$  is reset to 0 every time a stable segment is detected and  $T_s$  is updated. The period  $T_c$  is then maintained constant during the concealment for the entire erased block.

As the last pulse of the excitation of the previous frame is used for the construction of the periodic part, its gain is approximately correct at the beginning of the concealed frame and can be set to 1. The gain is then attenuated linearly throughout the frame on a sample-by-sample basis to achieve the value of  $\alpha$  at the end of the frame.

The values of  $\alpha$  correspond to the Table 6.5-1 with the exception that they are modified for erasures following VOICED and ONSET frames to take into account the energy evolution of voiced segments. This evolution can be extrapolated to some extent by using the pitch excitation gain values of each subframe of the last good frame. In general, if these gains are greater than 1, the signal energy is

1 increasing, if they are lower than 1, the energy is decreasing.  $\alpha$  is thus multiplied by a correction  
2 factor  $f_b$  computed as follows:

$$3 \quad f_b = \sqrt{0.1b(0) + 0.2b(1) + 0.3b(2) + 0.4b(3)} \quad (6.5.1-1)$$

4  
5  
6 where  $b(0)$ ,  $b(1)$ ,  $b(2)$  and  $b(3)$  are the pitch gains of the four subframes of the last correctly  
7 received frame. The value of  $f_b$  is constrained between 0.98 and 0.85 before being used to scale the  
8 periodic part of the excitation. In this way, strong energy increases and decreases are avoided. For  
9 erased frames following a correctly received frame other than UNVOICED, the excitation buffer is  
10 updated with this periodic part of the excitation only. This update will be used to construct the  
11 adaptive codebook excitation in the next frame.

## 12 **6.5.2 Construction of the Random Part of the Excitation**

13  
14  
15 The innovative (non-periodic) part of the excitation is generated randomly. A simple random  
16 generator with approximately uniform distribution is used. Before adjusting the innovation gain, the  
17 randomly generated innovation is scaled to some reference value, fixed here to the unitary energy per  
18 sample. At the beginning of an erased block, the innovation gain  $g_s$  is initialized by using the  
19 innovative excitation gains of each subframe of the last good frame

$$20 \quad g_s = 0.1g(0) + 0.2g(1) + 0.3g(2) + 0.4g(3) \quad (6.5.2-1)$$

21  
22  
23 where  $g(0)$ ,  $g(1)$ ,  $g(2)$  and  $g(3)$  are the fixed-codebook, or innovation, gains of the four subframes of  
24 the last correctly received frame. The attenuation strategy of the random part of the excitation is  
25 somewhat different from the attenuation of the pitch excitation. The reason is that the pitch excitation  
26 (and thus the excitation periodicity) is converging to 0 while the random excitation is converging to the  
27 CNG excitation energy. The innovation gain attenuation is done as

$$28 \quad g_s^1 = \alpha g_s^0 + (1 - \alpha)g_n \quad (6.5.2-2)$$

29  
30  
31 where  $g_s^1$  is the innovative gain at the beginning of the next frame,  $g_s^0$  is the innovative gain at the  
32 beginning of the current frame,  $g_n$  is the gain of the excitation used during the comfort noise  
33 generation and  $\alpha$  is as defined in Table 6.5-1. Similarly to the periodic excitation attenuation, the gain  
34 is thus attenuated linearly throughout the frame on a sample-by-sample basis starting with  $g_s^0$  and  
35 going to the value of  $g_s^1$  that would be achieved at the beginning of the next frame.

36  
37 Finally, if the last correctly received frame is different from UNVOICED, the innovation excitation is  
38 filtered through a linear phase FIR high-pass filter with coefficients -0.0125, -0.109, 0.7813, -0.109,  
39 and -0.0125. To decrease the amount of noisy components during voiced segments, these filter  
40 coefficients are multiplied by an adaptive factor equal to  $(0.75 - 0.25 r_v)$ , with  $r_v$  denoting the voicing  
41 factor as defined in Equation (6.2.2-2). The random part of the excitation is then added to the  
42 adaptive excitation to form the total excitation signal. If the last good frame is UNVOICED, only the  
43 innovative excitation is used and it is further attenuated by a factor of 0.8. In this case, the past  
44 excitation buffer is updated with the innovation excitation, as no periodic part of the excitation is  
45 available.

## 46 **6.5.3 Spectral Envelope Concealment, Synthesis and Updates**

47  
48  
49 To synthesize the decoded speech, the LP filter parameters must be obtained. The spectral envelope  
50 is gradually moved to the estimated envelope of the background noise. Here the ISF representation  
51 of LP parameters is used

$$I^1(j) = \alpha I^0(j) + (1 - \alpha) I_n(j), \quad j = 0, \dots, 15 \quad (6.5.3-1)$$

In Equation (6.5.3-1),  $I^1(j)$  is the value of the  $j$ th ISF of the current frame,  $I^0(j)$  is the value of the  $j$ th ISF of the previous frame, and  $I_n(j)$  is the value of the  $j$ th ISF of the estimated comfort noise envelope.

The synthesized speech is obtained by filtering the excitation signal through the LP synthesis filter and post-processed similar to the procedure described in Sections 6.3 and 6.4. The filter coefficients are computed from the ISF representation and are interpolated for each subframe (four times per frame) as during normal decoder operation. As innovative gain quantizer and ISF quantizer both use a prediction, their memory will not be up to date after the normal operation is resumed. To reduce this effect, the quantizers' memories are estimated and updated at the end of each erased frame.

#### 6.5.4 Recovery of Normal Operation after an Erasure

The problem of the recovery after an erased block of speech frames is essentially due to the strong prediction. The CELP type speech coders achieve their high signal to noise ratio for voiced speech due to the fact that they exploit the past excitation signal to encode the present frame excitation (long-term or pitch prediction). Also, most of the quantizers (LP quantizers, gain quantizers) make use of a prediction.

##### 6.5.4.1 Artificial Onset Reconstruction

**Routine Name:** `voiced_onset`

**Inputs:**

- $u(n)$ : The excitation signal from the previous frame
- $h(i)$ : The LP synthesis filter impulse response
- $E$ : The synthesized speech energy
- $\tau_q$ : First glottal pulse position with respect to frame beginning
- $p(i)$ : Pitch lags for each subframe
- $g(i)$ : The algebraic codebook gains for each subframe

**Outputs:**

- $u'(n)$ : The excitation signal

**Initialization:**

- None

The most complicated case related to the use of the long-term prediction is when a voiced onset is lost. The lost onset means that the voiced speech onset happened at some point in the erased block. In this case, the last correctly received frame was unvoiced or inactive and thus no periodic excitation is found in the excitation buffer. The first good speech frame after the erased block is however voiced, the excitation buffer at the encoder is highly periodic and the adaptive excitation has been encoded using this periodic past excitation. As this periodic part of the excitation is completely missing at the decoder, it can take up to several frames to recover from this loss.

If an ONSET class is lost (i.e., a good VOICED frame is received after an erasure, but the last good frame before the erasure was UNVOICED) and the first good frame after the erasure is of Generic FR type, a special technique is used to artificially reconstruct the lost onset and to trigger the voiced synthesis. The technique is used only in Generic FR encoding type as this is the only encoding scheme where all the supplementary information for better frame erasure concealment is transmitted.

1 At the beginning of the 1<sup>st</sup> good frame after a lost onset, the periodic part of the excitation is  
 2 constructed artificially as a low-pass filtered periodic train of pulses separated by a pitch period. In  
 3 this case, the low-pass filter is a simple linear phase FIR filter with the impulse response  
 4  $h_{low} = \{-0.0125, 0.109, 0.7813, 0.109, \text{ and } -0.0125\}$ . The innovative part of the excitation is  
 5 constructed using normal CELP decoding.

6  
 7 In practice, the length of the artificial onset is limited so that at least one complete pitch period is  
 8 constructed by this method and the method is continued to the end of the current subframe. After  
 9 that, a normal ACELP processing is resumed. The pitch value considered is the rounded average of  
 10 the decoded pitch values of all subframes where the artificial onset reconstruction is used. The low-  
 11 pass filtered impulse train is realized by placing the impulse responses of the low-pass filter in the  
 12 adaptive excitation buffer (previously initialized to zero). The first impulse response will be centered at  
 13 the quantized position  $\tau_q$  (transmitted within the bitstream) with respect to the frame beginning and the  
 14 remaining impulses will be placed with the distance of the averaged pitch up to the end of the last  
 15 subframe affected by the artificial onset reconstruction.

16  
 17 As an example, let us assume that the pitch values in the first and the second subframe be  
 18  $p(0)=70.75$  and  $p(1)=71$ . Since this is larger than the subframe size of 64, then the artificial onset will  
 19 be constructed during the first two subframes and the pitch period will be equal to the pitch average of  
 20 the two subframes rounded to the nearest integer; i.e. 71. The last two subframes will be processed  
 21 by normal ACELP decoder.

22  
 23 The energy of the periodic part of the artificial onset excitation is then scaled by the gain  
 24 corresponding to the quantized and transmitted energy for FER concealment (as defined in Equations  
 25 (5.22.2.1-1) and (5.22.2.1-2)) and divided by the gain of the LP synthesis filter. The LP synthesis filter  
 26 gain is computed as

$$27 \quad g_{LP} = \sqrt{\sum_{i=0}^{63} h_{LP}^2(i)} \quad (6.5.4.1-1)$$

28  
 29 where  $h_{LP}(i)$  is the LP synthesis filter impulse response. Finally, the artificial onset gain is reduced by  
 30 multiplying the periodic part with 0.96. The LP filter for the output speech synthesis is not interpolated  
 31 in the case of an artificial onset construction. Instead, the received LP parameters are used for the  
 32 synthesis of the whole frame.

### 33 6.5.5 Energy Control

34  
 35 **Routine Name:** `scale_syn`

36 **Inputs:**

- 37 •  $\hat{s}(n)$ : The speech synthesis at 12.8 kHz sampling rate
- 38 •  $u'(n)$ : The excitation signal
- 39 •  $u(n)$ : The enhanced excitation signal
- 40 •  $\hat{a}_i$ : The quantized LP filter coefficients
- 41 •  $p(i)$ : Pitch lags for each subframe
- 42 •  $E$ : The synthesized speech energy

43 **Outputs:**

- 44 •  $u'(n)$ : The excitation signal with controlled energy

- $u(n)$ : The enhanced excitation signal with controlled energy

**Initialization:**

- Buffers, filter memories, and static variables are set to zero at initialization.

The most important task in the recovery after an erased block of speech frames is to properly control the energy of the synthesized signal. The synthesis energy control is needed because of the strong prediction. The energy control is most important when a block of erased frames happens during a voiced segment. When a frame erasure occurs after a voiced frame, the excitation of the last good frame is typically used during the concealment with some attenuation strategy. When a new LP filter occurs with the first good frame after the erasure, there can be a mismatch between the excitation energy and the gain of the new LP synthesis filter. The new synthesis filter can produce a synthesis signal with energy highly different from the energy of the last synthesized erased frame and also from the original signal energy.

The energy control is performed in all active speech frames following an erasure despite the fact that only in Generic FR all the supplementary information necessary for better erasure concealment is transmitted. In Signaling HR encoding scheme, the class information and the energy information is also preserved.

The energy control during the first good frame after an erased frame can be summarized as follows. The synthesized signal is scaled so that its energy is similar to the energy of the end of the last erased frame at the beginning of the frame and is converging to the transmitted energy towards the end of the frame with preventing a too important energy increase.

The energy control is done in the synthesized speech signal domain. Even if the energy is controlled in the speech domain, the excitation signal must be scaled as it serves as long-term prediction memory for the following frames. The synthesis is then repeated to smooth the transitions. Let  $g_0$  denote the gain used to scale the 1<sup>st</sup> sample in the current frame and  $g_1$  the gain used at the end of the frame. The excitation signal is then scaled as follows

$$u_s(i) = g_{AGC}(i)u(i), \quad i=0, \dots, L-1 \quad (6.5.5-1)$$

where  $u_s(i)$  is the scaled excitation,  $u(i)$  is the excitation before the scaling,  $L$  is the frame length and  $g_{AGC}(i)$  is the gain starting from  $g_0$  and converging exponentially to  $g_1$

$$g_{AGC}(i) = f_{AGC} g_{AGC}(i-1) + (1-f_{AGC}) g_1 \quad i=0, \dots, L-1 \quad (6.5.5-2)$$

with the initialization of  $g_{AGC}(-1) = g_0$ , where  $f_{AGC}$  is the attenuation factor set in this implementation to the value of 0.98. This value has been found experimentally as a compromise of having a smooth transition from the previous (erased) frame on one side, and scaling the last pitch period of the current frame as much as possible to the correct (transmitted) value on the other side. This is important because the transmitted energy value is estimated pitch synchronously at the end of the frame. The gains  $g_0$  and  $g_1$  are defined as

$$g_0 = \sqrt{\frac{E_{-1}}{E_0}} \quad (6.5.5-3)$$

$$g_1 = \sqrt{\frac{E_q}{E_1}} \quad (6.5.5-4)$$

where  $E_{-1}$  is the energy computed at the end of the previous (erased) frame,  $E_0$  is the energy at the beginning of the current (recovered) frame,  $E_1$  is the energy at the end of the current frame and  $E_q$  is the quantized transmitted energy at the end of the current frame, computed at the encoder from

Equations (5.22.2.1-1) and (5.22.2.1-2). If  $E_q$  is not available,  $E_q$  is set to  $E_1$ .  $E_{-1}$  and  $E_1$  are computed similarly using the synthesized speech signal  $\hat{s}(n)$ . When  $E_{-1}$  is computed pitch synchronously, it uses the concealment pitch period  $T_c$  and  $E_1$  uses the last subframe-rounded pitch  $T_3$ .  $E_0$  is computed similarly using the rounded pitch value  $T_0$  of the first subframe, the equations (5.22.2.1-1) and (5.22.2.1-2) being modified to

$$E = \max(\hat{s}^2(i)) \quad i = 0, \dots, t_E \quad (6.5.5-5)$$

for VOICED and ONSET frames.  $t_E$  equals to the rounded pitch lag or twice that length if the pitch is shorter than 64 samples. For other frames,

$$E = \frac{1}{t_E} \sum_{i=0}^{t_E} (\hat{s}^2(i)) \quad (6.5.5-6)$$

with  $t_E$  equal to the half of the frame length. The gains  $g_0$  and  $g_1$  are further limited to a maximum allowed value, to prevent strong energy. This value has been set to 1.2 with the exception of Signaling HR frames or very low energy frames ( $E_q < 1.1$ ). In these two cases,  $g_1$  is limited to 1. If the erasure occurs during a voiced speech segment (i.e. the last good frame before the erasure and the first good frame after the erasure are classified as VOICED TRANSITION, VOICED or ONSET) and  $E_q$  is not transmitted, further precautions must be taken because of the possible mismatch between the excitation signal energy and the LP filter gain, mentioned previously. When the LP filter gain of the first frame after an erasure is higher than that of the LP gain of the last frame before the erasure, the energy of the excitation is adjusted to the gain of the new LP filter

$$E_q = E_1 \frac{E_{LP0}}{E_{LP1}} \quad (6.5.5-7)$$

where  $E_{LP0}$  is the energy of the LP filter impulse response of the last good frame before the erasure and  $E_{LP1}$  is the energy of the LP filter of the first good frame after the erasure. The LP filters of the last subframes of uncorrupted voiced-type frames (VOICED TRANSITION, VOICED or ONSET) are used. Finally, the value of  $E_q$  is limited to the value of  $E_1$  in this case (voiced segment erasure without  $E_q$  information being transmitted).

