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TERMINAL EQUIPMENTS

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7 kHz AUDIO - CODING WITHIN 64 KBIT/S

ITU-T Recommendation G.722

(Extract from the *Blue Book*)

NOTES

1 ITU-T Recommendation G.722 was published in Fascicle III.4 of the *Blue Book*. This file is an extract from the *Blue Book.* While the presentation and layout of the text might be slightly different from the *Blue Book* version, the contents of the file are identical to the *Blue Book* version and copyright conditions remain unchanged (see below).

2 In this Recommendation, the expression "Administration" is used for conciseness to indicate both a telecommunication administration and a recognized operating agency.

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Recommendation G.722

7 kHz AUDIO-CODING WITHIN 64 KBIT/S

(Melbourne, 1988)

1 General

1.1 *Scope and outline description*

This Recommendation describes the characteristics of an audio (50 to 7 000 Hz) coding system which may be used for a variety of higher quality speech applications. The coding system uses sub-band adaptive differential pulse code modulation (SB-ADPCM) within a bit rate of 64 kbit/s. The system is henceforth referred to as 64 kbit/s (7 kHz) audio coding. In the SB-ADPCM technique used, the frequency band is split into two sub-bands (higher and lower) and the signals in each sub-band are encoded using ADPCM. The system has three basic modes of operation corresponding to the bit rates used for 7 kHz audio coding: 64, 56 and 48 kbit/s. The latter two modes allow an auxiliary data channel of 8 and 16 kbit/s respectively to be provided within the 64 kbit/s by making use of bits from the lower sub-band.

Figure 1/G.722 identifies the main functional parts of the 64 kbit/s (7 kHz) audio codec as follows:

i) 64 kbit/s (7 kHz) audio encoder comprising:

- a transmit audio part which converts an audio signal to a uniform digital signal which is coded using 14 bits with 16 kHz sampling;
- a SB-ADPCM encoder which reduces the bit rate to 64 kbit/s.

ii) 64 kbit/s (7 kHz) audio decoder comprising:

- a SB-ADPCM decoder which performs the reverse operation to the encoder, noting that the effective audio coding bit rate at the input of the decoder can be 64, 56 or 48 kbit/s depending on the mode of operation;
- a receive audio part which reconstructs the audio signal from the uniform digital signal which is encoded using 14 bits with 16 kHz sampling.

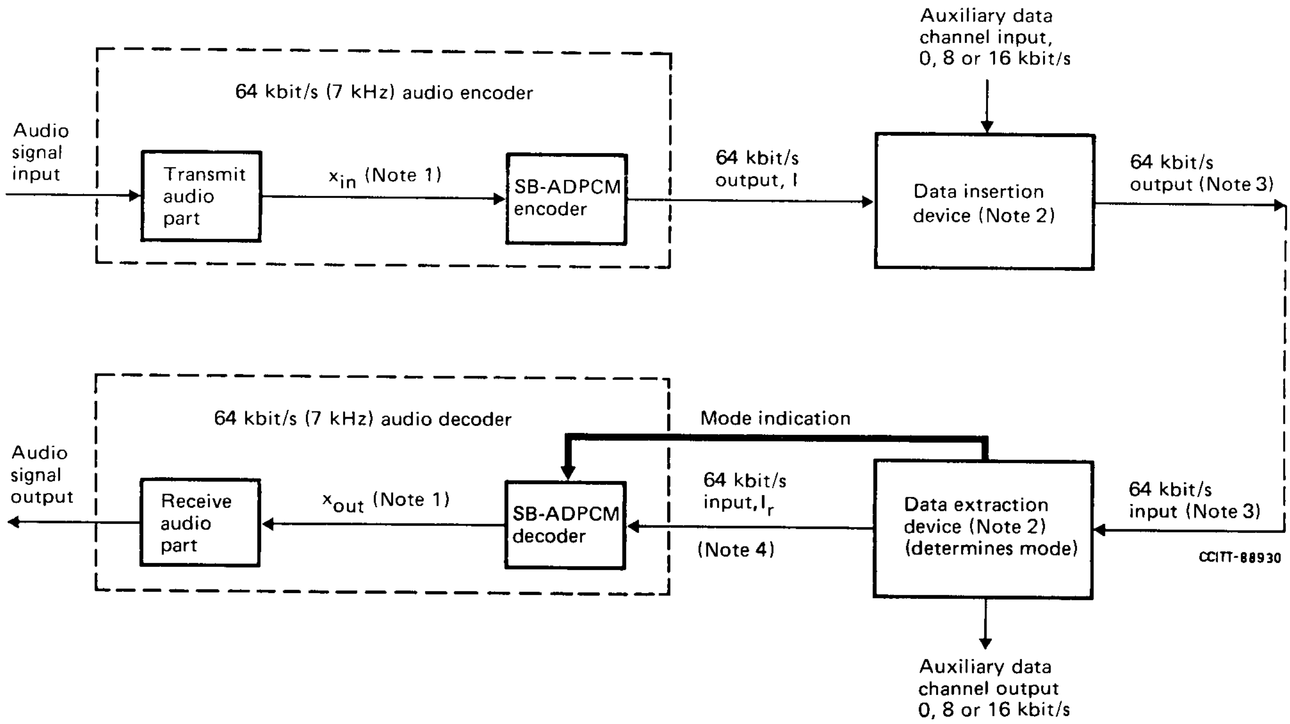
The following two parts, identified in Figure 1/G.722 for clarification, will be needed for applications requiring an auxiliary data channel within the 64 kbit/s:

- a data insertion device at the transmit end which makes use of, when needed, 1 or 2 audio bits per octet depending on the mode of operation and substitutes data bits to provide an auxiliary data channel of 8 or 16 kbit/s respectively;
- a data extraction device at the receive end which determines the mode of operation according to a mode control strategy and extracts the data bits as appropriate.

Paragraph 1.2 contains a functional description of the transmit and receive audio parts, § 1.3 describes the modes of operation and the implication of inserting data bits on the algorithms, whilst §§ 1.4 and 1.5 provide the functional descriptions of the SB-ADPCM encoding and decoding algorithms respectively. Paragraph 1.6 deals with the timing requirements. Paragraph 2 specifies the transmission characteristics of the 64 kbit/s (7 kHz) audio codec and of the transmit and receive audio parts, §§ 3 and 4 give the principles of the SB-ADPCM encoder respectively whilst §§ 5 and 6 specify the computational details of the Quadrature Mirror Filters (QMF) and of the ADPCM encoders and decoders respectively.

Networking aspects and test sequences are addressed in Appendices I and II respectively to this Recommendation.

Recommendation G.725 contains specifications for in-channel handshaking procedures for terminal identification and for mode control strategy, including interworking with existing 64 kbit/s PCM terminals.



Note 1 – x_{in} and x_{out} are digital signals uniformly coded with 14 bits and 16 kHz sampling.

Note 2 – These devices are only necessary for applications requiring an auxiliary data channel within the 64 kbit/s.

Note 3 – Comprises 64, 56 or 48 kbit/s for audio coding and 0, 8 or 16 kbit/s for data.

Note 4 – 64 kbit/s signal comprising 64, 56 or 48 kbit/s for audio coding depending on the mode of operation.

FIGURE 1/G.722

Simplified functional block diagram

1.2 *Functional description of the audio parts*

Figure 2/G.722 shows a possible arrangement of audio parts in a 64 kbit/s (7 kHz) audio coding terminal. The microphone, pre-amplifier, power amplifier and loudspeaker are shown simply to identify the audio parts and are not considered further in this Recommendation.

In order to facilitate the measurement of the transmission characteristics as specified in § 2, test points A and B need to be provided as shown. These test points may either be for test purposes only or, where the audio parts are located in different units from the microphone, loudspeaker, etc., correspond to physical interfaces.

The transmit and receive audio parts comprise either the following functional units or any equivalent items satisfying the specifications of § 2:

i) transmit:

- an input level adjustment device,
- an input anti-aliasing filter,
- a sampling device operating at 16 kHz,
- an analogue-to-uniform digital converter with 14 bits and with 16 kHz sampling;

ii) receive:

- a uniform digital-to-analogue converter with 14 bits and with 16 kHz sampling,
- a reconstructing filter which includes x/sin x correction,
- an output level adjustment device.

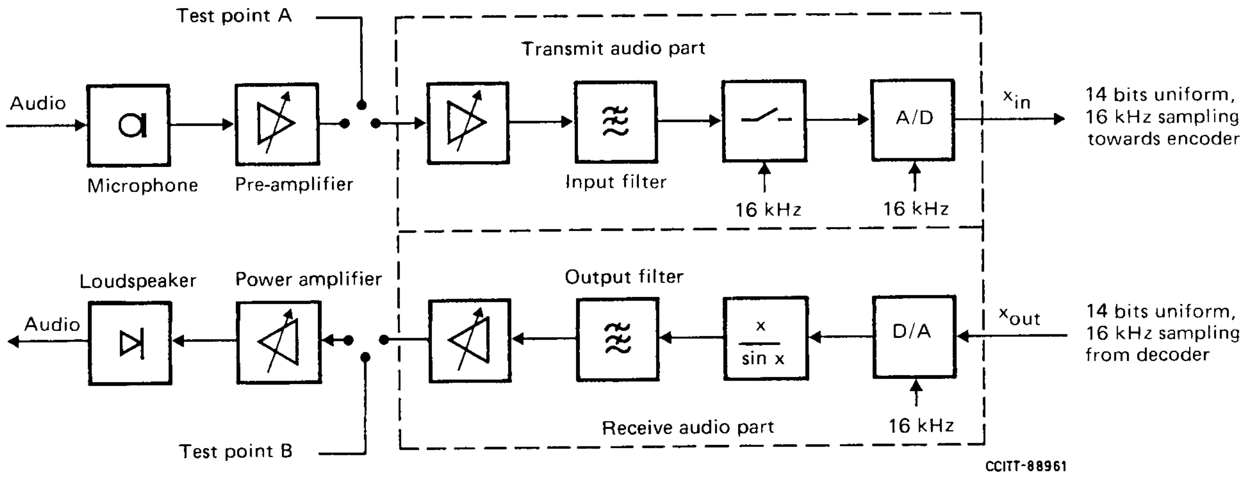


FIGURE 2/G.722

Possible implementation of the audio parts

1.3 *Possible modes of operation and implications of inserting data*

The three basic possible modes of operation which correspond to the bit rates available for audio coding at the input of the decoder are defined in Table 1/G.722.

TABLE 1/G.722

Basic possible modes of operation

Mode	7 kHz audio coding bit rate	Auxiliary data channel bit rate
1	64 kbit/s	0 kbit/s
2	56 kbit/s	8 kbit/s
3	48 kbit/s	16 kbit/s

See Appendix I for examples of applications using one or several of these modes and for their corresponding subjective quality.

The 64 kbit/s (7 kHz) audio encoder uses 64 kbit/s for audio coding at all times irrespective of the mode of operation. The audio coding algorithm has been chosen such that, without sending any indication to the encoder, the least significant bit or two least significant bits of the lower sub-band may be used downstream from the 64 kbit/s (7 kHz) audio encoder in order to substitute the auxiliary data channel bits. However, to maximize the audio performance for a given mode of operation, the 64 kbit/s (7 kHz) audio decoder must be optimized to the bit rate available for audio coding. Thus, this Recommendation describes three variants of the SB-ADPCM decoder and, for applications requiring an auxiliary data channel, an indication must be forwarded to select in the decoder the variant appropriate to the mode of operation. Figure 1/G.722 illustrates the arrangement. It should be noted that the bit rate at the input of the 64 kbit/s (7 kHz) audio decoder is always 64 kbit/s but comprising 64, 56 or 48 kbit/s for audio coding depending on the mode of operation. From an algorithm viewpoint, the variant used in the SB-ADPCM decoder can be changed in any octet during the transmission. When no indication about the mode of operation is forwarded to the decoder, the variant corresponding to Mode 1 should be used.

A mode mismatch situation, where the variant used in the 64 kbit/s (7 kHz) audio decoder for a given octet does not correspond to the mode of operation, will not cause misoperation of the decoder. However, to maximize the audio performance, it is recommended that the mode control strategy adopted in the data extraction device should be such as to minimize the duration of the mode mismatch. Appendix I gives further information on the effects of a mode mismatch. To ensure compatibility between various types of 64 kbit/s (7 kHz) audio coding terminals, it is recommended that, as a minimum, the variant corresponding to Mode 1 operation is always implemented in the decoder.

The mode control strategy could be derived from the auxiliary data channel protocol (see Recommendation G.725).

1.4 *Functional description of the SB-ADPCM encoder*

Figure 3/G.722 is a block diagram of the SB-ADPCM encoder. A functional description of each block is given below in §§ 1.4.1 to 1.4.4.

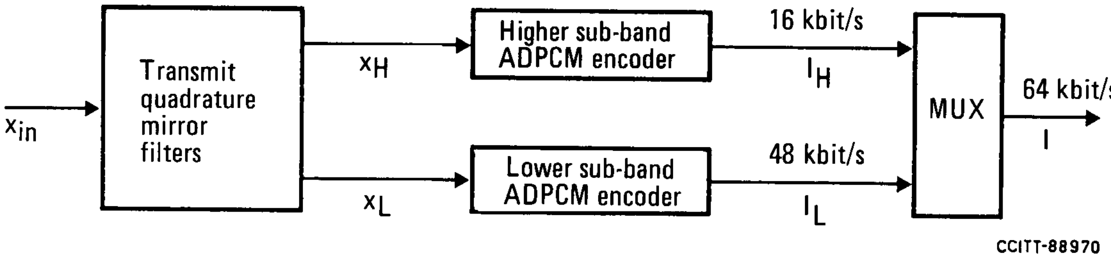


FIGURE 3/G.722

Block diagram of the SB-ADPCM encoder

1.4.1 *Transmit quadrature mirror filters* (QMFs)

The transmit QMFs comprise two linear-phase non-recursive digital filters which split the frequency band 0 to 8000 Hz into two sub-bands: the lower sub-band (0 to 4000 Hz) and the higher sub-band (4000 to 8000 Hz). The input to the transmit QMFs, x_{in}, is the output from the transmit audio part and is sampled at 16 kHz. The outputs, x_L and x_H, for the lower and higher sub-bands respectively, are sampled at 8 kHz.

1.4.2 *Lower sub-band ADPCM encoder*

Figure 4/G.722 is a block diagram of the lower sub-band ADPCM encoder. The lower sub-band input signal, x_L after subtraction of an estimate, s_L , of the input signal produces the difference signal, e_L. An adaptive 60-level non linear quantizer is used to assign six binary digits to the value of the difference signal to produce a 48 kbit/s signal, I_L.

In the feedback loop, the two least significant bits of I_L are deleted to produce a 4-bit signal I_{Lt}, which is used for the quantizer adaptation and applied to a 15-level inverse adaptive quantizer to produce a quantized difference signal, d_{Lt}. The signal estimate, s_L is added to this quantized difference signal to produce a reconstructed version, r_{Lt}, of the lower sub-band input signal. Both the reconstructed signal and the quantized difference signal are operated upon by an adaptive predictor which produce the estimate, s_L, of the input signal, thereby completing the feedback loop.

4-bit operation, instead of 6-bit operation, in the feedback loops of both the lower sub-band ADPCM encoder, and the lower sub-band ADPCM decoder allows the possible insertion of data in the two least significant bits as described in § 1.3 without causing misoperation in the decoder. Use of a 60-level quantizer (instead of 64-level) ensures that the pulse density requirements as described in Recommendation G.802 are met under all conditions and in all modes of operation.

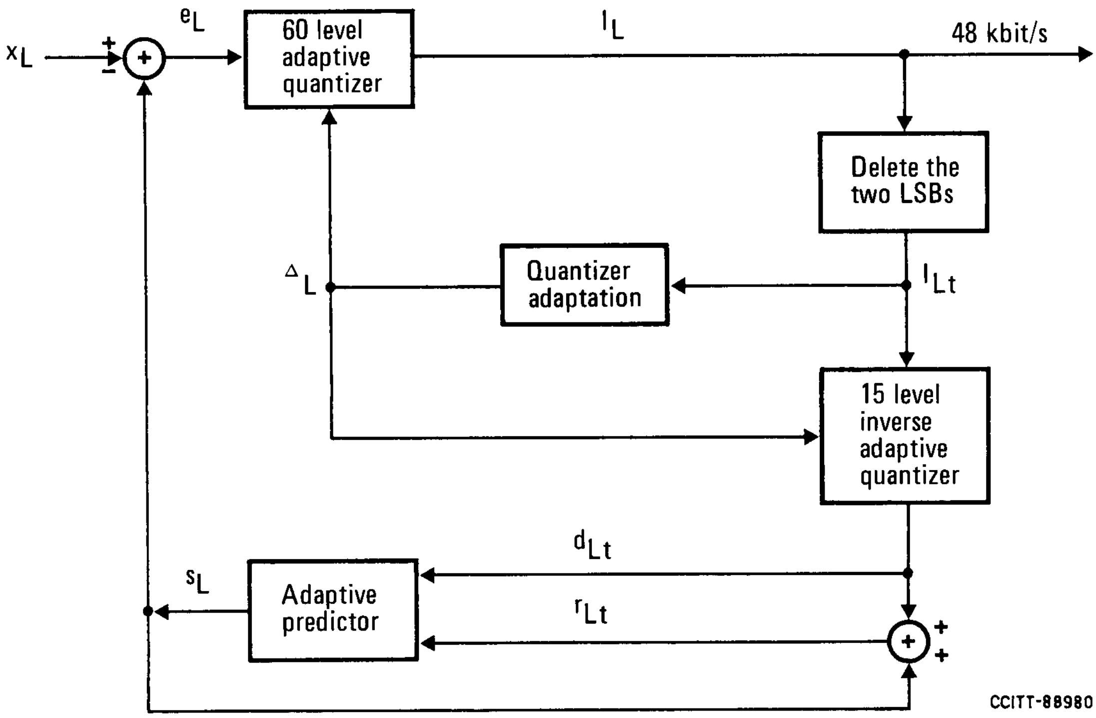


FIGURE 4/G.722

Block diagram of the lower sub-band ADPCM encoder

1.4.3 *Higher sub-band ADPCM encoder*

Figure 5/G.722 is a block diagram of the higher sub-band ADPCM encoder. The higher sub-band input signal, x_H after subtraction of an estimate, s_H, of the input signal, produces the difference signal, e_H. An adaptive 4-level non linear quantizer is used to assign two binary digits to the value of the difference signal to produce a 16 kbit/s signal, I_H.

An inverse adaptive quantizer produces a quantized difference signal, d_H, from these same two binary digits. The signal estimate, s_H, is added to this quantized difference signal to produce a reconstructed version, r_H, of the higher sub-band input signal. Both the reconstructed signal and the quantized difference signal are operated upon by an adaptive predictor which produces the estimate, s_H, of the input signal, thereby completing the feedback loop.

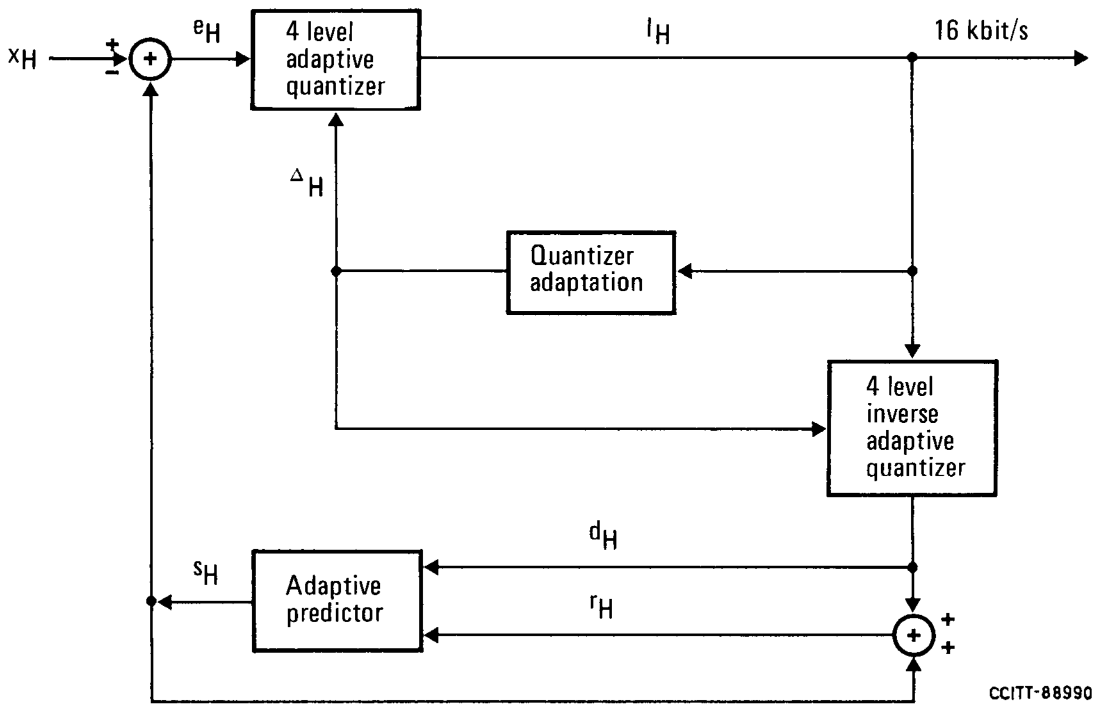


FIGURE 5/G.722

Block diagram of the higher sub-band ADPCM encoder

1.4.4 *Multiplexer*

The multiplexer (MUX) shown in Figure 3/G.722 is used to combine the signals, I_L *and* I_H, from the lower and higher sub-band ADPCM encoders respectively into a composite 64 kbit/s signal, I, with an octet format for transmission.

The output octet format, after multiplexing, is as follows:

$$I_{H1}\ I_{H2}\ I_{L1}\ I_{L2}\ I_{L3}\ I_{L4}\ I_{L5}\ I_{L6}$$

where I_{H1} is the first bit transmitted, and where I_{H1} and I_{L1} are the most significant bits of I_H and I_L respectively, whilst I_{H2} and I_{L6} are the least significant bits of I_H and I_L respectively.

1.5 *Functional description of the SB-ADPCM decoder*

Figure 6/G.722 is a block diagram of the SB-ADPCM decoder. A functional description of each block is given below in §§ 1.5.1 to 1.5.4.

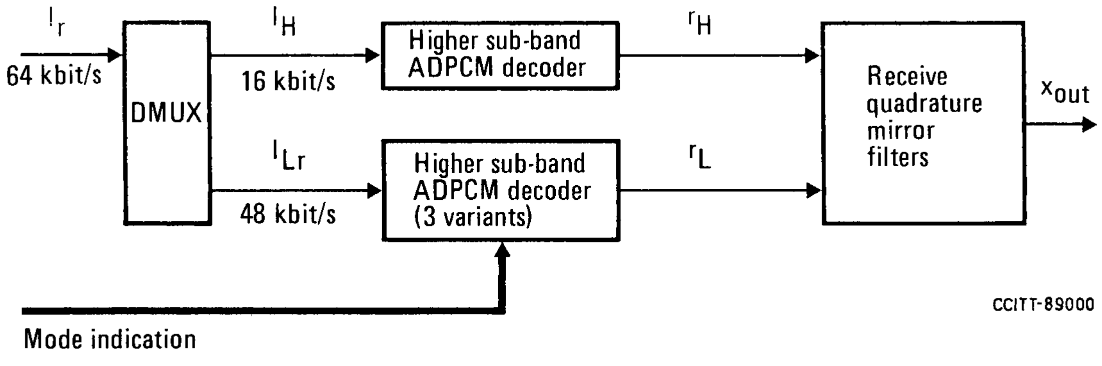


FIGURE 6/G.722

Block diagram of the SB-ADPCM decoder

1.5.1 *Demultiplexer*

The demultiplexer (DMUX) decomposes the received 64 kbit/s octet-formatted signal, I_r, into two signals, I_{Lr} and I_H, which form the codeword inputs to the lower and higher sub-band ADPCM decoders respectively.

1.5.2 *Lower sub-band ADPCM decoder*

Figure 7/G.722 is a block diagram of the lower sub-band ADPCM decoder. This decoder can operate in any of three possible variants depending on the received indication of the mode of operation.

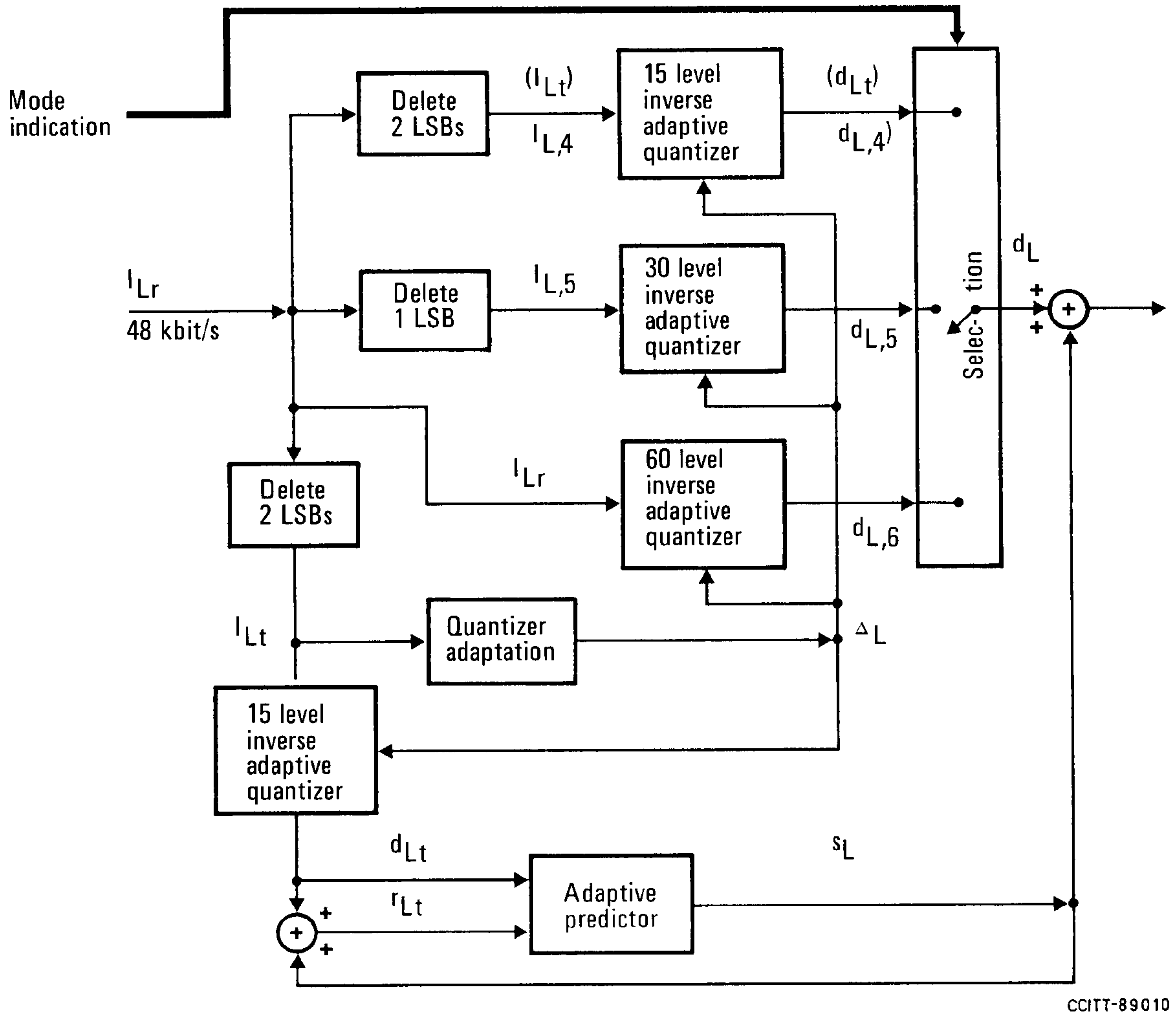


FIGURE 7/G.722

Block diagram of the lower sub-band ADPCM decoder

The path which produces the estimate, s_L, of the input signal including the quantizer adaptation, is identical to the feedback portion of the lower sub- band ADPCM encoder described in § 1.4.2. The reconstructed signal, r_L, is produced by adding to the signal estimate one of three possible quantized difference signals, $d_{L,6}$, $d_{L,5}$ or $d_{L,4}$ *(=* d_{Lt} - see note), selected according to the received indication of the mode of operation. For each indication, Table 2/G.722 shows the quantized difference signal selected, the inverse adaptive quantizer used and the number of least significant bits deleted from the input codeword.