The following exceptions, all related to transitions in speech signal, further overwrite the computation of  $g_0$ . If artificial onset is used in the current frame,  $g_0$  is set to 0.5  $g_1$ , to make the onset energy increase gradually. In the case of a first good frame after an erasure classified as ONSET, the gain  $g_0$  is prevented to be higher than  $g_1$ . This precaution is taken to prevent a positive gain adjustment at the beginning of the frame (which is probably still at least partially unvoiced) from amplifying the voiced onset (at the end of the frame). Finally, during a transition from voiced to unvoiced (i.e. that last good frame being classified as VOICED TRANSITION, VOICED or ONSET and the current frame being classified UNVOICED) or during a transition from a non-active speech period to active speech period (last correctly received frame being encoded as comfort noise, current frame being encoded as active speech, but the lost voiced onset not being detected), the value of  $g_0$  is set to  $g_1$ .

The synthesized speech is obtained by filtering the excitation signal through the LP synthesis filter and post-processed similar to the procedure described in Sections 6.3 and 6.4.

### 6.5.6 Frame Erasure Concealment in Rate-Set I

The frame erasure concealment in case of Rate-Set I is the same as the one described above for Rate-Set II (Sections 6.5.1 to 6.5.5). However, only the energy information is transmitted in case of RS-I Voiced FR. Since no phase information is transmitted, the artificial onset reconstruction described in Section 6.5.4.1 is not used (used only in RS-II Generic FR).

1 The class information is not transmitted but estimated at the decoder as will be described in Section  
 2 6.5.6.1 below. Note that for Voiced and Unvoiced encoding types, the class information is implicitly  
 3 transmitted.

4  
 5 Due to the fact that the reliability of the class estimated at the decoder is lower than if it is estimated  
 6 in the encoder and transmitted, the attenuation factor  $\alpha$  is lowered to 0.8 for erased frames following  
 7 Generic RS-I frames classified as ONSET (2<sup>nd</sup> line of Table 6.5-1).

8  
 9 Further, if an erasure follows a Generic RS-I frame classified as UNVOICED TRANSITION, the  
 10 periodic part of the excitation is reset and the random part is made to converge to 0 at the end of the  
 11 frame ( $g_s^1=0$ ).

12  
 13 Finally, if an erasure follows a RS-I Unvoiced HR (NELP) frame, the stability factor  $\theta$  is set to 1, i.e.  
 14 the energy is maintained unchanged during the erasure. If the next correctly received frame is other-  
 15 than-Unvoiced active speech frame, the adaptive codebook is then reset before the excitation  
 16 decoding.

### 18 6.5.6.1 Signal Classification at the Decoder for Rate-Set I Frame Error Concealment and 19 Recovery

20  
 21 **Routine Name:** clas\_dec

#### 22 **Inputs:**

- 23 • class: Frame classification
- 24 • coder\_type: Speech encoding type
- 25 • previous frame class : Classification of last received good frame
- 26 •  $\hat{s}(n)$  : The synthesized speech signal (including the past synthesis buffer)
- 27 •  $\bar{E}_T$  : The long-term pitch synchronous energy

#### 28 **Outputs:**

- 29 • class: Frame classification
- 30 •  $\bar{E}_T$  : The long-term pitch synchronous energy

#### 31 **Initialization:**

- 32 • Initially,  $\bar{E}_T$  is set to 60.

33  
 34  
 35 The frame classification at the decoder in Rate-Set I is performed in case of Generic coding types  
 36 based on the following parameters: zero crossing parameter  $zcf$ , pitch-synchronous normalized  
 37 correlation  $R_T$ , pitch coherence parameter  $pc$ , spectral tilt  $e'$ , and pitch synchronous relative energy at  
 38 the end of the frame  $dE_T$ .

39  
 40 The zero-crossing parameter  $zcf$  is averaged over the 4 subframes. That is,

$$41 \quad zcf = \sum_{i=0}^3 zcs[i]$$

42  
 43 where  $zcs[i]$  is the number of times the signal sign changes from positive to negative during subframe  
 44  $i$ .

1 The pitch synchronous normalized correlation is computed based on a pitch lag  $T_0$ , where  $T_0$  is the  
 2 integer part of the pitch lag of the last subframe, or the average of the pitch lag of the last two  
 3 subframes if it is larger than  $3N/2$ , where  $N=64$  is the subframe size. That is  
 4

$$5 \quad T_0 = \begin{cases} \lfloor p_3 \rfloor & \text{if } \lfloor p_3 \rfloor \leq 3N/2 \\ \text{round}(p_2 + p_3) & \text{if } \lfloor p_3 \rfloor > 3N/2 \end{cases}$$

6 where  $p_i$  is the fractional pitch lag at subframe  $i$ .  
 7  
 8

9 The pitch synchronous normalized correlation computed at the end of the frame is given by  
 10

$$11 \quad R_T = \begin{cases} C_N(T_0) & \text{for } T_0 > N \\ 0.5C_N(T_0) + 0.5C_N(2T_0) & \text{for } T_0 \leq N \end{cases}$$

12 where  
 13

$$14 \quad C_N(kT_0) = \frac{\sum_{n=L-kT_0}^{L-1-(k-1)T_0} \hat{s}(n-(k-1)T_0)\hat{s}(n-kT_0)}{\sqrt{\sum_{n=L-kT_0}^{L-1-(k-1)T_0} \hat{s}(n-(k-1)T_0)\hat{s}(n-(k-1)T_0)} \sqrt{\sum_{n=L-kT_0}^{L-1-(k-1)T_0} \hat{s}(n-kT_0)\hat{s}(n-kT_0)}}, \quad k=1,2$$

15  
 16 where  $L$  is the frame size and  $\hat{s}(n)$  is the synthesized speech signal.  
 17  
 18

19 The pitch coherence parameter is given by  
 20

$$21 \quad pc = |p_3 + p_2 - p_1 - p_0|$$

22 where  $p_i$  is the fractional pitch lag at subframe  $i$ .  
 23  
 24  
 25

26 The spectral tilt parameter  $e'_t$  is estimated based on the last 3 subframes and given by  
 27

$$28 \quad e' = \frac{\sum_{n=N}^{L-1} \hat{s}(n)\hat{s}(n-1)}{\sum_{n=N}^{L-1} \hat{s}(n)\hat{s}(n)}$$

29 The pitch synchronous relative energy at the end of the frame is given by  
 30  
 31

$$32 \quad dE_T = E_T - \bar{E}_T$$

33 where  
 34

$$35 \quad E_T = 10 \log_{10} \left( \frac{1}{T'} \sum_{n=0}^{T'-1} \hat{s}^2(L-T'+n) \right)$$

36 and  $\bar{E}_T$  is the long-term energy.  $\bar{E}_T$  is updated only when a current frame is classified as VOICED  
 37 using the relation  
 38

$$\bar{E}_T = 0.99\bar{E}_T + 0.01E_T$$

The pitch lag value  $T'$  over which the energy  $E_T$  is computed is given by

$$p = \text{round}(0.5p_2 + 0.5p_3)$$

$$T' = p \quad \text{if } p \geq N$$

$$T' = 2p \quad \text{if } p < N$$

To make the classification more robust, the classification parameters are considered together forming a function of merit  $f_m$ . For that purpose, the classification parameters are first scaled so that each parameter's value typical for unvoiced signal translates in 0 and each parameter's value typical for voiced signal translates into 1. A linear function is used between them. The scaled version  $p^s$  of a certain parameter  $p_x$  is obtained using

$$p^s = k_p p_x + c_p$$

and in case of pc the scaled parameter is constrained by  $0 \leq pc^s \leq 1$ .

The function coefficients  $k_p$  and  $c_p$  have been found experimentally for each of the parameters so that the signal distortion due to the concealment and recovery techniques used in presence of frame errors is minimal. The values used are summarized in Table 6.5-3.

**Table 6.5-3: Signal Classification Parameters at decoder and the coefficients of their respective scaling functions**

Parameter	Meaning	$k_p$	$c_p$
$R_T$	Normalized Correlation	0.8547	0.2479
$e_t$	Spectral Tilt	0.8333	0.2917
$pc$	Pitch Coherence	-0.0357	1.6071
$dE_T$	Relative Frame Energy	0.04	0.56
$zcf$	Zero Crossing Counter	-0.04	2.52

The merit function has been defined as

$$f = \frac{1}{6}(2R_T^s + e^s + pc^s + dE_T^s + zcf^s)$$

where the superscript  $s$  indicates the scaled version of the parameters. In the case of 8 kHz sampled output, the merit function  $f$  is further multiplied by 0.9.

The classification is performed using the merit function  $f_m$  and following the rules summarized in Table 6.5-4. The default class is UNVOICED. Note that the class ARTIFICIAL ONSET is set at the decoder if the frame follows an erased frame and artificial onset reconstruction is used.

**Table 6.5-4: Signal Classification Rules at the decoder in RS-I**

Previous Frame Class	Rule	Current Frame Class
ONSET	$f_m \geq 0.63$	VOICED
ARTIFICIAL ONSET	$0.63 > f_m \geq 0.39$	VOICED TRANSITION
VOICED	$f_m < 0.39$	UNVOICED
VOICED TRANSITION		
UNVOICED TRANSITION	$f_m > 0.56$	ONSET
	$0.56 \geq f_m > 0.45$	UNVOICED TRANSITION
	$f_m \leq 0.45$	UNVOICED

## 6.6 Decoding of Inactive Speech Frames (CNG-ER and CNG-QR)

**Routine Name:** CNG\_dec, CNG\_synthesis

**Inputs:**

- Decoded quantization indices

**Outputs:**

- $u'(n)$ : The excitation signal

**Initialization:**

- Initialization is done the same way as described in Section 5.23.

In RS-II CNG ER, the quantized value of the energy per sample information in the  $\log_2$  domain is found by

$$\hat{E}_{s,2} = \frac{24}{63} \text{index} - 2 \quad (6.6-1)$$

and in RS-I CNG ER, it is given by (only 5 bits used)

$$\hat{E}_{s,2} = \frac{24}{31} \text{index} - 2$$

The quantized energy per sample in the linear domain is then found as

$$\hat{E}_s = 2^{\hat{E}_{s,2}} \quad (6.6-2)$$

The received ISP indices are used to obtain the quantized ISP parameter vector. The initialization and update of the smoothed energy and the LP filter are performed as described in Section 5.23. Also, generating and scaling of the random excitation is performed similar to Section 5.23. Furthermore, memory update is similar to Section 5.21. Note that in case of CNG-ER and CNG-QR frames, the procedures described in Sections 6.1, 6.2, 6.4, and 6.5 are not performed. The synthesized speech is obtained by filtering the excitation signal through the LP synthesis filter and post-processed the same way as described in Section 6.3.

## 6.7 Detection and Concealment of Frames with Corrupted Rate Information

The detection and concealment of frames with corrupted rate information in the VMR-WB decoder is described in this section. Note that this procedure is not performed in the storage format. The algorithm comprises a number of tests that are performed in two stages. The first stage is performed by default, whereas the second stage is performed only if compilation option BRH\_LEVEL2 in VMR-WB C simulation is selected.

### 6.7.1 Test of Frame Structure

The first level in the bad rate determination algorithm consists of checking the structure of the received frames. The permissible bit combinations for each encoding type are defined in Section 8. All bits in the frame are verified by the decoder and any undefined combination is considered as an indication of a frame with corrupted rate information. Such a frame is subsequently declared and processed as an erasure.

For example, the bit allocation in some encoding types contains some unused bits that are always set to 0 by the encoder. Whenever one of these unused bits is found to be equal to 1, the frame is

1 considered as a bad frame and is processed as a frame erasure. The decoder also checks for some  
2 invalid bit combinations.

### 3 4 **6.7.2 CNG Frames**

5  
6 The following bad rate determination and concealment procedure applies only to CNG-ER encoding  
7 type. For CNG-QR encoding type, on the other hand, the test of frame structure is sufficient to detect  
8 most bad rate frames and no further verification is performed.

9  
10 Bad rate determination for CNG-ER frames is to verify that a received frame labeled as an eighth rate  
11 frame is indeed a CNG-ER frame and it is further based on the ISF ordering and on the variation of  
12 the quantized energy per sample. The determination and handling procedures are described in the  
13 following two sections.

#### 14 **6.7.2.1 Test of the ISF Ordering**

15  
16 A basic property of the ISF parameters is that they are naturally ordered, that is  $ISF[0] < ISF[1] < \dots <$   
17  $ISF[14]$ . This property can be used to check whether the decoded ISF vector corresponds to a valid  
18 predictor.