TABLE 2/G.722

Lower sub-band ADPCM decoder variants

Received indication of mode of operation	Quantized difference signal selected	Inverse adaptive quantizer used	Number of least significant bits deleted from input codeword, I_{Lr}
Mode 1	$d_{L,6}$	60-level	0
Mode 2	$d_{L,5}$	30-level	1
Mode 3	$d_{L,4}$	15-level	2

Note - For clarification purposes, all three inverse quantizers have been indicated in the upper portion of Figure 7/G.722. In an optimized implementation, the signal d_{Lt}, produced in the predictor loop, could be substituted for $d_{L,4}$.

1.5.3 *Higher sub-band ADPCM decoder*

Figure 8/G.722 is a block diagram of the higher sub-band ADPCM decoder. This decoder is identical to the feedback portion of the higher sub-band ADPCM encoder described in § 1.4.3, the output being the reconstructed signal, r_H.

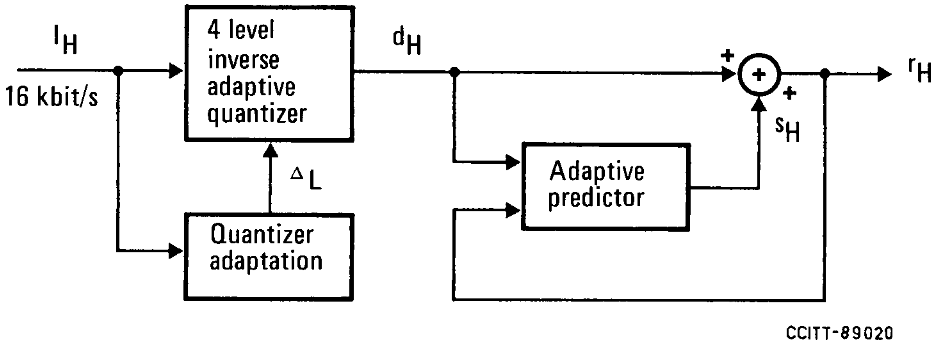


FIGURE 8/G.722

Block diagram of the higher sub-band ADPCM decoder

1.5.4 *Receive QMFs*

The receive QMFs shown in Figure 6/G.722 are two linear-phase non-recursive digital filters which interpolate the outputs, r_L and r_H, of the lower and higher sub-band ADPCM decoders from 8 kHz to 16 kHz and which then produce an output, x_{out}, sampled at 16 kHz which forms the input to the receive audio parts.

Excluding the ADPCM coding processes, the combination of the transmit and the receive QMFs has an impulse response which closely approximates a simple delay whilst, at the same time, the aliasing effects associated with the 8 kHz sub-sampling are cancelled.

1.6 *Timing requirements*

64 kHz bit timing and 8 kHz octet timing should be provided by the network to the audio decoder.

For a correct operation of the audio coding system, the precision of the 16 kHz sampling frequencies of the A/D and D/A converters must be better than $\pm 50 \cdot 10^{-6}$.

2 Transmission characteristics

2.1 *Characteristics of the audio ports and the test points*

Figure 2/G.722 indicates the audio input and output ports and the test points (A and B). It is for the designer to determine the characteristics of the audio ports and the test points (i.e. relative levels, impedances, whether balanced or unbalanced). The microphone, pre-amplifier, power amplifier and loudspeaker should be chosen with reference to the specifications of the audio parts: in particular their nominal bandwidth, idle noise and distortion.

It is suggested that input and ouput impedances should be high and low, respectively, for an unbalanced termination whilst for a balanced termination these impedances should be 600 ohms. However, the audio parts should meet all audio parts specifications for their respective input and output impedance conditions.

2.2 *Overload point*

The overload point for the analogue-to-digital and digital-to-analogue converters should be + 9 dBm0 ± 0.3 dB. This assumes the same nominal speech level (see Recommendation G.232) as for 64 kbit/s PCM, but with a wider margin for the maximum signal level which is likely to be necessary with conference arrangements. The measurement method of the overload point is under study.

2.3 *Nominal reference frequency*

Where a nominal reference frequency of 1000 Hz is indicated below, the actual frequency should be chosen equal to 1020 Hz. The frequency tolerance should be + 2 to - 7 Hz.

2.4 *Transmission characteristics of the 64 kbit/s (7 kHz) audio codec*

The values and limits specified below should be met with a 64 kbit/s (7 kHz) audio encoder and decoder connected back-to-back. For practical reasons, the measurements may be performed in a looped configuration as shown in Figure 9a)/G.722. However, such a looped configuration is only intended to simulate an actual situation where the encoder and decoder are located at the two ends of a connection.

These limits apply to operation in Mode 1.

2.4.1 *Nominal bandwidth*

The nominal 3 dB bandwidth is 50 to 7000 Hz.

2.4.2 *Attenuation/frequency distortion*

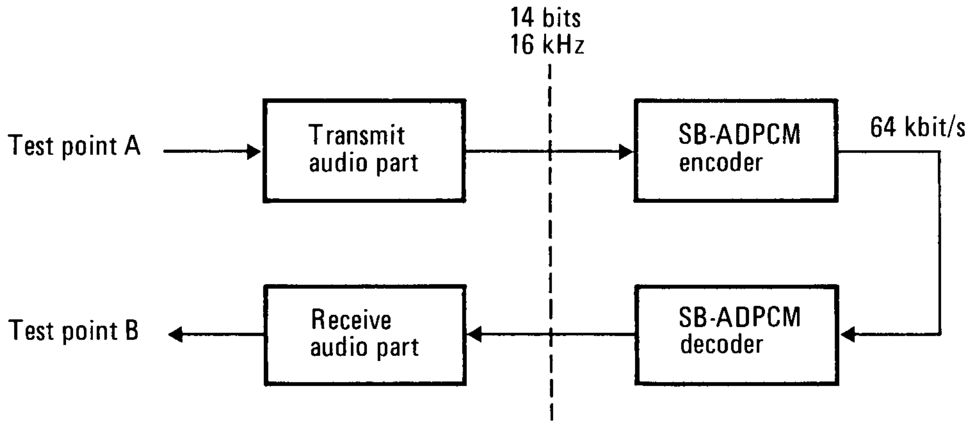
The variation with frequency of the attenuation should satisfy the limits shown in the mask of Figure 10/G.722. The nominal reference frequency is 1000 Hz and the test level is -10 dBm0.

2.4.3 *Absolute group delay*

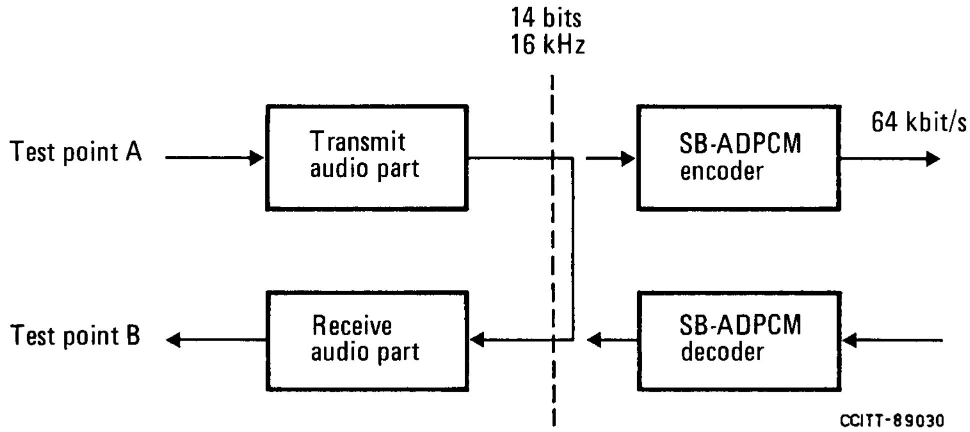
The absolute group delay, defined as the minimum group delay for a sine wave signal between 50 and 7000 Hz, should not exceed 4 ms. The test level is -10 dBm0.

2.4.4 *Idle noise*

The unweighted noise power measured in the frequency range 50 to 7000 Hz with no signal at the input port (test point A) should not exceed -66 dBm0. When measured in the frequency range 50 Hz to 20 kHz the unweighted noise power should not exceed -60 dBm0.



a)



b)

FIGURE 9/G.722

Looped measurement configurations

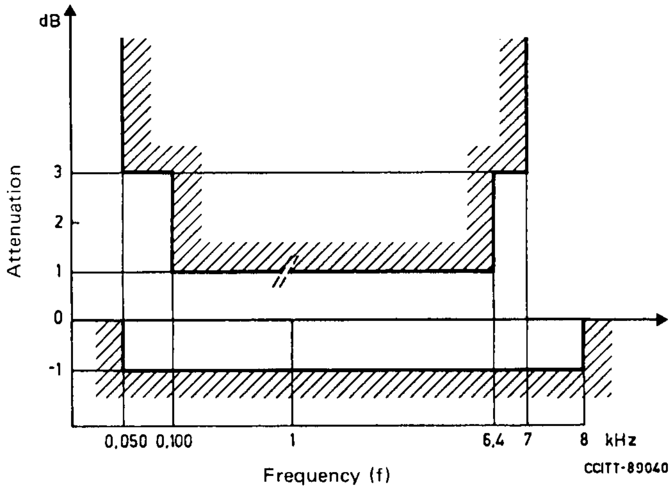


FIGURE 10/G.722

Attenuation distortion versus frequency

2.4.5 *Single frequency noise*

The level of any single frequency (in particular 8000 Hz, the sampling frequency and its multiples), measured selectively with no signal at the input port (test point A) should not exceed -70 dBm0.

2.4.6 *Signal-to-total distortion ratio*

Under study.

2.5 *Transmission characteristics of the audio parts*

When the measurements indicated below for the audio parts are from audio-to-audio, a looped configuration as shown in Figure 9b)/G.722 should be used. The audio parts should also meet the specifications of § 2.4 with the measurement configuration of Figure 9b)/G.722.

2.5.1 *Attenuation/frequency response of the input anti-aliasing filter*

The in-band and out-of-band attenuation/frequency response of the input anti-aliasing filter should satisfy the limits of the mask shown in Figure 11/G.722. The nominal reference frequency is 1000 Hz and the test level for the in-band characteristic is -10 dBm0. Appropriate measurements should be made to check the out-of-band characteristic taking into account the aliasing due to the 16 kHz sampling.

2.5.2 *Attenuation/frequency response of the output reconstructing filter*

The in-band and out-of-band attenuation/frequency response of the output reconstructing filter should satisfy the limits of the mask shown in Figure 12/G.722. The nominal reference frequency is 1000 Hz and the test level for the in-band characteristic is -10 dBm0. Appropriate measurements should be made to check the out-of-band characteristic taking into account the aliasing due to the 16 kHz sampling. The mask of Figure 12/G.722 is valid for the whole of the receive audio part including any pulse amplitude modulation distortion and x/sin x correction.

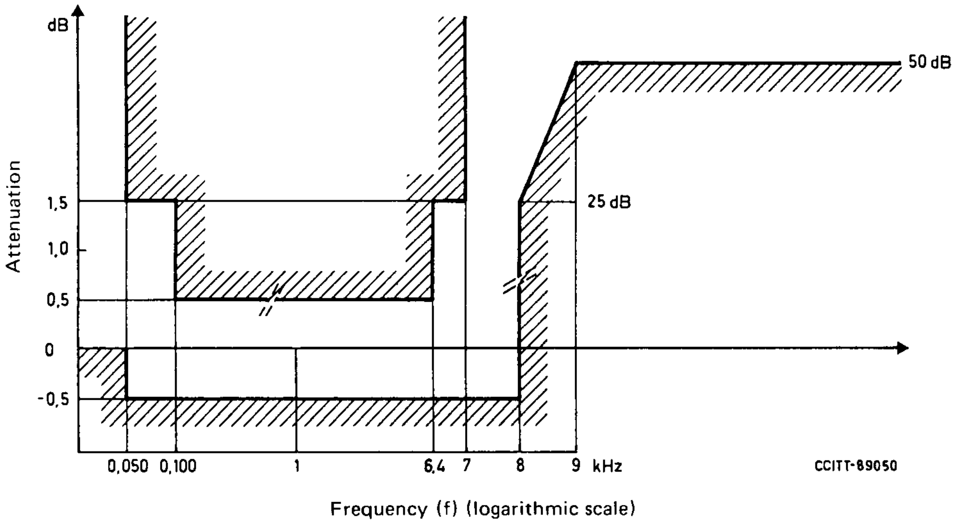


FIGURE 11/G.722

Attenuation/frequency response of the input antialiasing filter

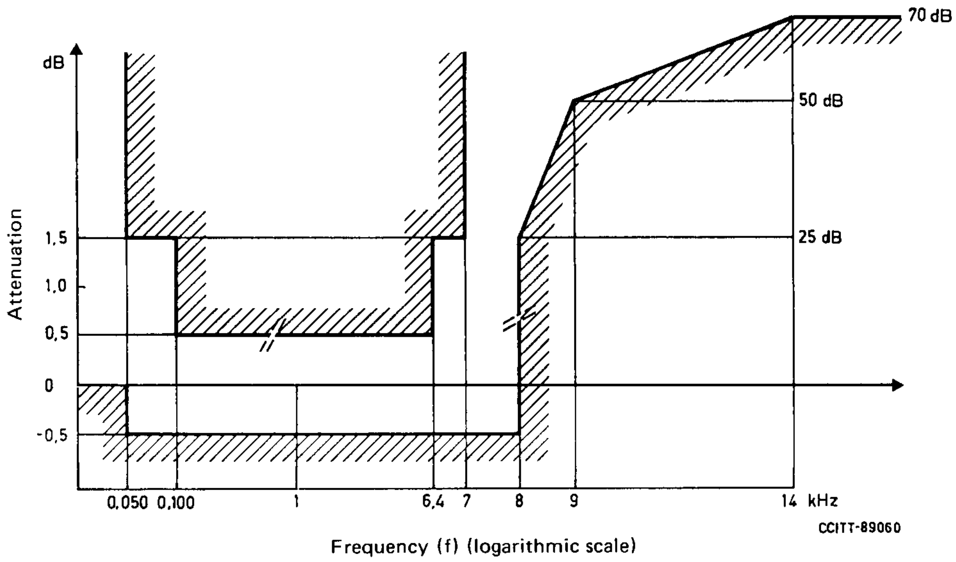


FIGURE 12/G.722

Attenuation/frequency response of the output reconstructing filter (including x/sin x correction)

2.5.3 *Group-delay distortion with frequency*

The group-delay distortion, taking the minimum value of group delay as a reference, should satisfy the limits of the mask shown in Figure 13/G.722.

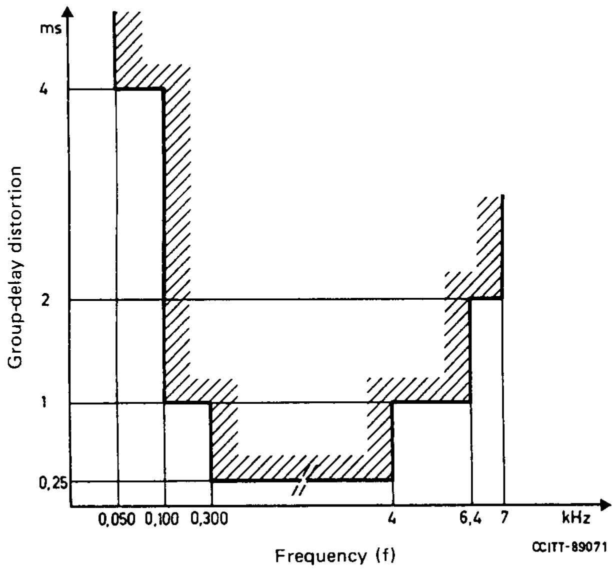


FIGURE 13/G.722

Group-delay distortion versus frequency

2.5.4 *Idle noise for the receive audio part*

The unweighted noise power of the receive audio part measured in the frequency range 50 to 7000 Hz with 14-bit all-zero signal at its input should not exceed -75 dBm0.

2.5.5 *Signal-to-total distortion ratio as a function of input level*

With a sine wave signal at a frequency excluding simple harmonic relationships with the 16 kHz sampling frequency, applied to test point A, the ratio of signal-to-total distortion power as a function of input level measured unweighted in the frequency range 50 to 7000 Hz at test point B, should satisfy the limits of the mask shown in Figure 14/G.722. Two measurements should be performed, one at a frequency of about 1 kHz and the other at a frequency of about 6 kHz.

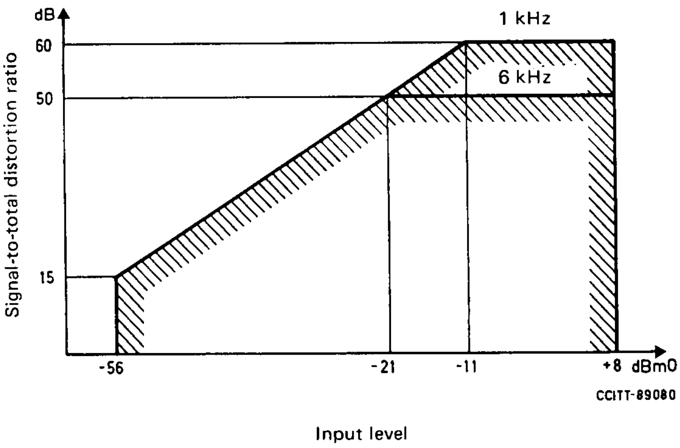


FIGURE 14/G.722

Signal-to-total distortion ratio as a function of input level

2.5.6 *Signal-to-total distortion ratio as a function of frequency*

With a sine wave signal at a level of -10 dBm0 applied to test point A, the ratio of signal-to-total distortion power as a function of frequency measured unweighted in the frequency range 50 to 7000 Hz at test point B should satisfy the limits of the mask shown in Figure 15/G.722.

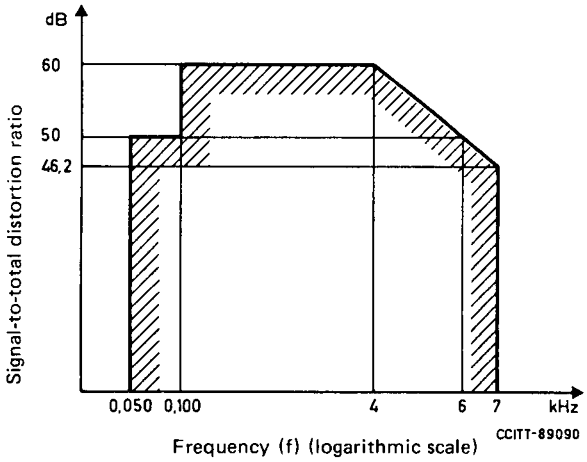


FIGURE 15/G.722

Signal-to-total distortion ratio as a function of frequency

2.5.7 *Variation of gain with input level*

With a sine wave signal at the nominal reference frequency of 1000 Hz, but excluding the sub-multiple of the 16 kHz sampling frequency, applied to test point A, the gain variation as a function of input level relative to the gain at an input level of -10 dBm0 measured selectively at test point B, should satisfy the limits of the mask shown in Figure 16/G.722.

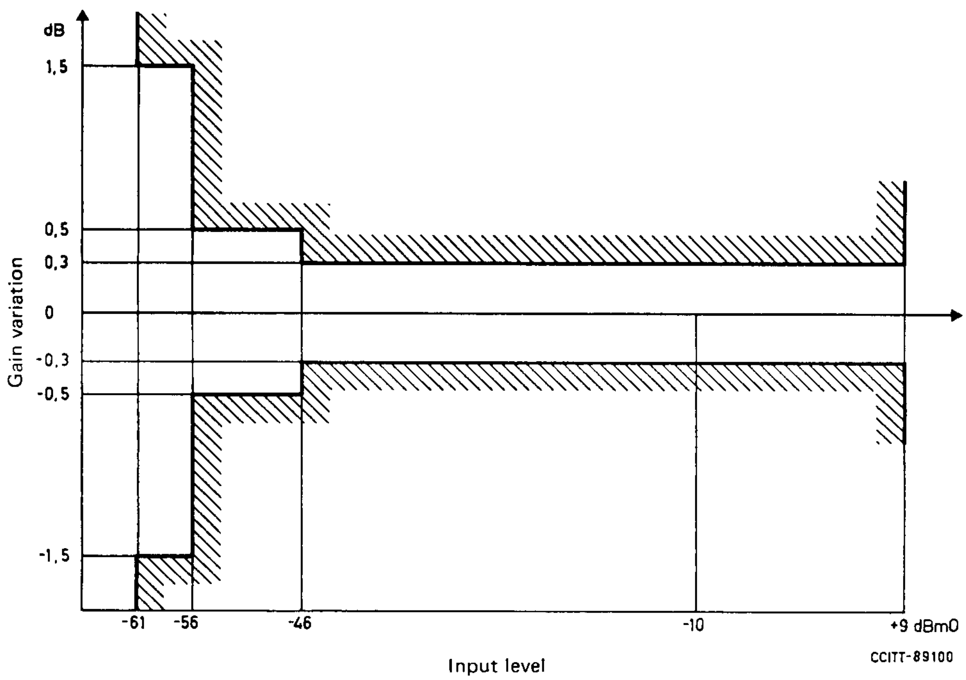


FIGURE 16/G.722

Variation of gain with input level

2.5.8 *Intermodulation*

Under study.

2.5.9 *Go/return crosstalk*

The crosstalk from the transmit direction to the receive direction should be such that, with a sine wave signal at any frequency in the range 50 to 7000 Hz and at a level of +6 dBm0 applied to test point A, the crosstalk level measured selectively at test point B should not exceed -64 dBm0. The measurement should be made with a 14-bit all-zero digital signal at the input to the receive audio part.

The crosstalk from the receive direction to the transmit direction should be such that, with a digitally simulated sine wave signal at any frequency in the range of 50 to 7000 Hz and a level of +6 dBm0 applied to the input of the receive audio part, the crosstalk level measured selectively and with the measurement made digitally at the output of the transmit audio part should not exceed -64 dBm0. The measurement should be made with no signal at test point A, but with the test point correctly terminated.

2.6 *Transcoding to and from 64 kbit/s PCM*

For compatibility reasons with 64 kbit/s PCM, transcoding between 64 kbit/s (7 kHz) audio coding and 64 kbit/s PCM should take account of the relevant specifications of Recommendations G.712, G.713 and G.714. When the audio signal is to be heard through a loudspeaker, more stringent specifications may be necessary. Further information may be found in Appendix I.

3 SB-ADPCM encoder principles

A block diagram of the SB-ADPCM encoder is given in Figure 3/G.722. Block diagrams of the lower and higher sub-band ADPCM encoders are given respectively in Figures 4/G.722 and 5/G.722.

Main variables used for the descriptions in §§ 3 and 4 are summarized in Table 3/G.722. In these descriptions, index (j) indicates a value corresponding to the current 16 kHz sampling interval, index (j-1) indicates a value corresponding to the previous 16 kHz sampling interval, index (n) indicates a value corresponding to the current 8 kHz sampling interval, and index (n-1) indicates a value corresponding to the previous 8 kHz sampling interval. Indices are not used for internal variables, i.e. those employed only within individual computational blocks.

3.1 *Transmit QMF*

A 24-coefficient QMF is used to compute the lower and higher sub-band signal components. The QMF coefficient values, h_i, are given in Table 4/G.722.

The output variables, $x_L(n)$ and $x_H(n)$, are computed in the following way:

$$x_L(n) = x_A + x_B \tag{3-1}$$

$$x_H(n) = x_A - x_B \tag{3-2}$$

$$x_A = \sum_{i=0}^{11} h_{2i} \cdot x_{in}(j-2i) \tag{3-3}$$

$$x_B = \sum_{i=0}^{11} h_{2i+1} \cdot x_{in}(j-2i-1) \tag{3-4}$$

3.2 *Difference signal computation*

The difference signals, $e_L(n)$ *and* $e_H(n)$, are computed by subtracting predicted values, $s_L(n)$ *and* $s_H(n)$, from the lower and higher sub-band input values, $x_L(n)$ and $x_H(n)$:

$$e_L(n) = x_L(n) - s_L(n) \tag{3-5}$$

$$e_H(n) = x_H(n) - s_H(n) \tag{3-6}$$

TABLE 3/G.722

Variables used in the SB-ADPCM encoder and decoder descriptions

Variable	Description
x_{in}	Input value (uniform representation)
x_L, x_H	QMF output signals
S_{Lp}, S_{Hp}	Pole-predictor output signals
$a_{L,i}$, $a_{H,i}$	Pole-predictor coefficients
r_L, r_{Lt}, r_H	Reconstructed signals (non truncated and truncated)
$b_{L,i}$, $b_{H,i}$	Zero-predictor coefficients
d_L, d_{Lt}, d_H	Quantized difference signals (non truncated and truncated)
S_{Lz}, S_{Hz}	Zero-predictor output signals
S_L, S_H	Predictor output signals
e_L, e_H	Difference signals to be quantized
∇_L, ∇_H	Logarithmic quantizer scale factors
Δ_L, Δ_H	Quantizer scale factor (linear)
I_L, I_{Lt}, I_H	Codewords (non truncated and truncated)
P_{Lt}, P_H	Partially reconstructed signals
I_{Lr}	Received lower sub-band codeword
X_{out}	Output value (uniform)

Note - Variables used exclusively within one section are not listed. Subscripts L and H refer to lower sub-band and higher sub-band values. Subscript Lt denotes values generated from the truncated 4-bit codeword as opposed to the nontruncated 6-bit (encoder) or 6-, 5- or 4-bit (decoder) codewords.

TABLE 4/G.722

Transmit and receive OMF coefficient values

h_0 , h_{23}	0.366211E-03
h_1 , h_{22}	-0.134277E-02
h_2 , h_{21}	-0.134277E-02
h_3 , h_{20}	0.646973E-02
h_4 , h_{19}	0.146484E-02
h_5 , h_{18}	-0.190430E-01
h_6 , h_{17}	0.390625E-02
h_7 , h_{16}	0.441895E-01
h_8 , h_{15}	-0.256348E-01
h_9 , h_{14}	-0.982666E-01
h_{10} , h_{13}	0.116089E+00
h_{11} , h_{12}	0.473145E+00

3.3 *Adaptive quantizer*

The difference signals, $e_L(n)$ and $e_H(n)$, are quantized to 6 and 2 bits for the lower and higher sub-bands respectively. Tables 5/G.722 and 6/G.722 give the decision levels and the output codes for the 6- and 2-bit quantizers respectively. In these tables, only the positive decision levels are indicated, the negative levels can be determined by symmetry. m_L *and* m_H are indices for the quantizer intervals. The interval boundaries, *LL*6, *LU*6, *HL* and *HU*, are scaled by computed scale factors, $\Delta_L(n)$ and $\Delta_H(n)$ (see § 3.5). Indices, m_L and m_H, are then determined to satisfy the following:

$$LL6(m_L) \cdot \Delta_L(n) \le e_L(n) < LU6(m_L) \cdot \Delta_L(n) \tag{3-7}$$

$$HL(m_H) \cdot \Delta_H(n) \le e_H(n) < HU(m_H) \cdot \Delta_H(n) \tag{3-8}$$

for the lower and higher sub-bands respectively.