- 19  
20 1. Detection Method: The frame is considered a bad rate frame and consequently declared as  
21 an erasure when the following condition on the decoded ISFs is not verified:

$$22 \quad ISF[n] > ISF[n-1] - 85.0, \text{ for } n=1 \text{ to } 14 \quad (6.7.2.1-1)$$

- 23  
24  
25 2. Concealment Method: When a frame is declared as a bad rate frame by the above  
26 procedure, it is processed as an erasure. Note that the bad rate determination and  
27 concealment procedure based on the ISF ordering for CNG-ER frames is subject to the  
28 selection of the compilation option BRH\_LEVEL2 in VMR-WB C simulation.

#### 29 **6.7.2.2 Test of the Variation of the Quantized Energy**

30  
31 For CNG-ER frames with correct ISF ordering, a second verification is done on the variation of the  
32 quantized energy per sample in the linear domain  $\hat{E}_s$  (Equation 6.6-2). During the verification process,  
33 the CNG parameters (energy per sample and ISP vector) are already decoded and their old values  
34 (last correctly received CNG parameters) are also temporarily stored.

- 35  
36 1. Detection Method: The received frame is considered as a bad rate frame when

$$37 \quad \hat{E}_s > 60.0 * \hat{E}_s^{\text{old}}, \quad (6.7.2.2-1)$$

38  
39 where  $\hat{E}_s^{\text{old}}$  is the old quantized energy per sample in the linear domain.

- 40  
41  
42 2. Concealment Method: When a frame is declared as a bad rate frame by the above  
43 procedure, the decoded CNG parameters (energy per sample and ISP vector) are replaced  
44 by their old value, and the frame is processed as a normal CNG frame.

45  
46 The above procedure is not performed in the following cases:

- 47  
48
  - For the first CNG frame received by the decoder (initialization phase)
  - When the previous frame was already a CNG frame (regardless of its bit rate)
  - When the previous frame was already classified as a bad rate frame in order to prevent consecutive false alarms.

### 49 50 51 52 **6.7.3 Active Speech Frames**

The bad rate determination is performed only on uncorrupted speech frames (i.e., not during frame erasures). There are three methods of bad rate determination for speech frames:

1. Detection based on the ISF ordering
2. Detection by testing the LP gain against the fixed-codebook gain
3. Detection by testing the synthesis energy against the transmitted FER energy information

The detailed bad rate determination procedures for those three methods and the corresponding bad rate concealment procedures are described in the following three sections.

#### 6.7.3.1 Test of the ISF Ordering

1. Detection Method: The frame is considered as a bad rate frame and consequently declared as an erasure when the following condition on the decoded ISFs is not satisfied:

$$\text{ISF}[n] > \text{ISF}[n-1]-60.0, \text{ for } n=1 \text{ to } 14 \quad (6.7.3.1-1)$$

2. Concealment Method: When a frame is declared as a bad rate frame by the above procedure, it is processed as an erasure. Note that the bad rate determination and concealment procedure based on the ISF ordering for active speech frames is subject to the selection of compilation option BRH\_LEVEL2 in the VMR-WB C simulation.

#### 6.7.3.2 Test of the LP Gain against the Fixed-Codebook Gain

This bad rate determination and concealment procedure is performed once per subframe during the decoding process after the decoding of the gains but before the computation of the global excitation. The following procedure is performed if at least 30 good (i.e., not erased) voiced speech frames have been received since this mechanism has detected the last bad rate frame.

This mechanism is based on the following two parameters:

1. The energy  $E_{lp}$  of the impulse response of the interpolated LP speech synthesis filter with quantized coefficients  $H(z)$  for the current subframe.
2. The gain of the fixed codebook  $\hat{g}_c$  normalized by the energy of the fixed-codebook and by the square root of a long-term estimation of the energy  $Ener_{lp}$ .

$Ener_{lp}$  is measured on voiced speech frames only and is relative to the nominal level of -26 dBov. It is initialized with a value of 0.1 corresponding to -16 dBov. It is updated at the end of each good voiced speech frame (i.e., it is not updated during CNG, bad or erased frames nor during unvoiced speech frames) using the following formula:

$$Ener_{lp} = 0.95 * OldEner_{lp} + 0.05 * FrameEner \quad (6.7.3.2-1)$$

where  $FrameEner$  is the mean energy per sample of the synthesis speech signal relative to the nominal level of -26 dBov (i.e., the sum of the squares of the reconstructed speech signal samples over the frame length, divided by the frame length and by the square of 1642.0). The value of  $Ener_{lp}$  is limited to be always above 0.1 corresponding to -36 dBov.

1. Detection Method: When the LPC gain  $E_{lp}$  is above 36, the frame is declared as a bad rate frame. Otherwise, the normalized fixed codebook gain is compared to a detection threshold taken from Table 6.7-2 depending on the encoding type as shown in Table 6.7-1. When the normalized fixed-codebook gain is above that threshold, the frame is declared as a bad rate frame.
2. Concealment Method: When a frame is declared as a bad rate frame, the following modifications are applied to the decoded parameters. First, the decoded fixed-codebook gain is multiplied by 0.1. For unvoiced encoding types, the resulting gain is also limited to a

1 maximum value of 2500. For all other encoding types, the decoded pitch gain is multiplied by  
 2 0.25. Finally, the coefficients of the interpolated predictor  $A(z)$  are replaced by the coefficients  
 3 of  $A'(z) = A(z/\gamma)$  with  $\gamma=0.80$ .

4  
 5 Note that the following subframes of the current frame are also considered as bad rate frames, and  
 6 that the same modifications are applied to the decoded parameters.

7  
 8  
 9  
 10 **Table 6.7-1: Threshold selection as a function of the encoding type for the Bad Rate**  
 11 **Determination algorithm**

Encoding Type	Threshold
Voiced HR	$T_{\text{Voiced}}$
Unvoiced QR, Unvoiced HR	$T_{\text{Unvoiced}}$
All other encoding types	$T_{\text{Other}}$

12  
 13 **Table 6.7-2: Detection thresholds for the ratio between the LPC gain and the normalized fixed-**  
 14 **codebook gain**

LPC Gain	$T_{\text{Voiced}}$	$T_{\text{Unvoiced}}$	$T_{\text{Other}}$
[0,2]	1594.633545	1077.620972	6929.179688
[2,4]	2012.288574	1603.834839	6918.305176
[4,6]	2012.288574	1719.556519	6915.586426
[6,8]	1962.630127	1883.688232	5165.335938
[8,10]	2608.796387	1883.688232	3597.000977
[10,12]	1842.001221	1883.687744	3198.981934
[12,14]	1262.718506	1459.894287	2306.064453
[14,16]	1078.688721	953.719238	2026.872803
[16,18]	651.626831	827.175537	1165.432739
[18,20]	445.751007	770.602051	942.135376
[20,22]	314.417175	935.903442	899.617188
[22,24]	281.579102	935.903442	1076.359131
[24,26]	198.705811	707.029358	848.216797
[26,28]	137.482468	488.619415	340.947937
[28,30]	96.404831	227.806274	168.581619
[30,32]	68.919113	154.388351	106.822678
[32,34]	62.047684	57.347088	71.641693
[34,36]	45.933205	33.086773	50.357224
[36,38]	39.077591	27.021694	34.045952

15  
 16 **6.7.3.3 Test of the Synthesis Energy against the FER Energy Information**

17  
 18 In the Generic FR, 14 bits are used to send supplementary information that improves frame erasure  
 19 concealment and the convergence and recovery of the decoder after erased frames (see Section  
 20 5.22). These parameters include, among others, the quantized energy of the synthesis signal  $E_q$  and  
 21 signal classification information. These parameters are also used to detect bad Generic FR frames.

22  
 23 This operation is performed only on Generic FR frames (for which the quantized synthesis energy  
 24 and the information on signal classification is transmitted as supplementary information). It is not  
 25 performed when the previous frame was an erasure.  
 26

1 The energy  $E_s$  of the decoded synthesized signal is computed after the synthesis filtering operation.  
 2 As it was mentioned in Section 5.22.2.1, the way the energy  $E$  is computed at the encoder depends  
 3 on the signal classification information. The energy  $E_s$  is computed the same way using the decoded  
 4 classification information.

- 5  
 6 1. Detection Method: The frame is declared as a bad rate frame when the square root of the  
 7 ratio between the quantized energy  $E_q$  and the energy of the synthesis signal  $E_s$  is less  
 8 than 0.1:  
 9

$$10 \quad \sqrt{\frac{E_q}{E_s}} \leq 0.1 \quad (6.7.3.3-1)$$

- 11  
 12 2. Concealment Method: When a frame is declared as a bad rate frame by the above  
 13 procedure, the synthesized speech signal is attenuated by a factor of 0.1 using the  
 14 energy control procedure described in Section 6.5.5. The following gains are used for the  
 15 energy control:  
 16

$$17 \quad g_0 = g_1 = 0.1. \quad (6.7.3.3-2)$$

18  
 19 The second synthesis operation, also described in Section 6.5.5, is done with modified linear  
 20 predictors. For each subframe, the coefficients of the interpolated predictors  $A(z)$  are replaced by the  
 21 coefficients of  $A^\gamma(z) = A(z/\gamma)$  with  $\gamma=0.80$ .  
 22

23 Note that this procedure not only detects bad rate frames when they occur. It can also detect  
 24 previously overlooked bad rate frames. This is particularly important when an undetected bad rate  
 25 frame causes a high-energy error. In that case, the bad rate determination procedure based on the  
 26 synthesis energy can detect the mismatch between the energy of the synthesized signal  $E_s$  and the  
 27 quantized energy  $E_q$  in the frames following the overlooked bad rate frame. The corresponding bad  
 28 rate handling procedure thus limits error propagation and improves the convergence of the decoder.  
 29  
 30  
 31

1

## 2 7 INTEROPERABLE INTERCONNECTION BETWEEN VMR-WB AND AMR-WB 3 CODECS IN CIRCUIT-SWITCHED NETWORKS

4

5 The AMR-WB interoperable mode of VMR-WB enables encoding and decoding compatibility between  
6 VMR-WB and AMR-WB (G.722.2). However, due to differences between the frame formats and other  
7 system specific requirements and dependencies in VMR-WB and AMR-WB codecs, interworking  
8 functions that reside in an intermediate gateway are required between the two codecs to enable bi-  
9 directional interoperability. Note that the interworking functions operate at bit-stream level and have  
10 minimal non-computational logic and are much simpler and more efficient than transcoding between  
11 the two codecs. Also, they do not result in speech quality degradation under nominal conditions.

12

13 In the following sections the forward link interworking function (i.e., AMR-WB to VMR-WB) and  
14 reverse-link interworking function (i.e., VMR-WB to AMR-WB) will be described. These interworking  
15 functions will ensure compliance with the requirements of the native systems is maintained  
16 throughout an interoperable interconnection.

17

18 The establishment of interoperable interconnection between AMR-WB and VMR-WB in packet-  
19 switched networks is described in Section 9.3 of [29].

### 20 7.1 VMR-WB to AMR-WB Interconnection (Reverse Link)

21

22 **Routine Name:** `vmr2amr`

23

**Inputs:**

24

- The speech data packets generated by VMR-WB encoder

25

**Outputs:**

26

- The speech data packet compatible with AMR-WB frame format

27

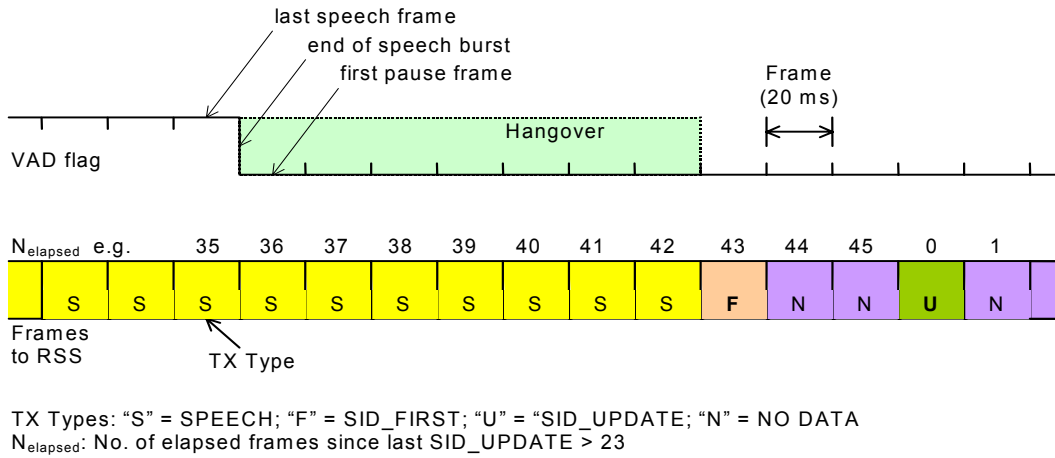
**Initialization:**

28

- The internal buffers are reset at initialization.

29

30 The source-controlled rate operation (i.e., the VAD/DTX/CNG mechanism) in AMR-WB is shown in  
31 Figure 7.1-1 and can be described as follows. Upon termination of an active speech interval, seven  
32 background noise frames are encoded as speech frames with the VAD flag set to zero (i.e., DTX  
33 hangover). A SID\_FIRST frame is then transmitted. In the SID\_FIRST frame the signal is not  
34 encoded and CNG parameters are derived from the DTX hangover (i.e., the previous 7 speech  
35 frames) at the decoder. Note that AMR-WB does not use DTX hangover after active speech periods  
36 that are shorter than 24 frames in order to reduce the DTX hangover overhead. After an SID\_FIRST  
37 frame, two frames are sent as NO\_DATA frames, followed by a SID\_UPDATE frame encoded at 1.75  
38 kbps (i.e., corresponding to 35 bits/frame). After that, seven NO\_DATA frames are transmitted  
39 followed by a SID\_UPDATE frame and so on. This procedure continues until an active speech frame  
40 is detected (VAD\_flag=1) [16].



**Figure 7.1-1: Normal VAD/DTX/CNG procedure for AMR-WB**

The VAD in the VMR-WB codec uses VAD hangover as long as necessary for preserving unvoiced stops and the CNG encoder is used whenever the VAD\_flag=0 to encode silence intervals. The VMR-WB encoder does not use DTX hangover, instead, the first background noise frame after an active speech interval is encoded at 1.75 kbps and transmitted using CNG-QR frame type followed by two frames encoded at Eighth-Rate followed by another frame at 1.75 kbps (i.e., CNG-QR). After that, seven frames are transmitted at Eighth-Rate followed by one CNG-QR frame and so on. This procedure roughly simulates the AMR-WB DTX operation with the exception that no DTX hangover is used in order to reduce the ADR of VMR-WB codec and thus reducing the capacity impact of the AMR-WB interoperable mode in the reverse link.

When signaling is requested by the cdma2000 system (e.g., dim and burst), the I-HR encoding type is used. This is to avoid declaring the speech frame as a lost frame. The I-HR encoding type consists of encoding the frame as an Interoperable Full-Rate frame then dropping the extra bits corresponding to the algebraic codebook indices in order to fit the packet size to that of CDMA Rate-Set II Half-Rate (e.g., 142 bits per frame in 12.65 kbps Interoperable Full-Rate). In this case, the bit rate is reduced to the CDMA Rate-Set II Half-Rate frame size. Note that I-HR encoding types (i.e., 12.65, 8.85, and 6.60 kbps) are generated directly on bitstream level from the corresponding I-FR encoding types. Consequently, each I-HR encoding type is used in the same way for dim-and-burst signaling (i.e. when the signaling request is sent to the encoder) and for packet-level signaling (i.e. when Full-Rate frames have to be converted to Half-Rate frames at the base station to accommodate signaling information). Figure 7.1-2 illustrates the techniques that have been used for efficient interoperability between VMR-WB and AMR-WB.

In VMR-WB → AMR-WB scenario, the speech frames are encoded in the AMR-WB interoperable mode of the VMR-WB encoder, which uses one of the following possible bit rates: (12.65, 8.85, or 6.60 kbps)/I-FR for active speech frames, (12.65, 8.85, or 6.60 kbps)/I-HR in case of dim-and-burst signaling, CNG-QR to encode silence intervals and background noise frames (one out of eight background noise frame as described above), and CNG-ER frames for most background noise frames (background noise frames not encoded as CNG-QR encoding type). The interworking function shall perform the following procedures:

- Invalid frames are transmitted to the AMR-WB decoder as erased frames (SPEECH\_LOST or NO\_DATA frame).
- I-FR encoding types are transmitted to AMR-WB decoder as 12.65, 8.85, or 6.60 kbps AMR-WB frames by simply discarding the first byte containing VMR-WB frame identifier. Also the padding bits at the end of the 8.85 and 6.60 I-FR frames shall be discarded. Note that appropriate AMR-WB frame type information is embedded in VMR-WB Interoperable Full-Rate frame structure (see Section 8).

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- CNG-QR frames are transmitted to the AMR-WB decoder as SID\_UPDATE frames by discarding unused bits at the end.
- CNG-ER frames are transmitted to AMR-WB decoder as NO\_DATA frames.
- I-HR encoding types are translated to 12.65, 8.85, or 6.60 kbps frames by generating the missing bits of the algebraic codebook indices. The bits are generated randomly. The first byte containing VMR-WB frame identifier shall also be discarded. The frame identifier bits are used to distinguish different half rate encoding types in the VMR-WB codec. Note that appropriate AMR-WB frame type information is embedded in VMR-WB Interoperable Half-Rate frame structure (see Section 8).

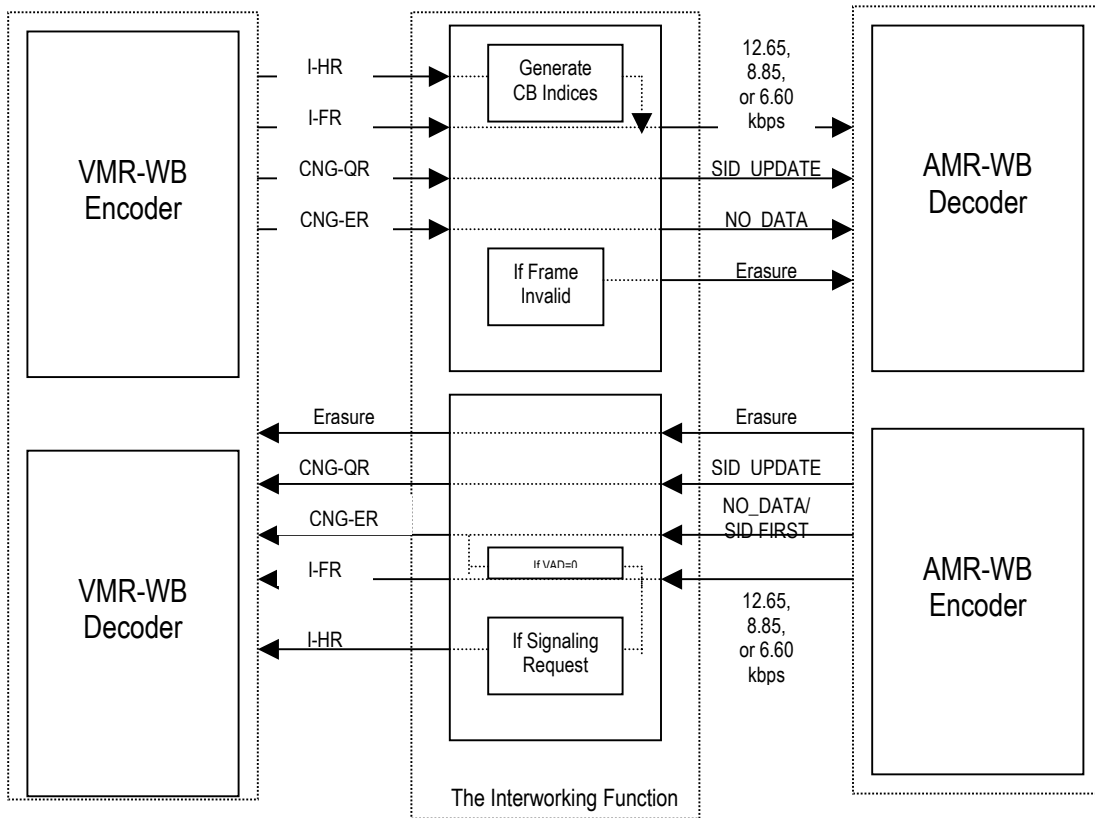


Figure 7.1-2: Interoperable interconnection between AMR-WB and VMR-WB.

7.2 AMR-WB to VMR-WB Interconnection (Forward Link)

**Routine Name:** amr2vmr

**Inputs:**

- The speech data packets from AMR-WB encoder

**Outputs:**

- The speech data packet compatible with VMR-WB frame format

**Initialization:**

- The internal buffers are reset at initialization.

1 In this scenario, some limitations are imposed by the AMR-WB DTX operation on the interworking  
 2 function. During active speech, the 1<sup>st</sup> speech data bit in the incoming AMR-WB bit-stream indicates  
 3 VAD\_flag=0 for DTX hangover period and VAD\_flag=1 for active speech. Therefore, the forward link  
 4 interworking function shall perform the following:

- 5
- 6 • SID\_UPDATE frames are forwarded to VMR-WB as CNG-QR frames.
- 7
- 8 • SID\_FIRST frames and NO\_DATA frames are forwarded as CNG-ER frames with a special bit  
 9 pattern indicating to VMR-WB decoder that the frame corresponds to an AMR-WB NO\_DATA  
 10 frame (see Section 8.4).
- 11
- 12 • Erased frames (speech lost) are forwarded as ER erasure frames indicated by an illegal bit  
 13 pattern.
- 14
- 15 • The first FR frame after active speech with VAD\_flag=0 is preserved as FR frame but the  
 16 following FR frames with VAD\_flag=0 are forwarded to VMR-WB as CNG-ER (NO\_DATA)  
 17 frames. This logic is, however, utilized only after at least one valid AMR-WB NO\_DATA,  
 18 SID\_FIRST or SID\_UPDATE frame has been received from AMR-WB. In this way frame blanking  
 19 is avoided if AMR WB is not operating in DTX mode.
- 20
- 21 • If the interworking function is requested to perform signaling operation (e.g., dim and burst  
 22 signaling) while receiving FR frames from AMR-WB encoder, then the frame is converted into an  
 23 I-HR frame. This consists of dropping the bits not fitting into CDMA HR frame corresponding to  
 24 algebraic codebook indices and adding a byte (i.e., VMR-WB frame identifier) in the beginning of  
 25 the bitstream with an appropriate VMR-WB I-HR encoding type identifier (see Figure 7.1-2 and  
 26 Section 8.2).
- 27

28 In the VMR-WB decoder, when CNG-ER frames with AMR-WB NO\_DATA bit pattern are detected,  
 29 they are processed by the CNG decoder using the last correctly received CNG parameters. The  
 30 exception is in the case of the first received AMR-WB NO\_DATA frame after an active speech  
 31 interval. Since the first frame with VAD\_flag=0 is transmitted as Full-Rate, the parameters from this  
 32 frame as well as smoothed last CNG parameters are used to initialize CNG operation:

$$33 \quad \bar{p} = 0.8\bar{p} + 0.2p_{-1}, \quad (7.2-1)$$

34 where  $\bar{p}$  denotes the smoothed CNG parameters (i.e., ISFs and energy) and  $p_{-1}$  is the corresponding  
 35 parameter from the previous Full-Rate frame. The energy information of the previous Full-Rate frame  
 36 is estimated using the memory of the ACELP adaptive codebook. AMR-WB NO\_DATA frames are  
 37 signaled to the VMR-WB decoder by setting the CNG-ER three ISF indices to 0, 0 and 1,  
 38 respectively.

39

40 Note that by default, 12.65 kbps Interoperable Full-Rate encoding type is used by VMR-WB encoder.  
 41 However, 8.85 or 6.60 kbps modes can be used by AMR-WB in compliance with the GSM link  
 42 adaptation mechanism that requires the use of lower rates in case of bad channel conditions. In this  
 43 case, the VMR-WB codec should be able to encode or decode the lower rates of AMR-WB. For this  
 44 purpose, special bit patterns are reserved in the frame structure of the VMR-WB Interoperable Full-  
 45 Rate encoding types to recognize the AMR-WB codec modes 12.65, 8.85, and 6.60 kbps from each  
 46 other. As shown in Section 8.1, the first byte of the VMR-WB Interoperable Full-Rate frame is the  
 47 same for all three encoding types (i.e., the frame identifier), then speech data bits that have been  
 48 arranged according to AMR-WB IF2 frame structure will follow. Therefore, it is sufficient to look into  
 49 AMR-WB frame type (i.e., first 4 bits of the 2<sup>nd</sup> byte) to differentiate among the three AMR-WB codec  
 50 modes. Similarly, three Interoperable Half-Rate encoding types are used corresponding to the three  
 51 Interoperable Full-Rate encoding types by dropping some of the bits corresponding to the algebraic  
 52 codebook indices. As shown in Section 8.2, the first byte of the frame is the same for all three  
 53 encoding types and they are differentiated by the AMR-WB frame type information. While support for

1 AMR-WB codec modes 6.60 and 8.85 kbps is included in VMR-WB, due to insufficient quality of  
2 these modes, the use of these modes for voice services is not recommended.  
3

4 It must be noted that mode switching is not allowed when VMR-WB operates in the AMR-WB  
5 interoperable mode. However, while in the AMR-WB interoperable mode (i.e., VMR-WB mode 3), the  
6 AMR-WB codec mode request can be signaled to the VMR-WB encoder by “-f” option in the C  
7 simulation. In this case, VMR-WB encoder can encode input speech using AMR-WB codec modes 0,  
8 1, and 2 corresponding to 6.60, 8.85, and 12.65 kbps, respectively.  
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## 8 VMR-WB FRAME STRUCTURE

To further simplify interoperable interconnections between VMR-WB and AMR-WB, the frame structure and bit stream of VMR-WB closely follow that of AMR-WB Interface Format 2 (IF2) [16]. This section describes the frame structure and detailed bit position corresponding to all existing encoding types of VMR-WB. In general, the frame structure of VMR-WB in various encoding schemes comprises a preamble (i.e., frame type identifier) and the speech data bits.

The speech data field in RS-II Generic Full-rate, RS-II Signaling Half-rate, and in all interoperable encoding types in Rate-Set II is formatted according to AMR-WB IF2 and follows the prioritization of bits in class A, class B, and class C as described in [16]. In the VMR-WB interoperable encoding types, the AMR-WB Frame Type (FT) field is appropriately set by the VMR-WB encoder according to the AMR-WB codec mode and the Frame Quality Indicator bit (FQI) is always set to 1 by the VMR-WB encoder.

### 8.1 Frame Structure of Full-Rate Encoding Types

The following sections describe the frame structure and the position of bits in the bit stream for various full-rate encoding schemes in VMR-WB encoder.

#### 8.1.1 Frame Structure of Generic Full-Rate in Rate-Set II

The mapping of bits for the RS-II Generic Full-Rate is shown in Table 8.1-1. To prioritize the speech data bits according to AMR-WB IF2 format, the following mapping function is used:

For  $j = 1$  to  $K-1$

$$D[j-1] = s(\text{table}_{2[j]}+1);$$

Where  $\text{table}_{2[j]}$  is given in Table 8.1-3 and  $K-1$  denotes the total number of speech data bits produced by VMR-WB codec for this encoding type ( $K=253$ ) excluding the FER protection bits. The source bits  $s(.)$  represent the encoder generated parameters in order of occurrence as shown in Table 8.1-2.