The output codes, *ILN and IHN*, represent negative intervals, whilst the output codes, *ILP and IHP*, represent positive intervals. The output codes, $I_L(n)$ and $I_H(n)$, are then given by:

$$I_L(n) = \begin{cases} ILP(m_L), \text{ if } e_L(n) \ge 0 \\ ILN(m_L), \text{ if } e_L(n) < 0 \end{cases} \tag{3-9}$$

$$I_H(n) = \begin{cases} IHP(m_H), \text{ if } e_H(n) \ge 0 \\ IHN(m_H), \text{ if } e_H(n) < 0 \end{cases} \tag{3-10}$$

for the lower and higher sub-bands respectively.

TABLE 5/G.722

Decision levels and output codes for the 6-bit lower sub-band quantizer

m_L	LL6	LU6	ILN	ILP
1	0.00000	0.06817	111111	111101
2	0.06817	0.14103	111110	111100
3	0.14103	0.21389	011111	111011
4	0.21389	0.29212	011110	111010
5	0.29212	0.37035	011101	111001
6	0.37035	0.45482	011100	111000
7	0.45482	0.53929	011011	110111
8	0.53929	0.63107	011010	110110
9	0.63107	0.72286	011001	110101
10	0.72286	0.82335	011000	110100
11	0.82335	0.92383	010111	110011
12	0.92383	1.03485	010110	110010
13	1.03485	1.14587	010101	110001
14	1.14587	1.26989	010100	110000
15	1.26989	1.39391	010011	101111
16	1.39391	1.53439	010010	101110
17	1.53439	1.67486	010001	101101
18	1.67486	1.83683	010000	101100
19	1.83683	1.99880	001111	101011
20	1.99880	2.19006	001110	101010
21	2.19006	2.38131	001101	101001
22	2.38131	2.61482	001100	101000
23	2.61482	2.84833	001011	100111
24	2.84833	3.14822	001010	100110
25	3.14822	3.44811	001001	100101
26	3.44811	3.86796	001000	100100
27	3.86796	4.28782	000111	100011
28	4.28782	4.99498	000110	100010
29	4.99498	5.70214	000101	100001
30	5.70214	∞	000100	100000

Note - If a transmitted codeword for the lower sub-band signal has been transformed, due to transmission errors to one of the four suppressed codewords "0000XX", the received code word is set at "111111".

TABLE 6/G.722

Decision levels and output codes for the 2-bit higher sub-band quantizer

m_H	HL	HH	IHN	IHP
1	0	1.10156	01	11
2	1.10156	∞	00	10

3.4 *Inverse adaptive quantizers*

3.4.1 *Inverse adaptive quantizer in the lower sub-band ADPCM encoder*

The lower sub-band output code, $I_L(n)$, is truncated by two bits to produce $I_{Lt}(n)$. The 4-bit code-word, $I_{Lt}(n)$, is converted to the truncated quantized difference signal, $d_{Lt}(n)$, using the $QL4^{-1}$ output values of Table 7/G.722, and scaled by the scale factor, $\Delta_L(n)$:

$$d_{Lt}(n) = QL4^{-1}[I_{Lt}(n)] \cdot \Delta_L(n) \cdot \mathrm{sgn}[I_{Lt}(n)] \quad (3\text{-}11)$$

where sgn $[I_{Lt}(n)]$ is derived from the sign of $e_L(n)$ defined in Equation 3-9.

There is a unique mapping, shown in Table 7/G.722, between four adjacent 6-bit quantizer intervals and the $QL4^{-1}$ output values. $QL4^{-1}[I_{Lt}(n)]$ is determined in two steps: first determination of the quantizer interval index, m_L, corresponding to $I_L(n)$ from Table 5/G.722, and then determination of $QL4^{-1}(m_L)$ by reference to Table 7/G.722.

TABLE 7/G.722

Output values and multipliers for 6, 5 and 4-bit lower sub-band inverse quantizers

m_L	$QL6^{-1}$	$QL5^{-1}$	$QL4^{-1}$	W_L
1	0.03409	0.06817	0.0000	-0.02930
2	0.10460			
3	0.17746	0.21389		
4	0.25300		0.29212	-0.01465
5	0.33124	0.37035		
6	0.41259			
7	0.49706	0.53929		
8	0.58518		0.63107	0.02832
9	0.67697	0.72286		
10	0.77310			
11	0.87359	0.92383		
12	0.97934		1.03485	0.08398
13	1.09036	1.14587		
14	1.20788			
15	1.33191	1.39391		
16	1.46415		1.53439	0.16309
17	1.60462	1.67486		
18	1.75585			
19	1.91782	1.99880		
20	2.09443		2.19006	0.26270
21	2.28568	2.38131		
22	2.49806			
23	2.73157	2.84833		
24	2.99827		3.14822	0.58496
25	3.29816	3.44811		
26	3.65804			
27	4.07789	4.28782		
28	4.64140		4.99498	1.48535
29	5.34856	5.70214		
30	6.05572			

3.4.2 *Inverse adaptive quantizer in the higher sub-band ADPCM encoder*

The higher sub-band output code, $I_H(n)$ is converted to the quantized difference signal, $d_H(n)$, using the $Q2^{-1}$ output values of Table 8/G.722 and scaled by the scale factor, $\Delta_H(n)$:

$$d_H(n) = Q2^{-1}\left[I_H(n)\right] \cdot \Delta_H(n) \cdot \mathrm{sgn}\left[I_H(n)\right] \qquad (3\text{-}12)$$

where $\mathrm{sgn}[I_H(n)]$ is derived from the sign of $e_H(n)$ defined in Equation (3-10), and where $Q_2^{-1}[I_H(n)]$ is determined in two steps: first determine the quantizer interval index, m_H, corresponding to $I_H(n)$ from Table 6/G.722 and then determine $Q2^{-1}(m_H)$ by reference to Table 8/G.722.

TABLE 8/G.722

Output values and multipliers for the 2-bit higher sub-band quantizer

m_H	$Q2^{-1}$	W_H
1 2	0.39453 1.80859	-0.10449 0.38965

3.5 *Quantizer adaptation*

This block defines $\Delta_L(n)$ and $\Delta_H(n)$, the scaling factors for the lower and higher sub-band quantizers. The scaling factors are updated in the log domain and subsequently converted to a linear representation. For the lower sub-band, the input is $I_{Lt}(n)$, the codeword truncated to preserve the four most significant bits. For the higher sub-band, the 2-bit quantizer output, $I_H(n)$, is used directly.

Firstly the log scaling factors, $\Delta_L(n)$ and $\Delta_H(n)$, are updated as follows:

$$\nabla_L(n) = \beta \cdot \nabla_L(n-1) + W_L\left[I_{Lt}(n-1)\right] \tag{3-13}$$

$$\nabla_H(n) = \beta \cdot \nabla_H(n-1) + W_H\left[I_H(n-1)\right] \tag{3-14}$$

where W_L and W_H are logarithmic scaling factors multipliers given in Tables 7/G.722 and 8/G.722, and B is a leakage constant equal to 127/128.

Then the log scaling factors are limited, according to:

$$0 \le \nabla_L(n) \le 9 \tag{3-15}$$

$$0 \le \nabla_H(n) \le 11 \tag{3-16}$$

Finally, the linear scaling factors are computed from the log scaling factors, using an approximation of the inverse $\log_2$ function:

$$\Delta_L(n) = 2\left[\nabla_L(n) + 2\right] \cdot \Delta_{min} \tag{3-17}$$

$$\Delta_H(n) = 2\nabla_H(n) \cdot \Delta_{min} \tag{3-18}$$

where Δ_{min} is equal to half the quantizer step size of the 14 bit analogue-to-digital converter.

3.6 *Adaptive prediction*

3.6.1 *Predicted value computations*

The adaptive predictors compute predicted signal values, $s_L(n)$ and $s_H(n)$, for the lower and higher sub-bands respectively.

Each adaptive predictor comprises two sections: a second-order section that models poles, and a sixth-order section that models zeroes in the input signal.

The second order pole sections (coefficients $a_{L,i}$ and $a_{H,}i$) use the quantized reconstructed signals, $rL_t(n)$ and $r_H(n)$, for prediction. The sixth order zero sections (coefficients $b_{L,i}$) and $b_{H,}i$) use the quantized difference signals, $d_{Lt}(n)$ and $d_H(n)$. The zero-based predicted signals, $s_{Lz}(n)$ *and* $s_{Hz}(n)$, are also employed to compute partially reconstructed signals as described in § 3.6.2.

Firstly, the outputs of the pole sections are computed as follows:

$$s_{Lp} = \sum_{i=1}^{2} a_{L,i}(n-1) \cdot r_{Lt}(n-i) \tag{3-19}$$

$$s_{Hp} = \sum_{i=1}^{2} a_{H,i}(n-1) \cdot r_{H}(n-i) \tag{3-20}$$

Similarly, the outputs of the zero sections are computed as follows:

$$s_{Lz}(n) = \sum_{i=1}^{6} b_{L,i}(n-1) \cdot d_{Lt}(n-i) \tag{3-21}$$

$$s_{Hz}(n) = \sum_{i=1}^{6} b_{H,i}(n-1) \cdot d_{H}(n-i) \tag{3-22}$$

Then, the intermediate predicted values are summed to produce the predicted signal values:

$$s_L(n) = s_{Lp} + s_{Lz}(n) \tag{3-23}$$

$$s_H(n) = s_{Hp} + s_{Hz}(n) \tag{3-24}$$

3.6.2 *Reconstructed signal computation*

The quantized reconstructed signals, $r_{Lt}(n)$ and $r_H(n)$, are computed as follows:

$$r_{Lt}(n) = s_L + d_{Lt}(n) \tag{3-25}$$

$$r_H(n) = s_H + d_H(n) \tag{3-26}$$

The partially reconstructed signals, $p_{Lt}(n)$ and $p_{\mathrm{H}}(n)$, used for the pole section adaptation, are then computed:

$$p_{Lt}(n) = d_{Lt}(n) + s_{Lz}(n) \tag{3-27}$$

$$p_H(n) = d_H(n) + s_{Hz}(n) \tag{3-28}$$

3.6.3 *Pole section adaptation*

The second order pole section is adapted by updating the coefficients, $a_{L,1}$, $a_{H,1}$, $a_{H,2}$, using a simplified gradient algorithm:

$$a_{L,1}(n) = (1-2^{-8})a_{L,1}(n-1) + 3 \cdot 2^{-8} \cdot p_A \tag{3-29}$$

$$a_{L,2}(n) = (1-2^{-7})a_{L,2}(n-1) + 2^{-7} \cdot p_B - 2^{-7} \cdot f \cdot p_A \tag{3-30}$$

where

$$p_A = \mathrm{sgn2}\left[p_{Lt}(n)\right] \cdot \mathrm{sgn2}\left[p_{Lt}(n-1)\right] \tag{3-31}$$

$$p_B = \mathrm{sgn2}\left[p_{Lt}(n)\right] \cdot \mathrm{sgn2}\left[p_{Lt}(n-2)\right] \tag{3-32}$$

with

$$\mathrm{sgn2}(q) = \begin{cases} +1, & q \ge 0 \\ -1, & q < 0 \end{cases} \tag{3-33}$$

and

$$f = \begin{cases} 4a_{L,1}(n-1), & \left|a_{L,1}\right| \le 1/2 \\ 2\,\mathrm{sgn}\left[a_{L,1}(n-1)\right], & \left|a_{L,1}\right| > 1/2 \end{cases} \tag{3-34}$$

Then the following stability constraints are imposed:

$$\left|a_{L,2}\right| \le 0{,}75 \tag{3-35}$$

$$\left|a_{L,1}\right| \le 1 - 2^{-4} - a_{L,2} \tag{3-36}$$

$a_{H,1}(n)$ and $a_{H,2}(n)$ are similarly computed, replacing $a_{L,1}(n)$, $a_{L,2}(n)$ and $P_{Lt}(n)$, by $a_{H,1}(n)$, $a_{H,2}(n)$ and $P_H(n)$, respectively.

3.6.4 *Zero section adaptation*

The sixth order zero predictor is adapted by updating the coefficients $b_{L,i}$ and $b_{H,i}$ using a simplified gradient algorithm:

$$b_{L,i}(n) = (1 - 2^{-8})b_{L,i}(n-1) + 2^{-7}\,\mathrm{sgn}\,3\left[d_{Lt}(n)\right].\ \mathrm{sgn}\,2\left[d_{Lt}(n-i)\right] \tag{3-37}$$

for $i = 1, 2 \dots 6$

and with

$$\mathrm{sgn3}(q) = \begin{cases} +1, & q > 0 \\ 0, & q = 0 \\ -1, & q < 0 \end{cases} \tag{3-38}$$

where $b_{L,i}(n)$ is implicitly limited to ± 2.

$b_{H,i}(n)$ are similarly updated, replacing $b_{L,i}(n)$ and $d_{Lt}(n)$ by $b_{H,i}(n)$ and $d_H(n)$ respectively.

4 SB-ADPCM decoder principles

A block diagram of the SB-ADPCM decoder is given in Figure 6/G.722 and block diagrams of the lower and higher sub-band ADPCM decoders are given respectively in Figures 7/G.722 and 8/G.722.

The input to the lower sub-band ADPCM decoder, $I_{L,r,}$, may differ from I_L even in the absence of transmission errors, in that one or two least significant bits may have been replaced by data.

4.1 *Inverse adaptive quantizer*

4.1.1 *Inverse adaptive quantizer selection for the lower sub-band ADPCM decoder*

According to the received indication of the mode of operation the number of least significant bits which should be truncated from the input codeword I_{Lr}, and the choice of the inverse adaptive quantizer are determined, as shown in Table 2/G.722.

For operation in mode 1, the 6-bit codeword, $I_{Lr}(n)$, is converted to the quantized difference, $d_L(n)$, according to $QL6^{-1}$ output values of Table 7/G.722, and scaled by the scale factor, $\Delta_L(n)$:

$$d_L(n) = QL6^{-1}\left[I_{Lr}(n)\right] \cdot \Delta_L(n) \cdot \operatorname{sgn}\left[I_{Lr}(n)\right] \tag{4-1}$$

where $\operatorname{sgn}[I_{Lr}(n)]$ is derived from the sign of $I_L(n)$ defined in equation (3-9).

Similarly, for operations in mode 2 or mode 3, the truncated codeword (by one or two bits) is converted to the quantized difference signal, $d_L(n)$, according to $QL5^{-1}$ or $QL4^{-1}$ output values of Table 7/G.722 respectively.

There are unique mappings, shown in Table 7/G.722, between two or four adjacent 6-bit quantizer intervals and the $QL5^{-1}$ or $QL4^{-1}$ output values respectively.

In the computations above, the output values are determined in two steps: first determination of the quantizer interval index, m_L, corresponding to $I_{Lr}(n)$ from Table 5/G.722, and then determination of the output values corresponding to m_L by reference to Table 7/G.722.

The inverse adaptive quantizer, used for the computation of the predicted value and for adaptation of the quantizer and predictor, is described in § 3.4.1, but with $I_L(n)$ replaced by $I_{Lr}(n)$.

4.1.2 *Inverse adaptive quantizer for the higher sub-band ADPCM decoder*

See § 3.4.2.

4.2 *Quantizer adaptation*

See § 3.5.

4.3 *Adaptive prediction*

4.3.1 *Predicted value computation*

See § 3.6.1.

4.3.2 *Reconstructed signal computation*

See § 3.6.2.

The output reconstructed signal for the lower sub-band ADPCM decoder, $r_L(n)$, is computed from the quantized difference signal, $d_L(n)$, as follows:

$$r_L(n) = s_L(n) + d_L(n) \tag{4-2}$$

4.3.3 *Pole section adaptation*

See § 3.6.3.

4.3.4 *Zero section adaptation*

See § 3.6.4.

4.4 *Receive QMF*

A 24-coefficient QMF is used to reconstruct the output signal, $x_{\text{out}}(j)$, from the reconstructed lower and higher sub-band signals, $r_L(n)$ and $r_H(n)$. The QMF coefficient values, h_i, are the same as those used in the transmit QMF and are given in Table 4/G.722.

The output signals, $x_{out}(j)$ and $x_{out}(j + 1)$, are computed in the following way:

$$x_{out}(j) = 2\sum_{i=0}^{11} h_{2i} \cdot x_d(i) \tag{4-3}$$

$$x_{out}(j+1) = 2\sum_{i=0}^{11} h_{2i+1} \cdot x_s(i) \tag{4-4}$$

where

$$x_d(i) = r_L(n-i) - r_H(n-i) \tag{4-5}$$

$$x_s(i) = r_L(n-i) + r_H(n-i) \tag{4-6}$$

5 Computational details for QMF

5.1 *Input and output signals*

Table 9/G.722 defines the input and output signals for the transmit and receive QMF. All input and output signals have 16-bit word lengths, which are limited to a range of - 16384 to 16383 in 2's complement notation. Note that the most significant magnitude bit of the A/D output and the D/A input appears at the third bit location in XIN and XOUT, respectively.

TABLE 9/G.722

Representation of input and output signals

Transmit QMF			
	Name	Binary representation	Description
Input Output Output	XIN XL XH	S, S, -2, -3, . . ., -14, -15 S, S, -2, -3, . . ., -14, -15 S, S, -2, -3, . . ., -14, -15	Input value (uniformly quantized) Output signal for lower sub-band encoder Outband signal for higher sub-band encoder
Receive QMF			
	Name	Binary representation	Description
Input Input Output	RL RH XOUT	S, S, -2, -3, . . ., -14, -15 S, S, -2, -3, . . ., -14, -15 S, S, -2, -3, . . ., -14, -15	Lower sub-band reconstructed signal Higher sub-band reconstructed signal Output value (uniformly quantized)

Note - XIN and XOUT are represented in a sign-extended 15-bit format, where the LSB is set to "0" for 14-bit converters.

5.2 *Description of variables and detailed specification of sub-blocks*

This section contains a detailed expansion of the transmit and receive QMF. The expansions are illustrated in Figures 17/G.722 and 18/G.722 with the internal variables given in Table 10/G.722, and the QMF coefficients given in Table 11/G.722. The word lengths of internal variables, XA, XB and WD must be equal to or greater than 24 bits (see Note). The other internal variables have a minimum of 16 bit word lengths. A brief functional description and the full specification is given for each sub-block.

The notations used in the block descriptions are as follows:

>> n	denotes an *n*-bit arithmetic shift right operation (sign extension),
+	denotes arithmetic addition with saturation control which forces the result to the minimum or maximum representable value in case of underflow or overflow, respectively,
-	denotes arithmetic subtraction with saturation control which forces the result to the minimum or maximum representable value in case of underflow or overflow, respectively.
*	denotes arithmetic multiplication which can be performed with either truncation or rounding,
<	denotes the "less than" condition as $x < y$; x is less than y,
>	denotes the "greater than" condition, as $x > y$; x is greater than y,
=	denotes the substitution of the right-hand variable for the left-hand variable.

Note 1 - Some freedom is offered for the implementation of the accumulation process in the QMF: the word lengths of the internal variables can be equal to or greater than 24 bits, and the arithmetic multiplications can be performed with either truncation or rounding. It allows a simplified implementation on various types of processors. The counterpart is that it excludes the use of digital test sequence for the test of the QMF.

TABLE 10/G.722

Representation of internal processing variables and QMF coefficients

Transmit QMF		
Name	Binary representation	Description
XA	S, -1, -2, -3, . . ., -y+1, -y	Output signal of sub-block, ACCUMA
XB	S, -1, -2, -3, . . ., -y+1, -y	Output signal of sub-block, ACCUMB
XIN1, XIN2, . . ., XIN23	S, S, -2, -3, . . ., -14, -15	Input signal with delays 1 to 23
Receive QMF		
Name	Binary representation	Description
XD, XD1, . . ., XD11	S, -1, -2, -3, . . ., -14, -15	Input signal for sub-block, ACCUMC, with delays 0 to 11
XOUT1	S, S, -2, -3, . . ., -14, -15	8 kHz sampled output value
XOUT2	S, S, -2, -3, . . ., -14, -15	8 kHz sampled output value
XS, XS1, . . ., XS11	S, -1, -2, -3, . . ., -14, -15	Input signal for sub-block, ACCUMD, with delays 0 to 11
WD	S, -1, -2, -3, . . ., -y+1, -y	Partial sum
QMF coefficient		
Name	Binary representation	Description
H0, H1, . . ., H23	S, -2, -3, -4, . . ., -12, -13	Filter coefficient values

Note - y is equal to or greater than 23.

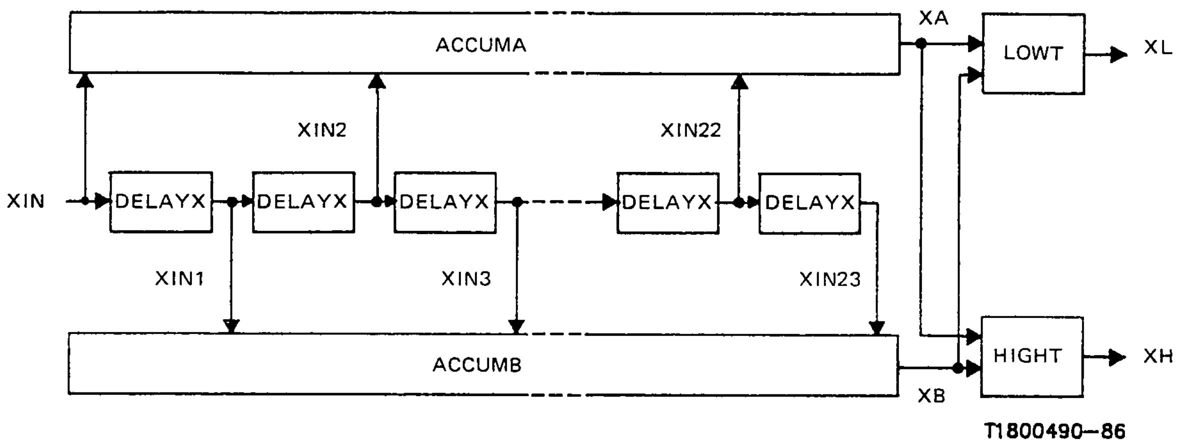
TABLE 11/G.722

QMF coefficient

Coefficient	Scaled values (see Note)
H0, H23	3
H1, H22	-11
H2, H21	-11
H3, H20	53
H4, H19	12
H5, H18	-156
H6, H17	32
H7, H16	362
H8, H15	-210
H9, H14	-805
H10, H13	951
H11, H12	3876

Note - QMF coefficients are scaled by 2^{13} with respect to the representation specified in Table 10/G.722.

5.2.1 *Description of the transmit QMF*



Note – XA and XB are computed with respect to the same 16 kHz input sampling instants and are output at an 8 kHz sampling rate.

FIGURE 17/G.722

Transmit QMF

DELAYX

Input: x

Output: y

Note - Index (j) indicates the current 16-kHz sample period, while index (j - 1) indicates the previous one.

Function: Memory block. For any input x, the output is given by:

$y(j) = x(j - 1)$

ACCUMA

Inputs: XIN, XIN2, XIN4,..., XIN22

Output: XA

Note 1 - H0, H2, ..., H22 are obtained from Table 11/G.722.

Note 2 - The values XIN, XIN2, ..., XIN22 and H0, H2, ..., H22 may be shifted before multiplication, if so desired. The result XA must be rescaled accordingly, In performing these scaling operations the following rules must be obeyed:

1) the precision of XIN, XIN2, ..., XIN22 and H0, H2, ..., H22 as given in Table 9/G.722 and Table 10/G.722 must be retained,

2) the partial products and the ouptut signal XA must be retained to a significance of at least 2^{-23},

3) no saturation should occur in the calculation of the function XA.

Note 3 - No order of summation is specified in accumulating the partial products.

Function: Multiply the even order QMF coefficients by the appropriately delayed input signals, and accumulate these products.

XA = (XIN $*$ H0) + (XIN2 $*$ H2) + (XIN4 $*$ H4) + ... + (XIN22 $*$ H22)

ACCUMB

Inputs: XIN1, XIN3, XIN5,. . . , XIN23

Output: XB

Note 1 - H1, H3, . . ., H23 are obtained from Table 11/G.722.

Note 2 - The values XIN1, XIN3, ..., XIN23 and HI, H3, ..., H23 may be shifted before multiplication, if so desired. The result XB must be rescaled accordingly. In performing these scaling operations the following rules must be obeyed:

1) the precision of XIN1, XIN3, ..., XIN23 and HI, H3, ..., H23 as given in Table 9/G.722 and Table 10/G.722 must be retained,

2) the partial products and the output signal X3 must be retained to a significance of at least 2^{-23},

3) no saturation should occur in the calculation of the function XB.

Note 3 - No order of summation is specified in accumulating the partial products.

Function: Multiply the odd order QMF coefficients by the appropriately delayed input signals, and accumulate these products.

XB = (XIN1 $*$ H1) + (XIN3 $*$ H3) + (XIN5 $*$ H5) + ... + (XIN23 $*$ H23)

LOWT

Inputs: XA, XB

Output: XL

Function: Compute the lower sub-band signal component.

XL = (XA + XB) >> (y - 15)

$$XL = \begin{cases} 16383, & \text{if } XL > 16383 \\ -16384, & \text{if } XL < -16384 \end{cases}$$

HIGHT

Inputs: XA, XB

Output: XH

Function Compute the higher sub-band signal component.

XH = (XA - XB) >> (y - 15)

$$XH = \begin{cases} 16383, & \text{if } XH > 16383 \\ -16384, & \text{if } XH < -16384 \end{cases}$$

5.2.2 *Description of the receive QMF*

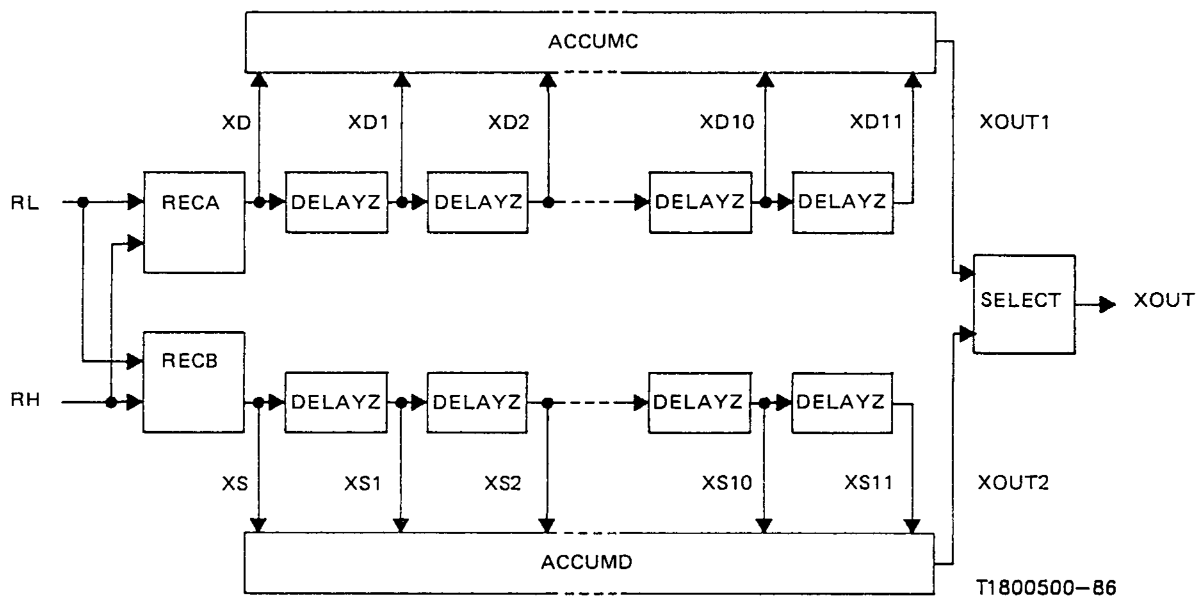


FIGURE 18/G.722

Receive QMF

RECA

Inputs: RL, RH

Output: XD

Function: Compute the input signal to the receive QMF

XD = RL - RH

RECB

Inputs: RL, RH

Output: XS

Function: Compute the input signal to the receive QMF

XS = RL + RH

DELAYZ

Input: x

output: y

Note - Index (*n*) indicates the current 8-kHz sample period, while index (*n* - 1) indicates the previous one.