**Table 8.1-1: Mapping of bits in VMR-WB RS-II Generic Full-Rate**

Octet	Mapping of Bits in VMR-WB RS-II Generic Full-Rate							
	MSB Bit 8	Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	LSB Bit 1
1	Frame Type Identifier/ Frame Error Protection Bits Any combination of 6 bits other than 111110						Frame Error Protection Bits	
	FER[0]	FER[1]	FER[2]	FER[3]	FER[4]	FER[5]	FER[6]	FER[7]
2	Frame Error Protection Bits						Speech Data	
	FER[8]	FER[9]	FER[10]	FER[11]	FER[12]	FER[13]	D[0]	D[1]
3	Speech Data							
	D[2]	D[3]	D[4]	D[5]	D[6]	D[7]	D[8]	D[9]
4-33	Speech Data							
	D[10]	D[11]	..	..	..	..	D[248]	D[249]
34	Speech Data							
	D[250]	D[251]						

The following table shows the order of the bits produced by the speech encoder prior to reordering according to AMR-WB IF2 format. Note that the most significant bit (MSB) of each codec parameter is always used first.

1

**Table 8.1-2: Source encoder output parameters in order of occurrence**

Bits (MSB-LSB)	Description
s1	VAD-flag
s2 – s9	Index of 1st ISF sub-vector
s10 – s17	Index of 2nd ISF sub-vector
s18 – s23	Index of 3rd ISF sub-vector
s24 – s30	Index of 4th ISF sub-vector
s31 – s37	Index of 5th ISF sub-vector
s38 – s42	Index of 6th ISF sub-vector
s43 – s47	Index of 7th ISF sub-vector
Subframe 1	
s48 – s56	Adaptive codebook index
s57	LTP-filtering-flag
s58 – s66	Fixed-Codebook index for track 1
s67 – s75	Fixed-Codebook index for track 2
s76 – s84	Fixed-Codebook index for track 3
s85 – s93	Fixed-Codebook index for track 4
s94 – s100	VQ gain
Subframe 2	
s101 – s106	Adaptive codebook index (relative)
s107 – s150	Same description as s57 – s100
Subframe 3	
s151 – s203	Same description as s48 – s100
Subframe 4	
s204 – s253	Same description as s101 – s150

- 2 The following ordering table shall be read from left to right so that the first element (top left corner) of  
3 the table has index 0 and the last element (the rightmost element of the last row) has the index  $K-1$   
4 where  $K$  is the total number of speech bits in the specific encoding type. Note that in Generic FR the  
5 VAD bit s1 is replaced by FER[13].



1  
2

**Table 8.1-3: Ordering of the speech encoder bits for RS-II Generic Full-Rate:  $table_2(j)$**

0	4	6	93	143	196	246	7	5	3
47	48	49	50	51	150	151	152	153	154
94	144	197	247	99	149	202	252	96	146
199	249	97	147	200	250	100	203	98	148
201	251	95	145	198	248	52	2	1	101
204	155	19	21	12	17	18	20	16	25
13	10	14	24	23	22	26	8	15	53
156	31	102	205	9	33	11	103	206	54
157	28	27	104	207	34	35	29	46	32
30	55	158	37	36	39	38	40	105	208
41	42	43	44	45	56	106	159	209	57
66	75	84	107	116	125	134	160	169	178
187	210	219	228	237	58	108	161	211	62
112	165	215	67	117	170	220	71	121	174
224	76	126	179	229	80	130	183	233	85
135	188	238	89	139	192	242	59	109	162
212	63	113	166	216	68	118	171	221	72
122	175	225	77	127	180	230	81	131	184
234	86	136	189	239	90	140	193	243	60
110	163	213	64	114	167	217	69	119	172
222	73	123	176	226	78	128	181	231	82
132	185	235	87	137	190	240	91	141	194
244	61	111	164	214	65	115	168	218	70
120	173	223	74	124	177	227	79	129	182
232	83	133	186	236	88	138	191	241	92
142	195	245							

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**8.1.2 Frame Structure of 12.65 kbps Interoperable Full-Rate in Rate-Set II**

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The mapping of bits for the 12.65 kbps Interoperable Full-Rate is shown in Table 8.1-4. To prioritize the speech data bits according to AMR-WB IF2 format, the following mapping function is used:

9  
10

For  $j = 0$  to  $K-1$

11

$$D[j] = s(table_2[j]+1);$$

12  
13  
14

Where  $table_2[j]$  is given in Table 8.1-3 and  $K$  denotes the total number of speech data bits produced by VMR-WB codec for this encoding type ( $K=253$ ). The source bits  $s(.)$  represent the encoder generated parameters in order of occurrence as shown in Table 8.1-2.

15

**Table 8.1-4: Mapping of bits in VMR-WB 12.65 kbps Interoperable Full-Rate**

Octet	Mapping of bits in VMR-WB 12.65 kbps Interoperable Full-Rate							MSB	LSB
	Bit 8	Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	
1	Frame Type Identifier								
	1	1	1	1	1	0	0	0	
2	AMR-WB Frame Type				FQI	Speech Data			
	0	0	1	0	1	D[0]	D[1]	D[2]	
3	Speech Data								
	D[3]	D[4]	D[5]	D[6]	D[7]	D[8]	D[9]	D[10]	
4-33	Speech Data								
	D[11]	D[12]	..	..	..	..	D[249]	D[250]	
34	Speech Data								
	D[251]	D[252]							

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**8.1.3 Frame Structure of 8.85 kbps Interoperable Full-Rate in Rate-Set II**

The mapping of bits for the 8.85 kbps Interoperable Full-Rate is shown in Table 8.1-5. To prioritize the speech data bits according to AMR-WB IF2 format, the following mapping function is used:

For  $j = 0$  to  $K-1$

$$D[j] = s(\text{table}_{1[j]}+1);$$

Where  $\text{table}_{1[j]}$  is given in Table 8.1-7 and  $K$  denotes the total number of speech data bits produced by VMR-WB codec for this encoding type ( $K=177$ ). The source bits  $s(.)$  represent the encoder generated parameters in order of occurrence as shown in Table 8.1-6. Padding bits (zeros) are added to the end of the frame to adjust the frame size to 266.

**Table 8.1-5: Mapping of bits in VMR-WB 8.85 kbps Interoperable Full-Rate**

Octet	Mapping of bits in VMR-WB 8.85 kbps Interoperable Full-Rate							MSB	LSB
	Bit 8	Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1	
1	Frame Type Identifier								
	1	1	1	1	1	0	0	0	
2	AMR-WB Frame Type				FQI	Speech Data			
	0	0	0	1	1	D[0]	D[1]	D[2]	
3	Speech Data								
	D[3]	D[4]	D[5]	D[6]	D[7]	D[8]	D[9]	D[10]	
4-23	Speech Data								
	D[11]	D[12]	..	..	..	..	D[169]	D[170]	
24	Speech Data						Padding Bits		
	D[171]	D[172]	D[173]	D[174]	D[175]	D[176]	0	0	
25-33	Padding Bits								
	0	0	..	..	..	..	0	0	
34	Padding Bits								
	0	0							

The following table shows the order of the bits produced by the speech encoder. Note that the most significant bit (MSB) of each codec parameter is always used first.

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**Table 8.1-6: Source encoder output parameters in order of occurrence**

Bits (MSB-LSB)	Description
s1	VAD-flag
s2 – s9	Index of 1st ISF sub-vector
s10 – s17	Index of 2nd ISF sub-vector
s18 – s23	Index of 3rd ISF sub-vector
s24 – s30	Index of 4th ISF sub-vector
s31 – s37	Index of 5th ISF sub-vector
s38 – s42	Index of 6th ISF sub-vector
s43 – s47	Index of 7th ISF sub-vector
Subframe 1	
s48 – s55	Adaptive codebook index
s56 – s60	Fixed-Codebook index for track 1
s61 – s65	Fixed-Codebook index for track 2
s66 – s70	Fixed-codebook index for track 3
s71 – s75	Fixed-Codebook index for track 4
s76 – s81	VQ gain
Subframe 2	
s82 – s86	Adaptive codebook index (relative)
s87 – s112	Same description as s56 – s81
Subframe 3	
s113 – s146	Same description as s48 – s81
Subframe 4	
s147 – s177	Same description as s82 – s112

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The following ordering table shall be read from left to right so that the first element (top left corner) of the table has index 0 and the last element (the rightmost element of the last row) has the index  $K-1$  where  $K$  is the total number of speech bits in the specific encoding type.

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**Table 8.1-7: Ordering of the speech encoder bits for 8.85 kbps Interoperable Full-Rate:  $table_1(j)$**

0	4	6	7	5	3	47	48	49	112
113	114	75	106	140	171	80	111	145	176
77	108	142	173	78	109	143	174	79	110
144	175	76	107	141	172	50	115	51	2
1	81	116	146	19	21	12	17	18	20
16	25	13	10	14	24	23	22	26	8
15	52	117	31	82	147	9	33	11	83
148	53	118	28	27	84	149	34	35	29
46	32	30	54	119	37	36	39	38	40
85	150	41	42	43	44	45	55	60	65
70	86	91	96	101	120	125	130	135	151
156	161	166	56	87	121	152	61	92	126
157	66	97	131	162	71	102	136	167	57
88	122	153	62	93	127	158	67	98	132
163	72	103	137	168	58	89	123	154	63
94	128	159	68	99	133	164	73	104	138
169	59	90	124	155	64	95	129	160	69
100	134	165	74	105	139	170			

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**8.1.4 Frame Structure of 6.60 kbps Interoperable Full-Rate in Rate-Set II**

The mapping of bits for the 6.60 kbps Interoperable Full-Rate is shown in Table 8.1-8. To prioritize the speech data bits according to AMR-WB IF2 format, the following mapping function is used:

For  $j = 0$  to  $K-1$

1  $D[j] = s(\text{table}_{o[j]}+1);$

2 Where  $\text{table}_{o[j]}$  is given in Table 8.1-10 and  $K$  denotes the total number of speech data bits produced  
 3 by VMR-WB codec for this encoding type ( $K=132$ ). The source bits  $s(.)$  represent the encoder  
 4 generated parameters in order of occurrence as shown in Table 8.1-9. Padding bits (zeros) shall be  
 5 added to the end of the frame to adjust the frame size to 266.

6 **Table 8.1-8: Mapping of bits in VMR-WB 6.60 kbps Interoperable Full-Rate**

Octet	MSB	Mapping of bits in VMR-WB 6.60 kbps Interoperable Full-Rate						LSB
	Bit 8	Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1
1	Frame Type Identifier							
	1	1	1	1	1	0	0	0
2	AMR-WB Frame Type				FQI	Speech Data		
	0	0	0	0	1	D[0]	D[1]	D[2]
3	Speech Data							
	D[3]	D[4]	D[5]	D[6]	D[7]	D[8]	D[9]	D[10]
4-18	Speech Data							
	D[11]	D[12]	..	..	..	..	D[129]	D[130]
19	Speech Data	Padding Bits						
	D[131]	0	0	0	0	0	0	0
20-33	Padding Bits							
	0	0	..	..	..	..	0	0
34	Padding Bits							
	0	0						

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 8 The following table shows the order of the bits produced by the speech encoder. Note that the most  
 9 significant bit (MSB) of each codec parameter is always used first.

10 **Table 8.1-9: Source encoder output parameters in order of occurrence**

Bits (MSB-LSB)	Description
s1	VAD-flag
s2 – s9	Index of 1st ISF sub-vector
s10 – s17	Index of 2nd ISF sub-vector
s18 – s24	Index of 3rd ISF sub-vector
s25 – s31	Index of 4th ISF sub-vector
s32 – s37	Index of 5th ISF sub-vector
Subframe 1	
s38 – s45	Adaptive codebook index
s46 - 57	Fixed-Codebook Index
s58 – s63	VQ gain
Subframe 2	
s64 – s68	Adaptive codebook index (relative)
s69 – s86	Same description as s46 – s63
Subframe 3	
s87 – s109	Same description as s64 – s86
Subframe 4	
s110 – s132	Same description as s64 – s86

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 12 The following ordering table shall be read from left to right so that the first element (top left corner) of  
 13 the table has index 0 and the last element (the rightmost element of the last row) has the index  $K-1$   
 14 where  $K$  is the total number of speech bits in the specific encoding type.  
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**Table 8.1-10: Ordering of the speech encoder bits for 6.60 kbps Interoperable Full-Rate:**  
*table<sub>0</sub>(j)*

0	5	6	7	61	84	107	130	62	85
8	4	37	38	39	40	58	81	104	127
60	83	106	129	108	131	128	41	42	80
126	1	3	57	103	82	105	59	2	63
109	110	86	19	22	23	64	87	18	20
21	17	13	88	43	89	65	111	14	24
25	26	27	28	15	16	44	90	66	112
9	11	10	12	67	113	29	30	31	32
34	33	35	36	45	51	68	74	91	97
114	120	46	69	92	115	52	75	98	121
47	70	93	116	53	76	99	122	48	71
94	117	54	77	100	123	49	72	95	118
55	78	101	124	50	73	96	119	56	79
102	125								

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**8.1.5 Frame Structure of Generic Full-Rate in Rate-Set I**

The mapping of bits for the RS-I Generic Full-Rate is shown in Table 8.1-11.