Function: Memory block. For any input x, the output is given by:

y (*n*) = x (*n* -1)

ACCUMC

Inputs: XD, XDi (i = 1 to 11)

Output: XOUT1

Note 1 - H0, H2, ..., H22 are obtained from Table 11/G.722.

Note 2 - The values XD, XD1, ..., XD11 and H0, H2, ..., H22 may be shifted before multiplication, if so desired. The result WD must be rescaled accordingly. In performing these scaling operations the following rules must be obeyed:

1) the precision of XD, XD1, ..., XD11 and H0, H2, ..., H22 as given in Table 9/G.722 and Table 10/G.722 must be retained,

2) the partial products and the output signal WD must be retained to a significance of at least 2^{-23};

3) no saturation should occur in the calculation of the function WD.

Note 3 - No order of summation is specified in accumulating the partial products.

Function: Multiply the even order QMF coefficients by the appropriately delayed input signals, and accumulate these products.

WD = (XD $*$ H0) + (XD1 $*$ H2) + (XD2 $*$ H4) + ... + (XD11 $*$ H22)

XOUT1 = WD >> (y - 16)

$$\text{XOUT1} = \begin{cases} 16383, & \text{if XOUT1} > 16383 \\ -16384, & \text{if XOUT1} < -16384 \end{cases}$$

ACCUMD

Inputs: XS, XSi (i = 1 to 11)

Output: XOUT2

Note 1 - H1, H3,..., H23 are obtained from Table 11/G.722.

Note 2 - The values XS, XS1, ..., XS11 and H1, H3, ..., H23 may be shifted before multiplication, if so desired. The result WD must be rescaled accordingly. In performing these scaling operations the following rules must be obeyed:

1) the precision of XS, XS1, ..., XS11 and H1, H3, ..., H23 as given in Table 9/G.722 and Table 10/G.722 must be retained,

2) the partial products and the output signal WD must be retained to a significance of at least 2^{-23};

3) no saturation should occur in the calculation of the function WD.

Note 3 - No order of summation is specified in accumulating the partial products.

Function: Multiply the odd order QMF coefficients by the appropriately delayed input signals, and accumulate these products.

$$WD = (XS * H1) + (XS1 * H3) + (XS2 * H5) + ... + (XS11 * H23)$$

XOUT2 = WD >> (y - 16)

$$XOUT2 = \begin{cases} 16383, & \text{if } XOUT2 > 16383 \\ -16384, & \text{if } XOUT2 < -16384 \end{cases}$$

SELECT

Inputs: XOUT1, XOUT2

Output: XOUT

Note 1 - Index (j) indicates the current 16-kHz sample period, while index (j + 1) indicates the next one. With respect to the input sampling instant XOUT1 is selected first, followed by XOUT2.

Function: Select one of the 8 kHz sampled input signals alternately to produce the 16 kHz sampled output signal.

XOUT(j) = XOUT1

XOUT(j + 1) = XOUT2

---

6 Computational details for lower and higher sub-band ADPCM

6.1 *Input and output signals*

Table 12/G.722 defines the input and output signals for the lower and higher sub-band encoders and decoders. The signal RS represents a reset function that sets all internal memory elements to a specified condition, so that encoders or decoders can be forced into a known state. The signal MODE represents a mode indication. The three basic modes of operation are described in Table 1/G.722. The mode identification is performed in every 8 kHz sampling interval.

TABLE 12/G.722

Input and output signals

Lower sub-band encoder		
	Name	Description
Input	XL	15-bit uniformly quantized input signal
Input	RS	Reset
Output	IL	6-bit ADPCM codeword
Higher sub-band encoder		
	Name	Description
Input	XH	15-bit uniformly quantized input signal
Input	RS	Reset
Output	IH	2-bit ADPCM codeword
Lower sub-band decoder		
	Name	Description
Input	ILR	Received 6-bit ADPCM codeword
Input	MODE	Mode indication
Input	RS	Reset
Output	RL	15-bit uniformly quantized output signal
Higher sub-band decoder		
	Name	Description
Input	IH	2-bit ADPCM codeword
Input	RS	Reset
Output	RH	15-bit uniformly quantized output signal

6.2 *Description of variables and detailed specification of sub-blocks*

This section contains a detailed expansion of all blocks in Figures 4/G.722, 5/G.722, 7/G.722 and 8/G.722 described in §§ 3 and 4. The expansions are illustrated in Figures 19/G.722 to 31/G.722 with the internal processing variables in Table 13/G.722, the constant values in Tables 14/G.722 and 15/G.722, and conversion tables in Tables 16/G.722 to 21/G.722. All internal variables have 16-bit word lengths, and are represented in 2's complement notation. Constant values with 13-bit precision as given in Tables 14/G.722 and 15/G.722 are used in sub-blocks with 16-bit representation, extending the sign to the first three MSBs. A brief functional description and full specification is given for each sub-block.

The notations used in the block descriptions are as follows:

$<< n$ denotes an n-bit arithmetic shift left operation (zero fill),

$>> n$ denotes an n-bit arithmetic shift right operation (sign extension); if n is negative, $>> n$ means $<< (- n)$;

$>>> n$ denotes an n-bit logical shift right operation (zero fill),

$<<< n$ denotes an n-bit logical shift left operation (zero fill),

& denotes the logical "and" operation,

+ denotes arithmetic addition. (The result is set at +32767 when overflow occurs, or at -32768 when underflow occurs.),

- denotes arithmetic subtraction. (The result is set at +32767 when overflow occurs, or at -32768 when underflow occurs.),

$*$ denotes the multiplication defined by the following arithmetic operation:

$A * B = (A \text{ times } B) >> 15;$

= = denotes the "equal to" condition,

! = denotes the "not equal to" condition,

< denotes the "less than" condition, as $x < y$; x is less than y;

> denotes the "greater than" condition, as $x > y$; x is greater than y;

= denotes the substitution of right-hand variable for the left-hand variable,

| delineates comments to equations.

TABLE 13/G.722

Internal processing variables

Lower sub-band ADPCM		
Name	Binary representation	Description
AL1*, AL2*	S, 0, -1, -2, ..., -13, -14	Delayed second-order pole section coefficients
APL1,APL2	S, 0, -1, -2, ..., -13, -14	Second-order pole section coefficient
BL1*,..., BL6*	S, 0, -1, -2, ..., -13, -14	Delayed sixth-order zero section coefficients
BPL1,..., BPL6	S, 0, -1, -2, ..., -13, -14	Sixth-order zero section coefficients
DEPL	S, -4, -5, -6, ..., -17, -18	Quantizer scale factor
DETL*	S, -4, -5, -6, ..., -17, -18	Delayed quantizer scale factor
DLT	S, -1, -2, -3, ..., -14, -15	Quantized difference signal for the adaptative predictor with delay 0
DLT1*,..., DLT6*	S, -1, -2, -3, ..., -14, -15	Quantized difference signal for the adaptative predictor with delays 1 to 6
DL	S, -1, -2, -3, ..., -14, -15	Quantized difference signal for decoder output
EL	S, -1, -2, -3, ..., -14, -15	Difference signal
NBL*	S, 3, 2, 1,0, ..., -10, -11	Delayed logarithmic quantizer scale factor
NBPL	S, 3, 2, 1,0, ..., -10, -11	Logarithmic quantizer scale factor
PLT	S, -1, -2, -3, ..., -14, -15	Partially reconstructed signal with delay 0
PLT1*, PLT2*	S, -1, -2, -3, ..., -14, -15	Partially reconstructed signal with delays 1 and 2
YL	S, -1, -2, -3, ..., -14, -15	Output reconstructed signal
RLT	S, -1, -2, -3, ..., -14, -15	Reconstructed signal for the adaptive predictor with delay 0
RLT1*, RLT2*	S, -1, -2, -3, ..., -14, -15	Reconstructed signal for the adaptive predictor with delay 1 and 2
SL	S, -1, -2, -3, ..., -14, -15	Predictor output value
SPL	S, -1, -2, -3, ..., -14, -15	Pole section output signal
SZL	S, -1, -2, -3, ..., -13, -14	Zero section output signal
AH1*, AH2*	S, 0, -1, -2, ..., -13, -14	Delayed second-order pole section coefficients
APH1, APH2*	S, 0, -1, -2, ..., -13, -14	Second-order pole section coefficients
BH1*,..., BH6*	S, 0, -1, -2, ..., -13, -14	Delayed sixth-order zero section coefficients
BPH1,..., BPH6	S, 0, -1, -2, ..., -13, -14	Sixth-order zero section coefficients
DEPH	S, -4, -5, -6, ..., -17, -18	Quantizer scale factor
DETH*	S, -4, -5, -6, ..., -17, -18	Delayed quantizer scale factor
DH	S, -1, -2, -3, ..., -14, -15	Quantizer difference signal with delay 0
DH1*,..., DH6*	S, -1, -2, -3, ..., -14, -15	Quantized difference signal with delays 1 to 6
EH	S, -1, -2, -3, ..., -14, -15	Difference signal
NBH	S, 3, 2, 1, 0, ..., -10, -11	Delayed logarithmic quantizer scale factor
NBPH	S, 3, 2, 1, 0, ..., -10, -11	Logarithmic quantizer scale factor
PH	S, -1, -2, -3, ..., -14, -15	Partially reconstructed signal with delay 0
PH1*, PH2*	S, -1, -2, -3, ..., -14, -15	Partially reconstructed signal with delays 1 and 2
YH	S, -1, -2, -3, ..., -14, -15	Quantized reconstructed signal with delay 0
RH1*, RH2*	S, -1, -2, -3, ..., -14, -15	Quantized reconstructed signal with delays 1 and 2
SH	S, -1, -2, -3, ..., -14, -15	Predictor output value
SPH	S, -1, -2, -3, ..., -14, -15	Pole section output signal
SZH	S, -1, -2, -3, ..., -14, -15	Zero section output signal

Note - * indicates variables which should be initialized to a specific value when a reset condition is applied.

TABLE 14/G.722

Quantizer decision levels and output values

Quantizer constant representation		
Name	Binary representation	Description
Qi	S, 2, 1, 0, -1, ..., -8, -9	Quantizer decision level
QQi	S, 2, 1, 0, -1, ..., -8, -9	Inverse quantizer output
WL, WH	S, 0, -1, -2, ..., -10, -11	Logarithmic scaling factor multiplier

Lower sub-band quantizer					
Address	Q6	QQ6	QQ5	QQ4	WL
0					
1	35	17	35	0	-60
2	72	54	110	150	-30
3	110	91	190	323	58
4	150	130	276	530	172
5	190	170	370	786	334
6	233	211	473	1121	538
7	276	254	587	1612	1198
8	323	300	714	2557	3042
9	370	347	858		
10	422	396	1023		
11	473	447	1219		
12	530	501	1458		
13	587	558	1765		
14	650	618	2195		
15	714	682	2919		
16	786	750			
17	858	822			
18	940	899			
19	1023	982			
20	1121	1072			
21	1219	1170			
22	1339	1279			
23	1458	1399			
24	1612	1535			
25	1765	1689			
26	1980	1873			
27	2195	2088			
28	2557	2376			
29	2919	2738			
30		3101			

Higher sub-band quantizer			
Address	Q2	QQ2	WH
1	564	202	-214
2		926	798

TABLE 15/G.722

Log-to-linear conversion table

Conversion table constants		
Name	Binary representation	Description
ILA	S, -5, -6, -7, ..., -15, -16	353-entry table constants
ILB	S, 0, -1, -2, ..., -15, -16	32-entry table constants

ILA

i \ j	0	1	2	3	4	5	6	7
0	1	1	1	1	1	1	1	1
8	1	1	1	1	1	1	1	1
16	1	1	1	2	2	2	2	2
24	2	2	2	2	2	2	2	2
32	3	3	3	3	3	3	3	3
40	3	3	3	4	4	4	4	4
48	4	4	4	5	5	5	5	5
56	5	5	6	6	6	6	6	6
64	7	7	7	7	7	7	8	8
72	8	8	8	9	9	9	9	10
80	10	10	10	11	11	11	11	12
88	12	12	13	13	13	13	14	14
96	15	15	15	16	16	16	17	17
104	18	18	18	19	19	20	20	21
112	21	22	22	23	23	24	24	25
120	25	26	27	27	28	28	29	30
128	31	31	32	33	33	34	35	36
136	37	37	38	39	40	41	42	43
144	44	45	46	47	48	49	50	51
152	52	54	55	56	57	58	60	61
160	63	64	65	67	68	70	71	73
168	75	76	78	80	82	83	85	87
176	89	91	93	95	97	99	102	104
184	106	109	111	113	116	118	121	124
192	127	129	132	135	138	141	144	147
200	151	154	157	161	165	168	172	176
208	180	184	188	192	196	200	205	209
216	214	219	223	228	233	238	244	249
224	255	260	266	272	278	284	290	296
232	303	310	316	323	331	338	345	353
240	361	369	377	385	393	402	411	420
248	429	439	448	458	468	478	489	500
256	511	522	533	545	557	569	582	594
264	607	621	634	648	663	677	692	707
272	723	739	755	771	788	806	823	841

TABLE 15/G.722 *(cont.)*

ILA								
i \ j	0	1	2	3	4	5	6	7
280	860	879	898	918	938	958	979	1001
288	1023	1045	1068	1092	1115	1140	1165	1190
296	1216	1243	1270	1298	1327	1356	1386	1416
304	1447	1479	1511	1544	1578	1613	1648	1684
312	1721	1759	1797	1837	1877	1918	1960	2003
320	2047	2092	2138	2185	2232	2281	2331	2382
328	2434	2488	2542	2598	2655	2713	2773	2833
336	2895	2959	3024	3090	3157	3227	3297	3370
344	3443	3519	3596	3675	3755	3837	3921	4007
352	4095							

ILB								
i \ j	0	1	2	3	4	5	6	7
0	2048	2093	2139	2186	2233	2282	2332	2383
8	2435	2489	2543	2599	2656	2714	2774	2834
16	2896	2960	3025	3091	3158	3228	3298	3371
24	3444	3520	3597	3676	3756	3838	3922	4008

Note 1 - A table address is obtained by adding i and j.