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**Table 8.1-11: Mapping of bits in VMR-WB RS-I Generic Full-Rate**

Octet	MSB	Mapping of bits in VMR-WB RS-I Generic Full-Rate						LSB
	Bit 8	Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1
1	Identifier	ISFs						
	0	ISF[23]	ISF[24]	ISF[0]	ISF[1]	ISF[2]	ISF[3]	ISF[4]
2	ISFs							
	ISF[5]	ISF[6]	ISF[7]	ISF[8]	ISF[9]	ISF[10]	ISF[11]	ISF[12]
3	ISFs							
	ISF[13]	ISF[14]	ISF[15]	ISF[16]	ISF[17]	ISF[18]	ISF[19]	ISF[20]
4	ISFs		Pitch Delay for Sub-Frame 1					
	ISF[21]	ISF[22]	P[0]	P[1]	P[2]	P[3]	P[4]	P[5]
5	Gains for Sub-Frame 1							
	P[6]	P[7]	G[0]	G[1]	G[2]	G[3]	G[4]	G[5]
6	Pitch Delay for Sub-Frame 2							
	G[6]	P[0]	P[1]	P[2]	P[3]	P[4]	G[0]	G[1]
7	Gains for Sub-Frame 2					Pitch Delay for Sub-Frame 3		
	G[2]	G[3]	G[4]	G[5]	G[6]	P[0]	P[1]	P[2]
8	Pitch Delay for Sub-Frame 3					Gains for Sub-Frame 3		
	P[3]	P[4]	P[5]	P[6]	P[7]	G[0]	G[1]	G[2]
9	Gains for Sub-Frame 3				Pitch Delay for Sub-Frame 4			
	G[3]	G[4]	G[5]	G[6]	P[0]	P[1]	P[2]	P[3]
10	Gains for Sub-Frame 4							
	P[4]	G[0]	G[1]	G[2]	G[3]	G[4]	G[5]	G[6]
11	FCB Indices for Sub-Frame 1							
	FCB[0]	FCB[1]	FCB[2]	FCB[3]	FCB[4]	FCB[5]	FCB[6]	FCB[7]
12	FCB Indices for Sub-Frame 1							
	FCB[8]	FCB[9]	FCB[10]	FCB[11]	FCB[12]	FCB[13]	FCB[14]	FCB[15]
13	FCB Indices for Sub-Frame 1				FCB Indices for Sub-Frame 2			
	FCB[16]	FCB[17]	FCB[18]	FCB[19]	FCB[0]	FCB[1]	FCB[2]	FCB[3]

14	FCB Indices for Sub-Frame 2							
	FCB[4]	FCB[5]	FCB[6]	FCB[7]	FCB[8]	FCB[9]	FCB[10]	FCB[11]
15	FCB Indices for Sub-Frame 2							
	FCB[12]	FCB[13]	FCB[14]	FCB[15]	FCB[16]	FCB[17]	FCB[18]	FCB[19]
16	FCB Indices for Sub-Frame 3							
	FCB[0]	FCB[1]	FCB[2]	FCB[3]	FCB[4]	FCB[5]	FCB[6]	FCB[7]
17	FCB Indices for Sub-Frame 3							
	FCB[8]	FCB[9]	FCB[10]	FCB[11]	FCB[12]	FCB[13]	FCB[14]	FCB[15]
18	FCB Indices for Sub-Frame 3				FCB Indices for Sub-Frame 4			
	FCB[16]	FCB[17]	FCB[18]	FCB[19]	FCB[0]	FCB[1]	FCB[2]	FCB[3]
19	FCB Indices for Sub-Frame 4							
	FCB[4]	FCB[5]	FCB[6]	FCB[7]	FCB[8]	FCB[9]	FCB[10]	FCB[11]
20	FCB Indices for Sub-Frame 4							
	FCB[12]	FCB[13]	FCB[14]	FCB[15]	FCB[16]	FCB[17]	FCB[18]	
21	ISFs							
	ISF[25]	ISF[26]	ISF[27]	ISF[28]	ISF[29]	ISF[30]	ISF[31]	ISF[32]
22	ISFs							
	ISF[33]	ISF[34]	ISF[35]					

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**8.1.6 Frame Structure of Voiced Full-Rate in Rate-Set I**

The mapping of bits for the RS-I Voiced Full-Rate is shown in Table 8.1-12.

**Table 8.1-12: Mapping of bits in VMR-WB RS-I Voiced Full-Rate**

Octet	Mapping of bits in VMR-WB RS-I Voiced Full-Rate								
	MSB	Bit 8	Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	LSB
1	Identifier		Unused	ISFs					
	1	0	0	ISF[0]	ISF[1]	ISF[2]	ISF[3]	ISF[4]	
2	ISFs								
	ISF[5]	ISF[6]	ISF[7]	ISF[8]	ISF[9]	ISF[10]	ISF[11]	ISF[12]	
3	ISFs								
	ISF[13]	ISF[14]	ISF[15]	ISF[16]	ISF[17]	ISF[18]	ISF[19]	ISF[20]	
4	ISFs								
	ISF[21]	ISF[22]	ISF[23]	ISF[24]	ISF[25]	ISF[26]	ISF[27]	ISF[28]	
5	ISFs								
	ISF[29]	ISF[30]	ISF[31]	ISF[32]	ISF[33]	ISF[34]	ISF[35]	P[0]	
6	Pitch Delay								
	P[1]	P[2]	P[3]	P[4]	P[5]	P[6]	P[7]	P[8]	
7	Gains for Sub-Frame 1								
	G[0]	G[1]	G[2]	G[3]	G[4]	G[5]	G[6]	G[7]	
8	Gains for Sub-Frame 2								
	G[0]	G[1]	G[2]	G[3]	G[4]	G[5]	G[6]	G[7]	
9	Gains for Sub-Frame 3								
	G[0]	G[1]	G[2]	G[3]	G[4]	G[5]	G[6]	G[7]	
10	Gains for Sub-Frame 4								
	G[0]	G[1]	G[2]	G[3]	G[4]	G[5]	G[6]	G[7]	
11	FCB Indices for Sub-Frame 1								
	FCB[0]	FCB[0]	FCB[0]	FCB[0]	FCB[0]	FCB[0]	FCB[0]	FCB[0]	
12	FCB Indices for Sub-Frame 1								
	FCB[8]	FCB[8]	FCB[8]	FCB[8]	FCB[8]	FCB[8]	FCB[8]	FCB[8]	
13	FCB Indices for Sub-Frame 1				FCB Indices for Sub-Frame 1				
	FCB[16]	FCB[16]	FCB[16]	FCB[16]	FCB[16]	FCB[16]	FCB[16]	FCB[16]	
14	FCB Indices for Sub-Frame 2								
	FCB[4]	FCB[4]	FCB[4]	FCB[4]	FCB[4]	FCB[4]	FCB[4]	FCB[4]	
15	FCB Indices for Sub-Frame 2								
	FCB[12]	FCB[12]	FCB[12]	FCB[12]	FCB[12]	FCB[12]	FCB[12]	FCB[12]	
16	FCB Indices for Sub-Frame 3								

	FCB[0]	FCB[0]	FCB[0]	FCB[0]	FCB[0]	FCB[0]	FCB[0]	FCB[0]
17	FCB Indices for Sub-Frame 3							
	FCB[8]	FCB[8]	FCB[8]	FCB[8]	FCB[8]	FCB[8]	FCB[8]	FCB[8]
18	FCB Indices for Sub-Frame 3				FCB Indices for Sub-Frame 3			
	FCB[16]	FCB[16]	FCB[16]	FCB[16]	FCB[16]	FCB[16]	FCB[16]	FCB[16]
19	FCB Indices for Sub-Frame 4							
	FCB[4]	FCB[4]	FCB[4]	FCB[4]	FCB[4]	FCB[4]	FCB[4]	FCB[4]
20	FCB Indices for Sub-Frame 4							
	FCB[12]	FCB[12]	FCB[12]	FCB[12]	FCB[12]	FCB[12]	FCB[12]	FCB[12]
21	Frame Error Protection Bits						Unused bits	
	FER[0]	FER[1]	FER[2]	FER[3]	FER[4]	FER[5]	0	0
22	Unused bits							
	0	0	0					

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2 **8.2 Frame Structure of Half-Rate Encoding Types**

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4 The following sections describe the frame structure and the position of bits in the bit stream for  
 5 various half-rate encoding schemes in VMR-WB encoder.

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7 **8.2.1 Frame Structure of Generic Half-Rate in Rate-Set II**

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9 The mapping of bits for the RS-II Generic Half-Rate is shown in Table 8.2-1.

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**Table 8.2-1: Mapping of bits in VMR-WB RS-II Generic Half-Rate**

Octet	MSB	Mapping of bits in VMR-WB RS-II Generic Half-Rate						LSB
	Bit 8	Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1
1	Identifier	ISFs						
	0	ISF[0]	ISF[1]	ISF[2]	ISF[3]	ISF[4]	ISF[5]	ISF[6]
2	ISFs							
	ISF[7]	ISF[8]	ISF[9]	ISF[10]	ISF[11]	ISF[12]	ISF[13]	ISF[14]
3	ISFs							
	ISF[15]	ISF[16]	ISF[17]	ISF[18]	ISF[19]	ISF[20]	ISF[21]	ISF[22]
4	ISFs							
	ISF[23]	ISF[24]	ISF[25]	ISF[26]	ISF[27]	ISF[28]	ISF[29]	ISF[30]
5	ISFs					Pitch Delay for Sub-Frame 1		
	ISF[31]	ISF[32]	ISF[33]	ISF[34]	ISF[35]	P[0]	P[1]	P[2]
6	Pitch Delay for Sub-Frame 1					FCB Indices for Sub-Frame 1		
	P[3]	P[4]	P[5]	P[6]	P[7]	FCB[0]	FCB[1]	FCB[2]
7	FCB Indices for Sub-Frame 1							
	FCB[3]	FCB[4]	FCB[5]	FCB[6]	FCB[7]	FCB[8]	FCB[9]	FCB[10]
8	Gain Flag		FCB Gain for Sub-Frame 1					
	FCB[11]	G_Flag	G[0]	G[1]	G[2]	G[3]	G[4]	G[5]
9	FCB Indices for Sub-Frame 2							
	FCB[0]	FCB[1]	FCB[2]	FCB[3]	FCB[4]	FCB[5]	FCB[6]	FCB[7]
10	FCB Indices for Sub-Frame 2				FCB Gain for Sub-Frame 2			
	FCB[8]	FCB[9]	FCB[10]	FCB[11]	G[0]	G[1]	G[2]	G[3]
11	Pitch Delay for Sub-Frame 3							
	G[4]	G[5]	P[0]	P[1]	P[2]	P[3]	P[4]	FCB[0]
12	FCB Indices for Sub-Frame 3							
	FCB[1]	FCB[2]	FCB[3]	FCB[4]	FCB[5]	FCB[6]	FCB[7]	FCB[8]
13	FCB Indices for Sub-Frame 3			Gain Flag		FCB Gain for Sub-Frame 3		
	FCB[9]	FCB[10]	FCB[11]	G_Flag	G[0]	G[1]	G[2]	G[3]
14	FCB Indices for Sub-Frame 4							
	G[4]	G[5]	FCB[0]	FCB[1]	FCB[2]	FCB[3]	FCB[4]	FCB[5]

15	FCB Indices for Sub-Frame 4						G[0]	G[1]
	FCB[6]	FCB[7]	FCB[8]	FCB[9]	FCB[10]	FCB[11]		
16	FCB Gain for Sub-Frame 4							
	G[2]	G[3]	G[4]	G[5]				

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**8.2.2 Frame Structure of Signaling Half-Rate in Rate-Set II**

The mapping of bits for the RS-II Signaling Half-Rate, shown in Table 8.2-2, is similar to that of RS-II Generic Full-Rate with the exception that the fixed-codebook indices exceeding the HR frame size are dropped from the end of the packet at bitstream level. The RS-II Signaling Half-Rate frame is generated during transmission from RS-II Generic FR frame to accommodate signaling information at the base station and is not generated by the VMR-WB encoder. To prioritize the speech data bits according to AMR-WB IF2 format, the following mapping function is used:

For  $j = 1$  to  $K-1$

$$D[j-1] = s(\text{table}_{2[j]}+1);$$

Where  $\text{table}_{2[j]}$  is given in Table 8.1-3 and  $K-1$  denotes the total number of speech data bits produced by VMR-WB codec for this encoding type excluding the FER protection bits ( $K=110$ ). The source bits  $s(.)$  represent the encoder generated parameters in order of occurrence as shown in Table 8.1-2.

**Table 8.2-2: Mapping of bits in VMR-WB RS-II Signaling Half-Rate**

Octet	Mapping of bits in VMR-WB RS-II Signaling Half-Rate							
	MSB Bit 8	Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	LSB Bit 1
1	Frame Type Identifier			Frame Error Protection Bits Any combination of 6 bits other than 111110				
	1	1	0	FER[0]	FER[1]	FER[2]	FER[3]	FER[4]
2	Frame Error Protection Bits Any combination of 6 bits other than 111110	Frame Error Protection Bits			Unused bits			
		FER[5]	FER[6]	FER[7]	0	0	0	D[0]
3	Speech Data							
	D[2]	D[3]	D[4]	D[5]	D[6]	D[7]	D[8]	D[9]
4-15	Speech Data							
	D[10]	D[11]	..	..	..	..	D[104]	D[105]
16	Speech Data							
	D[106]	D[107]	D[108]	D[109]				

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**8.2.3 Frame Structure of Voiced Half-Rate in Rate-Set II**

**Table 8.2-3: Mapping of bits in VMR-WB RS-II Voiced Half-Rate**

Octet	Mapping of bits in VMR-WB RS-II Voiced Half-Rate							
	MSB Bit 8	Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	LSB Bit 1
1	Frame Type Identifier			ISFs				
	1	1	1	ISF[0]	ISF[1]	ISF[2]	ISF[3]	ISF[4]
2	ISFs							
	ISF[5]	ISF[6]	ISF[7]	ISF[8]	ISF[9]	ISF[10]	ISF[11]	ISF[12]
3	ISFs							

	ISF[13]	ISF[14]	ISF[15]	ISF[16]	ISF[17]	ISF[18]	ISF[19]	ISF[20]
4	ISFs							
	ISF[21]	ISF[22]	ISF[23]	ISF[24]	ISF[25]	ISF[26]	ISF[27]	ISF[28]
5	ISFs							
	ISF[29]	ISF[30]	ISF[31]	ISF[32]	ISF[33]	ISF[34]	ISF[35]	P[0]
6	Pitch Delay							
	P[1]	P[2]	P[3]	P[4]	P[5]	P[6]	P[7]	P[8]
7	Pitch Filtering	FCB Indices for Sub-Frame 1						
	PF	FCB[0]	FCB[1]	FCB[2]	FCB[3]	FCB[4]	FCB[5]	FCB[6]
8	FCB Indices for Sub-Frame 1					Gain Flag		
	FCB[7]	FCB[8]	FCB[9]	FCB[10]	FCB[11]	G Flag	G[0]	G[1]
9	FCB Gain for Sub-Frame 1				FCB Indices for Sub-Frame 2			
	G[2]	G[3]	G[4]	G[5]	FCB[0]	FCB[1]	FCB[2]	FCB[3]
10	FCB Indices for Sub-Frame 2							
	FCB[4]	FCB[5]	FCB[6]	FCB[7]	FCB[8]	FCB[9]	FCB[10]	FCB[11]
11	FCB Gain for Sub-Frame 2						Pitch Filtering	
	G[0]	G[1]	G[2]	G[3]	G[4]	G[5]	PF	FCB[0]
12	FCB Indices for Sub-Frame 3							
	FCB[1]	FCB[2]	FCB[3]	FCB[4]	FCB[5]	FCB[6]	FCB[7]	FCB[8]
13	FCB Indices for Sub-Frame 3			Gain Flag	FCB Gain for Sub-Frame 3			
	FCB[9]	FCB[10]	FCB[11]	G Flag	G[0]	G[1]	G[2]	G[3]
14	FCB Indices for Sub-Frame 4							
	G[4]	G[5]	FCB[0]	FCB[1]	FCB[2]	FCB[3]	FCB[4]	FCB[5]
15	FCB Indices for Sub-Frame 4							
	FCB[6]	FCB[7]	FCB[8]	FCB[9]	FCB[10]	FCB[11]	G[0]	G[1]
16	FCB Gain for Sub-Frame 4							
	G[2]	G[3]	G[4]	G[5]				

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**8.2.4 Frame Structure of Unvoiced Half-Rate in Rate-Set II**

The mapping of bits for the RS-II Unvoiced Half-Rate encoding type is shown in Table 8.2-4.

**Table 8.2-4: Mapping of bits in VMR-WB RS-II Unvoiced Half-Rate**

Octet	Mapping of bits in VMR-WB RS-II Unvoiced Half-Rate								
	MSB	Bit 8	Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	LSB
1	Frame Identifier		ISFs						
	1	0	ISF[0]	ISF[1]	ISF[2]	ISF[3]	ISF[4]	ISF[5]	
2-5	ISFs								
	ISF[6]	ISF[7]	..	..	..	..	ISF[36]	ISF[37]	
6	ISFs								
	ISF[38]	ISF[39]	ISF[40]	ISF[41]	ISF[42]	ISF[43]	ISF[44]	ISF[45]	
7	FCB Indices for Sub-Frame 1								
	FCB[0]	FCB[1]	FCB[2]	FCB[3]	FCB[4]	FCB[5]	FCB[6]	FCB[7]	
8	FCB Indices for Sub-Frame 1					FCB Gain for Sub-Frame 1			
	FCB[8]	FCB[9]	FCB[10]	FCB[11]	FCB[12]	G[0]	G[1]	G[2]	
9	FCB Gain for Sub-Frame 1			FCB Indices for Sub-Frame 2					
	G[3]	G[4]	G[5]	FCB[0]	FCB[1]	FCB[2]	FCB[3]	FCB[4]	
10	FCB Indices for Sub-Frame 2								
	FCB[5]	FCB[6]	FCB[7]	FCB[8]	FCB[9]	FCB[10]	FCB[11]	FCB[12]	
11	FCB Gain for Sub-Frame 2								
	G[0]	G[1]	G[2]	G[3]	G[4]	G[5]	FCB[0]	FCB[1]	
12	FCB Indices for Sub-Frame 3								
	FCB[2]	FCB[3]	FCB[4]	FCB[5]	FCB[6]	FCB[7]	FCB[8]	FCB[9]	
13	FCB Indices for Sub-Frame 3			FCB Gain for Sub-Frame 3					
	FCB[10]	FCB[11]	FCB[12]	G[0]	G[1]	G[2]	G[3]	G[4]	
14	FCB Indices for Sub-Frame 4								
	G[5]	FCB[0]	FCB[1]	FCB[2]	FCB[3]	FCB[4]	FCB[5]	FCB[6]	

15	FCB Indices for Sub-Frame 4						G[0]	G[1]
	FCB[7]	FCB[8]	FCB[9]	FCB[10]	FCB[11]	FCB[12]		
16	FCB Gain for Sub-Frame 4							
	G[2]	G[3]	G[4]	G[5]				

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**8.2.5 Frame Structure of 12.65 kbps Interoperable Half-Rate**

The mapping of bits for the 12.65 kbps Interoperable Half-Rate, shown in Table 8.2-5, is similar to that of 12.65 kbps Interoperable Full-Rate with the exception that the fixed-codebook indices exceeding the HR frame size are dropped from the end of the packet. To prioritize the speech data bits according to AMR-WB IF2 format, the following mapping function is used:

For  $j = 0$  to  $K-1$

$$D[j] = s(\text{table}_2[j]+1);$$

where  $\text{table}_2[j]$  is given in Table 8.1-3 and  $K$  denotes the total number of speech data bits produced by the VMR-WB codec for this encoding type ( $K=111$ ). The source bits  $s(.)$  represent the encoder generated parameters, excluding those bits that correspond to the fixed-codebook indices exceeding the HR frame size, in order of occurrence as shown in Table 8.1-2.

**Table 8.2-5: Mapping of bits in VMR-WB 12.65 kbps Interoperable Half-Rate**

Octet	MSB	Mapping of bits in VMR-WB 12.65 kbps Interoperable Half-Rate						LSB
	Bit 8	Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1
1	Frame Type Identifier							
	1	1	0	1	1	1	1	1
2	AMR-WB Frame Type				FQI	Speech Data		
	0	0	1	0	1	D[0]	D[1]	D[2]
3	Speech Data							
	D[3]	D[4]	D[5]	D[6]	D[7]	D[8]	D[9]	D[10]
4-15	Speech Data							
	D[11]	D[12]	..	..	..	..	D[105]	D[106]
16	Speech Data							
	D[107]	D[108]	D[109]	D[110]				

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**8.2.6 Frame Structure of 8.85 kbps Interoperable Half-Rate**

The mapping of bits for the 8.85 kbps Interoperable Half-Rate is shown in Table 8.2-6. To prioritize the speech data bits according to AMR-WB IF2 format, the following mapping function is used:

For  $j = 0$  to  $K-1$

$$D[j] = s(\text{table}_1[j]+1);$$

where  $\text{table}_1[j]$  is given in Table 8.1-7 and  $K$  denotes the total number of speech data bits produced by the VMR-WB codec for this encoding type ( $K=111$ ). The source bits  $s(.)$  represent the encoder generated parameters, excluding those bits that correspond to the fixed-codebook indices exceeding the HR frame size, in order of occurrence as shown in Table 8.1-6.

**Table 8.2-6: Mapping of bits in VMR-WB 8.85 kbps Interoperable Half-Rate**

Octet	MSB	Mapping of bits in VMR-WB 8.85 kbps Interoperable Half-Rate						LSB
	Bit 8	Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1
1	Frame Type Identifier							
	1	1	0	1	1	1	1	1

2	AMR-WB Frame Type				FQI	Speech Data		
	0	0	0	1	1	D[0]	D[1]	D[2]
3	Speech Data							
	D[3]	D[4]	D[5]	D[6]	D[7]	D[8]	D[9]	D[10]
4-13	Speech Data							
	D[11]	D[12]	..	..	..	..	D[89]	D[90]
14	Speech Data							
	D[91]	D[92]	D[93]	D[94]	D[95]	D[96]	D[97]	D[98]
15	Speech Data							
	D[99]	D[100]	..	..	..	..	D[105]	D[106]
16	Speech Data							
	D[107]	D[108]	D[109]	D[110]				

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**8.2.7 Frame Structure of 6.60 kbps Interoperable Half-Rate**

The mapping of bits for the 6.60 kbps Interoperable Half-Rate is shown in Table 8.2-7. To prioritize the speech data bits according to AMR-WB IF2 format, the following mapping function is used:

For  $j = 0$  to  $K-1$

$$D[j] = s(\text{table}_{o[j]}+1);$$

where  $\text{table}_{o[j]}$  is given in Table 8.1-10 and  $K$  denotes the total number of speech data bits produced by the VMR-WB codec for this encoding type ( $K=111$ ). The source bits  $s(.)$  represent the encoder generated parameters, excluding those bits that correspond to the fixed-codebook indices exceeding the HR frame size, in order of occurrence as shown in Table 8.1-9.

**Table 8.2-7: Mapping of bits in VMR-WB 6.60 kbps Interoperable Half-Rate**

Octet	MSB	Mapping of bits in VMR-WB 6.60 kbps Interoperable Half-Rate						LSB
	Bit 8	Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1
1	Frame Type Identifier							
	1	1	0	1	1	1	1	1
2	AMR-WB Frame Type				FQI	Speech Data		
	0	0	0	0	1	D[0]	D[1]	D[2]
3	Speech Data							
	D[3]	D[4]	D[5]	D[6]	D[7]	D[8]	D[9]	D[10]
4-12	Speech Data							
	D[11]	D[12]	..	..	..	..	D[81]	D[82]
13	Speech Data							
	D[83]	D[84]	D[85]	D[86]	D[87]	D[88]	D[89]	D[90]
14-15	Speech Data							
	D[91]	D[92]	..	..	..	..	D[105]	D[106]
16	Speech Data							
	D[107]	D[108]	D[109]	D[110]				

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**8.2.8 Frame Structure of Unvoiced Half-Rate in Rate-Set I**

The mapping of bits for the RS-I Unvoiced Full-Rate is shown in Table 8.2-8.

**Table 8.2-8: Mapping of bits in VMR-WB RS-I Unvoiced Half-Rate**

Octet	MSB	Mapping of bits in VMR-WB RS-I Unvoiced Half-Rate						LSB
	Bit 8	Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1
1	Identifier				ISFs			
	1	1	0	0	ISF[0]	ISF[1]	ISF[2]	ISF[3]

2	ISFs							
	ISF[4]	ISF[5]	..	..	..	..	ISF[34]	ISF[35]
3	ISFs							
	ISF[36]	ISF[37]	ISF[38]	ISF[39]	ISF[40]	ISF[41]	ISF[42]	ISF[43]
4	Gains for Sub-Frame 1 and 2							
	ISF[44]	ISF[45]	G[0]	G[1]	G[2]	G[3]	G[4]	G[5]
5	Gains for Sub-Frame 1 and 2							
	G[6]	G[7]	G[8]	G[9]	G[10]	G[11]	G[12]	G[0]
6	Gains for Sub-Frame 3 and 4							
	G[1]	G[2]	G[3]	G[4]	G[5]	G[6]	G[7]	G[8]
7	Gains for Sub-Frame 3 and 4				Formant shaping	Unused bits		
	G[9]	S[0]	G[11]	G[12]	S[0]	0	0	0

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### 8.3 Frame Structure of Quarter-Rate Encoding Types in Rate-Set II

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The following sections describe the frame structure and the position of bits in the bit stream for various quarter-rate encoding schemes in VMR-WB encoder.

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#### 8.3.1 Frame Structure of CNG Quarter-Rate

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The mapping of the bits in the VMR-WB RS-II CNG Quarter-Rate is similar to that of AMR-WB SID\_UPDATE [16] and it is shown in Table 8.3-1.

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The AMR-WB mode indication bits denoted by xx in the following Table are set to '00', '01', or '10', if VMR-WB generates CNG frames for AMR-WB at 6.60, 8.85, or 12.65 kbps, respectively. By default, these bits are set to '10'.

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**Table 8.3-1: Mapping of bits in VMR-WB RS-II CNG Quarter-Rate**

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Octet	MSB	Mapping of bits VMR-WB RS-II CNG Quarter-Rate						LSB
	Bit 8	Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1
1	Frame Type Identifier				FQI	ISFs		
	1	0	0	1	1	ISF[0]	ISF[1]	ISF[2]
2	ISFs							
	ISF[3]	ISF[4]	ISF[5]	ISF[6]	ISF[7]	ISF[8]	ISF[9]	ISF[10]
3	ISFs							
	ISF[11]	ISF[12]	ISF[13]	ISF[14]	ISF[15]	ISF[16]	ISF[17]	ISF[18]
4	ISFs							
	ISF[19]	ISF[20]	ISF[21]	ISF[22]	ISF[23]	ISF[24]	ISF[25]	ISF[26]
5	ISFs	Index of logarithmic frame energy						Dithering flag
	ISF[27]	g[0]	g[1]	g[2]	g[3]	g[4]	g[5]	0
6	SID Type	AMR-WB Mode Indication				Padding Bits		
	1	0	0	x	x	0	0	0
7	Padding Bits							
	0	0	0	0	0	0	0	0

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#### 8.3.2 Frame Structure of Unvoiced Quarter-Rate

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The mapping of bits for the VMR-WB RS-II Unvoiced Quarter-Rate is shown in Table 8.3-2.

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**Table 8.3-2: Mapping of bits in VMR-WB Unvoiced Quarter-Rate**

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Octet	MSB	Mapping of bits in VMR-WB Unvoiced Quarter-Rate						LSB
	Bit 8	Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1
1	Identifier	ISFs						
	0	ISF[0]	ISF[1]	ISF[2]	ISF[3]	ISF[4]	ISF[5]	ISF[6]
2	ISFs							ISF[14]
	ISF[7]	ISF[8]	ISF[9]	ISF[10]	ISF[11]	ISF[12]	ISF[13]	
3	ISFs							ISF[22]
	ISF[15]	ISF[16]	ISF[17]	ISF[18]	ISF[19]	ISF[20]	ISF[21]	
4	ISFs							ISF[30]
	ISF[23]	ISF[24]	ISF[25]	ISF[26]	ISF[27]	ISF[28]	ISF[29]	
5	ISFs							ISF[31]
	FCB Gain for Sub-Frame 1						FCB Gain for Sub-Frame 2	
6	FCB Gain for Sub-Frame 2			FCB Gain for Sub-Frame 3				
	G[2]	G[3]	G[4]	G[0]	G[1]	G[2]	G[3]	G[4]
7	FCB Gain for Sub-Frame 4						Unused	0
	G[0]	G[1]	G[2]	G[3]	G[4]			

1

### 8.4 Frame Structure of CNG Eighth-Rate in Rate-Set II

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The mapping of bits for the VMR-WB RS-II Unvoiced Quarter-Rate is shown in Table 8.4-1.

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**Table 8.4-1: Mapping of bits in VMR-WB RS-II CNG Eighth-Rate**

Octet	MSB	Mapping of bits in VMR-WB RS-II CNG Eighth-Rate						LSB
	Bit 8	Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1
1	ISFs							ISF[7]
	ISF[0]	ISF[1]	ISF[2]	ISF[3]	ISF[4]	ISF[5]	ISF[6]	
2	ISFs						FCB Gain	
	ISF[8]	ISF[9]	ISF[10]	ISF[11]	ISF[12]	ISF[13]	G[0]	G[1]
3	FCB Gain				Excitation Gain			
	G[2]	G[3]	G[4]	G[5]				

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### 8.5 Frame Structure of CNG Eighth-Rate in Rate-Set I

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The mapping of bits for the VMR-WB RS-I Unvoiced Eighth-Rate is shown in Table 8.5-1.

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**Table 8.5-1: Mapping of bits in VMR-WB RS-I CNG Eighth-Rate**

Octet	MSB	Mapping of bits in VMR-WB RS-I CNG Eighth-Rate						LSB
	Bit 8	Bit 7	Bit 6	Bit 5	Bit 4	Bit 3	Bit 2	Bit 1
1	ISFs							ISF[7]
	ISF[0]	ISF[1]	ISF[2]	ISF[3]	ISF[4]	ISF[5]	ISF[6]	
2	ISFs			Excitation Gain				
	ISF[8]	ISF[9]	ISF[10]	G[0]	G[1]	G[2]	G[3]	G[4]

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### 8.6 VMR-WB Storage File Format

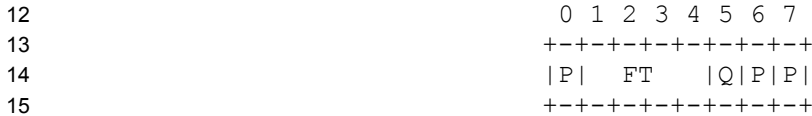
14

15

1 The storage file format in VMR-WB codec simulation is activated with command line parameter “-  
 2 mime”. The storage file format is used for storing VMR-WB encoded speech frames in a file or as an  
 3 e-mail attachment.

5 This format is used for file storage and defines the structure of the data bits (bit stream) contained in  
 6 a VMR-WB sample of a 3G2 file [19]. Note that the generic ISO-base media file format [32] specified  
 7 in 3GPP2 C.S0050-A v1.0 [19] is used for VMR-WB codec. The real-time transport format (RTP  
 8 payload) for VMR-WB codec has been developed by IETF and is specified in [29].

10 Each stored speech frame starts with a one-octet frame header with the following format:

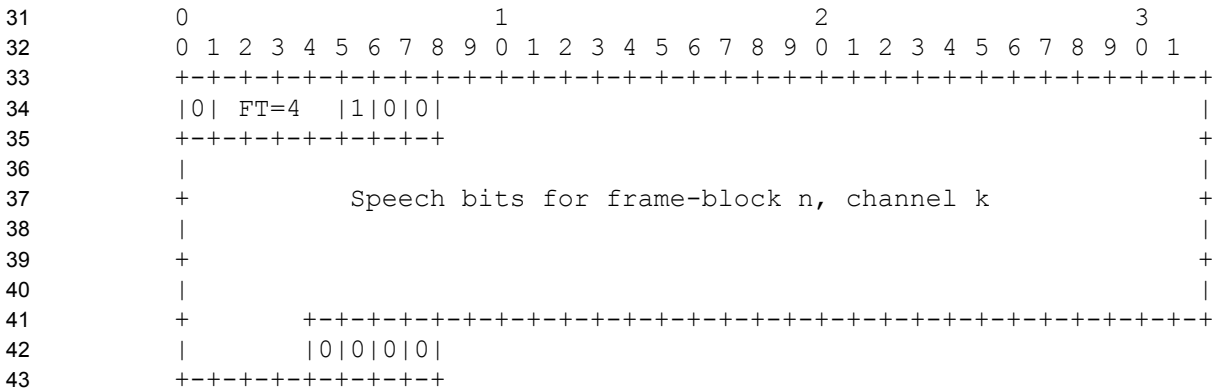


17 The Frame Type (FT) field is defined in Table 8.6-1. Padding bits P MUST be set to zero and MUST  
 18 be ignored by a receiver. Depending on the mode of operation, a subset of Frame Types are used. If  
 19 a frame with an invalid FT value is received, the frame shall be discarded.

21 Q (1 bit): Frame quality indicator. If set to 0, indicates the corresponding frame is corrupted. The  
 22 VMR-WB encoder always sets Q bit to 1. The VMR-WB decoder may ignore the Q bit.

24 Following this one-octet header and depending on the frame type, the speech bits are placed as  
 25 defined in Sections 8.1 through 8.5 of this document. The last octet of each frame is padded with  
 26 zeroes, if needed, to achieve octet alignment.

28 The following example shows a VMR-WB speech frame encoded at Half-Rate (with 124 speech bits)  
 29 in the storage format.



45 Frame-blocks or speech frames lost in transmission MUST be stored as Erasure/SPEECH\_LOST  
 46 (FT=14) and non-received frame-blocks between SID updates during non-speech periods (when  
 47 using DTX) MUST be stored as Blank/NO\_DATA frames (FT=15) in complete frame-blocks to  
 48 maintain synchronization with the original media.

50 **Table 8.6-1: VMR-WB frame types for storage formats.**

FT	Encoding Type	Frame Size (bits)
0	AMR-WB Interoperable Full-Rate (AMR-WB 6.6 kbps)	132
1	AMR-WB Interoperable Full-Rate (AMR-WB 8.85 kbps)	177
2	AMR-WB Interoperable Full-Rate (AMR-WB 12.65 kbps)	253
3	Full-Rate 13.3 kbps	266

4	Half-Rate 6.2 kbps	124
5	Quarter-Rate 2.7 kbps	54
6	Eighth-Rate 1.0 kbps	20
7	Full-Rate 8.55 kbps	171
8	Half-Rate 4.0 kbps	80
9	CNG (AMR-WB SID)	40
10	Quarter-Rate 2.0 kbps	40
11	Eighth-Rate 0.8 kbps	16
12	(Reserved)	-
13	(Reserved)	-
14	Erasure (AMR-WB SPEECH_LOST)	0
15	Blank (AMR-WB NO_DATA)	0

1

2 While support for AMR-WB codec modes 6.60 and 8.85 kbps is included in the VMR-WB file format,  
3 due to insufficient quality of these modes, content generation using these modes is not  
4 recommended.

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## 9 SUPPORT FOR TDD/TTY AND LOW-RATE IN-BAND DATA

The VMR-WB codec provides support for vocoder-independent in-band data transport applications such as TDD/TTY by reserving unique bit patterns in the Full-Rate and Half-Rate encoding operation of the codec.

Since the implementation of TDD/TTY or DTMF for the cdma2000 wideband speech codec is not a requirement, the VMR-WB codec provides only the necessary support for these applications and the actual implementation of these applications is beyond the scope of this standard.

In-band data may be transported in one of three different ways:

1. Rate 1 packet with data (256 bits/packet=12800 bps)
2. Rate 1 packet with data and speech (124 bits/packet=6300 bps)
3. Rate ½ packet with data (114 bits/packet=5700 bps)

### 9.1 TTY/TDD Frame Format in Rate-Set II

The following unique bit patterns are reserved in VMR-WB standard for in-band data transport. Note that the reserved bit-patterns are independent of the mode of operation and are not used by the VMR-WB encoder during the processing of audio/speech signals.

#### 9.1.1 Full-Rate with TDD/TTY Data

1111101101	256 bits of In-Band Data (TDD/TTY)	
Bit Position	# Of Bits	Content
1-10	10	Preamble (Frame Identifier)
11-266	256	In-Band Data

The preamble (i.e., the first 10 bits) of the frame indicates that the frame contains in-band data (e.g., TDD/TTY) and the next contiguous 256 bits (i.e., corresponding to 12800 bits/second) contain the actual non-speech data.

However, for the purpose of satisfying the frame-level signaling requirement and thereby a need for occasional Full-Rate to Half-Rate conversion without loss of in-band data, the number of data bits should be limited to 114. For Full-Rate to Half-Rate conversion, the Full-Rate preamble bits, and the unused bits are removed by the interworking function and a Half-Rate preamble is added to the beginning of the 114 bits of in-band data to form a valid Rate-Set II Half-Rate data frame.

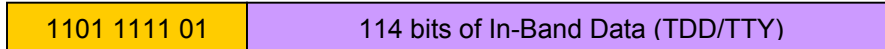
#### 9.1.2 Full-Rate with TDD/TTY Data + Speech Data

1111 1010 01	126 bits of In-Band Data (TDD/TTY)	124 bits of a Half-Rate speech packet	6 unused bits
Bit Position	# Of Bits	Content	
1-10	10	Preamble (Frame Identifier)	
11-136	126	In-Band Data	
137-260	124	Speech Data	
261-266	6	Unused	

The second type of Full-Rate frames with a combination of in-band and speech data uses a unique preamble as illustrated above. There are a maximum of 126 contiguous bits of in-band data (i.e., corresponding to 6300 bits/second) in addition to 124 bits of speech data (i.e., a valid Half-Rate frame) and some unused bits.

1 However, for the purpose of satisfying the frame-level signaling requirement and thereby a need for  
 2 occasional Full-Rate to Half-Rate conversion without loss of in-band data, the number of data bits  
 3 should be limited to 114 (i.e., corresponding to 5700 bits/second). For Full-Rate to Half-Rate  
 4 conversion, the Full-Rate preamble bits, and the unused bits are removed by the interworking  
 5 function and a Half-Rate preamble is added to the beginning of the 114 bits of in-band data to form a  
 6 valid Rate-Set II Half-Rate data frame.

7  
 8 **9.1.3 Half-Rate with TDD/TTY Data**



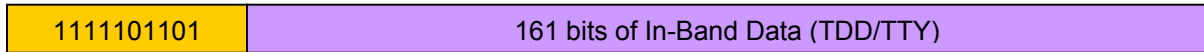
Bit Position	# Of Bits	Content
1-10	10	Preamble (Frame Identifier)
11-124	114	In-Band Data

11  
 12 The VMR-WB codec is also capable of transporting in-band data using the Half-Rate frames. The  
 13 frame structure of the Half-Rate frames containing non-speech data is illustrated above. Note the  
 14 unique bit pattern in the beginning of the non-speech data packet. The VMR-WB has the capability of  
 15 transporting a maximum of 114 bits (i.e., corresponding to 5700 bits/second) using the Half-Rate  
 16 encoding rate.

17  
 18 **9.2 TTY/TDD Frame Format in Rate-Set I**

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 20 The following unique bit patterns are reserved in VMR-WB standard operating in Rate-Set I for in-  
 21 band data transport. Note that the reserved bit-patterns are independent of the mode of operation and  
 22 are not used by the VMR-WB encoder during the processing of audio/speech signals. *Also note that*  
 23 *the reserved bit patterns (preambles) for Rate-Set I and Rate-Set II modes of operation are identical.*

24  
 25 **9.2.1 Full-Rate with TDD/TTY Data**

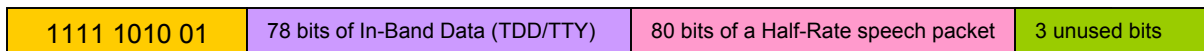


Bit Position	# Of Bits	Content
1-10	10	Preamble (Frame Identifier)
11-161	161	In-Band Data

28  
 29 The preamble (i.e., the first 10 bits) of the frame indicates that the frame contains in-band data (e.g.,  
 30 TDD/TTY) and the next contiguous 161 bits (i.e., corresponding to 8150 bits/second) contain the  
 31 actual non-speech data.

32  
 33 However, for the purpose of satisfying the frame-level signaling requirement and thereby a need for  
 34 occasional Full-Rate to Half-Rate conversion without loss of in-band data, the number of data bits  
 35 should be limited to 78. For Full-Rate to Half-Rate conversion, the Full-Rate preamble bits, and the  
 36 unused bits are removed by the interworking function and a Half-Rate preamble is added to the  
 37 beginning of the 80 bits of in-band data to form a valid Rate-Set II Half-Rate data frame.

38  
 39 **9.2.2 Full-Rate with TDD/TTY Data + Speech Data**

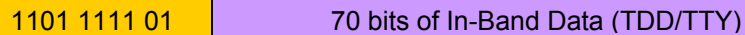


Bit Position	# Of Bits	Content
1-10	10	Preamble (Frame Identifier)
11-88	78	In-Band Data
89-168	80	Speech Data
169-171	3	Unused

1  
2 The second type of Full-Rate frames with a combination of in-band and speech data uses a unique  
3 preamble as illustrated above. There are a maximum of 78 contiguous bits of in-band data (i.e.,  
4 corresponding to 3900 bits/second) in addition to 80 bits of speech data (i.e., a valid Half-Rate frame)  
5 and some unused bits.

6  
7 However, for the purpose of satisfying the frame-level signaling requirement and thereby a need for  
8 occasional Full-Rate to Half-Rate conversion without loss of in-band data, the number of data bits  
9 should be limited to 70 (i.e., corresponding to 3500 bits/second). For Full-Rate to Half-Rate  
10 conversion, the Full-Rate preamble bits, and the unused bits are removed by the interworking  
11 function and a Half-Rate preamble is added to the beginning of the 70 bits of in-band data to form a  
12 valid Rate-Set I Half-Rate data frame.

### 13 9.2.3 Half-Rate with TDD/TTY Data

14  
15 

16

Bit Position	# Of Bits	Content
1-10	10	Preamble (Frame Identifier)
11-80	70	In-Band Data

17  
18 The VMR-WB codec is also capable of transporting in-band data using the Half-Rate frames. The  
19 frame structure of the Half-Rate frames containing non-speech data is illustrated above. Note the  
20 unique bit pattern in the beginning of the non-speech data packet. The VMR-WB has the capability of  
21 transporting a maximum of 70 bits (i.e., corresponding to 3500 bits/second) using the Half-Rate  
22 encoding rate.