Note 2 - Either a 353-entry or a 32-entry table may be used in accordance with the choice of log-to-linear conversion method, Method 1 or Method 2 (see §§ 6.2.1.3 and 6.2.2.3)

TABLE 16/G.722

Conversion from quantizer intervals to 6-bit output codewords

SIL	MIL	IL
-1	30	000100
-1	29	000101
-1	28	000110
-1	27	000111
-1	26	001000
-1	25	001001
-1	24	001010
-1	23	001011
-1	22	001100
-1	21	001101
-1	20	001110
-1	19	001111
-1	18	010000
-1	17	010001
-1	16	010010
-1	15	010011
-1	14	010100
-1	13	010101
-1	12	010110
-1	11	010111
-1	10	011000
-1	9	011001
-1	8	011010
-1	7	011011
-1	6	011100
-1	5	011101
-1	4	011110
-1	3	011111
-1	2	111110
-1	1	111111

SIL	MIL	IL
0	1	111101
0	2	111100
0	3	111011
0	4	111010
0	5	111001
0	6	111000
0	7	110111
0	8	110110
0	9	110101
0	10	110100
0	11	110011
0	12	110010
0	13	110001
0	14	110000
0	15	101111
0	16	101110
0	17	101101
0	18	101100
0	19	101011
0	20	101010
0	21	101001
0	22	101000
0	23	100111
0	24	100110
0	25	100101
0	26	100100
0	27	100011
0	28	100010
0	29	100001
0	30	100000

TABLE 17/G.722

Conversion from 4-bit codewords to quantizer intervals

RIL	SIL	IL4
0000	0	0
0001	-1	7
0010	-1	6
0011	-1	5
0100	-1	4
0101	-1	3
0110	-1	2
0111	-1	1
1111	0	0
1110	0	1
1101	0	2
1100	0	3
1011	0	4
1010	0	5
1001	0	6
1000	0	7

Note - It is possible for the decoder to receive the code-word 0000 due to transmission errors.

TABLE 18/G.722

Conversion from 6-bit codewords to quantizer intervals

RIL	SIL	IL6	RIL	SIL	IL6
000000	-1	1	111110	-1	2
000001	-1	1	111111	-1	1
000010	-1	1	111101	0	1
000011	-1	1	111100	0	2
000100	-1	30	111011	0	3
000101	-1	29	111010	0	4
000110	-1	28	111001	0	5
000111	-1	27	111000	0	6
001000	-1	26	110111	0	7
001001	-1	25	110110	0	8
001010	-1	24	110101	0	9
001011	-1	23	110100	0	10
001100	-1	22	110011	0	11
001101	-1	21	110010	0	12
001110	-1	20	110001	0	13
001111	-1	19	110000	0	14
010000	-1	18	101111	0	15
010001	-1	17	101110	0	16
010010	-1	16	101101	0	17
010011	-1	15	101100	0	18
010100	-1	14	101011	0	19
010101	-1	13	101010	0	20
010110	-1	12	101001	0	21
010111	-1	11	101000	0	22
011000	-1	10	100111	0	23
011001	-1	9	100110	0	24
011010	-1	8	100101	0	25
011011	-1	7	100100	0	26
011100	-1	6	100011	0	27
011101	-1	5	100010	0	28
011110	-1	4	100001	0	29
011111	-1	3	100000	0	30

Note - It is possible for the decoder to receive the codewords 000000, 000001, 000010 and 000011 due to transmission errors.

TABLE 19/G.722

Conversion from S-bit codewords to quantizer intervals

RIL	SIL	IL5		RIL	SIL	IL5
000000	-1	1		11111	-1	1
000001	-1	1		11110	0	1
000010	-1	15		11101	0	2
000011	-1	14		11100	0	3
000100	-1	13		11011	0	4
000101	-1	12		11010	0	5
000110	-1	11		11001	0	6
000111	-1	10		11000	0	7
001000	-1	9		10111	0	8
001001	-1	8		10110	0	9
001010	-1	7		10101	0	10
001011	-1	6		10100	0	11
001100	-1	5		10011	0	12
001101	-1	4		10010	0	13
001110	-1	3		10001	0	14
001111	-1	2		10000	0	15

Note - It is possible for the decoder to receive the codewords 00000 and 00001 due to transmission errors.

TABLE 20/G.722

Conversion from quantizer intervals to 2-bit output codewords

SIH	MIH	IH
-1	2	00
-1	2	01
0	1	11
0	2	10

TABLE 21/G.722

Conversion from 2-bit codewords to quantizer intervals

IH	SIH	IH2
00	-1	2
01	-1	1
11	0	1
10	0	2

6.2.1 *Description of the lower sub-band ADPCM*

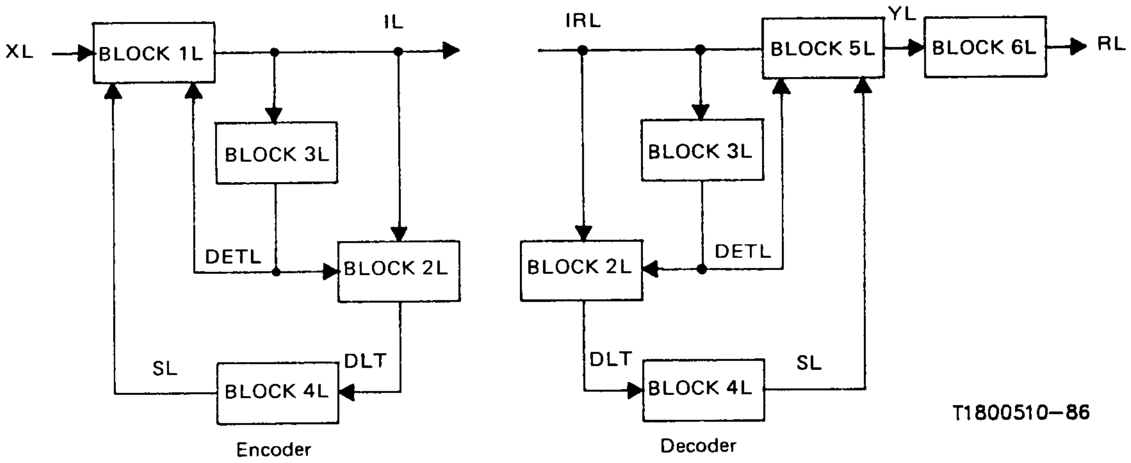


FIGURE 19/G.722

Lower sub-band ADPCM encoder and decoder

6.2.1.1 *Difference signal computation and quantization in the lower sub-band (BLOCK IL)*

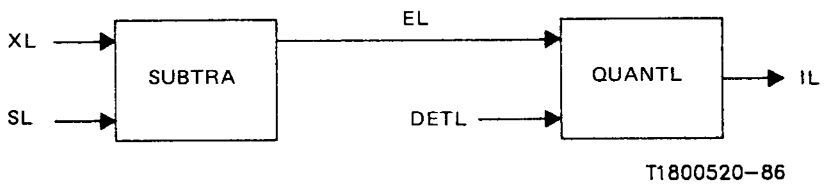


FIGURE 20/G.722

Difference signal computation and quantization in the lower sub-band

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SUBTRA

Inputs: XL, SL

Output: EL

Function: Compute the difference signal by subtracting from the input signal its predicted value.

EL = XL - SL

QUANTL

Inputs: EL, DETL

Output: IL

Note 1 - If WD falls exactly on a higher decision level, LDU, the larger adjacent MIL is used.

Note 2 - When both the lower and higher decision levels, LDL and LDU, are the same value, the value of MIL is excluded from that to be chosen.

Function: Quantize the difference signal in the lower sub-band.

$$SIL = EL >> 15 \qquad | \quad \text{Sign of EL}$$

$$WD = \begin{cases} EL, & \text{if } SIL == 0 \quad | \quad \text{Magnitude of EL} \\ 32767 - EL \,\&\, 32767, & \text{if } SIL == -1 \quad | \quad \text{(Magnitude of EL)} - 1 \end{cases}$$

Quantizer decision levels and corresponding MIL values:

WD		MIL
Lower decision level (LDL)	Higher decision level (LDU)	
0	(Q6 (1) << 3) * DETL	1
(Q6 (1) << 3) * DETL	(Q6 (2) << 3) * DETL	2
(Q6 (2) << 3) * DETL	(Q6 (3) << 3) * DETL	3
(Q6 (3) << 3) * DETL	(Q6 (4) << 3) * DETL	4
(Q6 (4) << 3) * DETL	(Q6 (5) << 3) * DETL	5
:	:	:
:	:	:
(Q6 (26) << 3) * DETL	(Q6 (27) << 3) * DETL	27
(Q6 (27) << 3) * DETL	(Q6 (28) << 3) * DETL	28
(Q6 (28) << 3) * DETL	(Q6 (29) << 3) * DETL	29
otherwise		30

Q6 is obtained from Table 14/G.722.

IL is obtained from Table 16/G.722 using SIL and MIL.

6.2.1.2 *Inverse quantization of the difference signal in the lower sub-band (BLOCK 2L)*

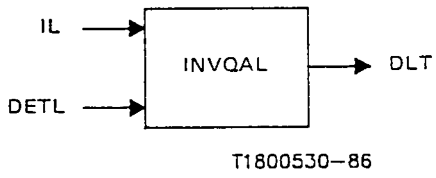


FIGURE 21/G.722

Inverse quantization of the difference signal in the lower sub-band

---

INVQAL

Inputs: IL (ILR in the decoder), DETL

Output DLT

Function: Compute the quantized difference signal for the adaptive predictor in the lower sub-band.

RIL = IL >>> 2	|	Delete 2 LSB
SIL and IL4 are obtained from Table 17/G.722 using RIL. Use IL4 as an address for QQ4 in Table 14/G.722	|	Derive sign of DLT

WD1 = QQ4(IL4) << 3

$$WD2 = \begin{cases} WD1 & \text{if } SIL == 0 \\ WD1 & \text{if } SIL == -1 \end{cases}$$

| Scale table constant Attach sign

DLT = DETL $*$ WD2

6.2.1.3 *Quantizer scale factor adaptation in the lower sub-band (BLOCK 3L)*

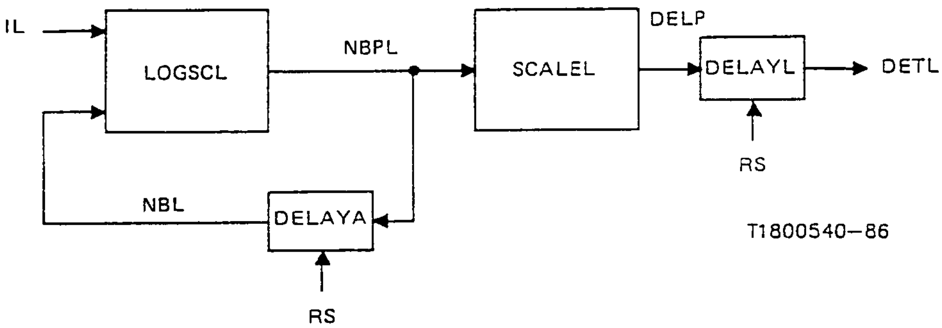


FIGURE 22/G.722

Quantizer scale factor adaptation in the lower sub-band

LOGSCL

Inputs: IL (ILR in the decoder), NBL

Output NBPL

Function: Update the logarithmic quantizer scale factor in the lower sub-band.

RIL = IL >>> 2	Delete 2 LSBs
IL4 are obtained from Table 17/G.722 using RIL. Use IL4 as an address for WL in Table 14/G.722	
WD = NBL ∗ 32512	Leakage factor of 127/128.
NBPL = WD + WL(IL4)	Add scale factor multiplier
$$NBPL = \begin{cases} 0, & \text{if } NBPL < 0 \\ 18432, & \text{if } NBPL > 18432 \end{cases}$$	Lower limit of 0, Upper limit of 9,

---

DELAYA

Inputs: x, RS

Output y

Function : Memory block. For any input x, the output is given by:

$$y(n) = \begin{cases} x(n-1), & \text{if } RS == 0 \\ 0, & \text{if } RS == 1 \end{cases}$$

Reset to 0.

SCALEL

Inputs: NBPL

Output: DEPL

Note - Either Method 1 or Method 2 is used.

Function: Compute the quantizer scale factor in the lower sub-band.

Method 1 (using 353-entry table)

WD1 = (NBPL >> 6) & 511	Compute table address for ILA
WD2 = WD1 + 64	
Use WD2 as an address for ILA in Table 15/G.722	
DEPL = (ILA(WD2) + 1) << 2	Scaling by 2-bit shift

Method 2 (using 32-entry table)

WD1 = (NBPL >> 6) & 31	Fractional part of NBPL.
WD2 = NBPL >> 11	Integer part of NBPL.
Use WD1 as an address for ILB in Table 15/G.722.	
WD3 = ILB(WD1 >> (8 - WD2)	Scaling with integer part
DEPL WD3 << 2	Scaling by 2-bit shift

DELAYL

Inputs: x, RS

Output y

Function: Memory block. For the input x, the output is given by:

$$y(n) = \begin{cases} x(n-1), & \text{if RS == 0} \\ 32, & \text{if RS == 1} \end{cases} \quad \text{Reset to minimum value}$$

6.2.1.4 *Adaptive predictor and reconstructed signal calculator in the lower sub-band (BLOCK 4L)*

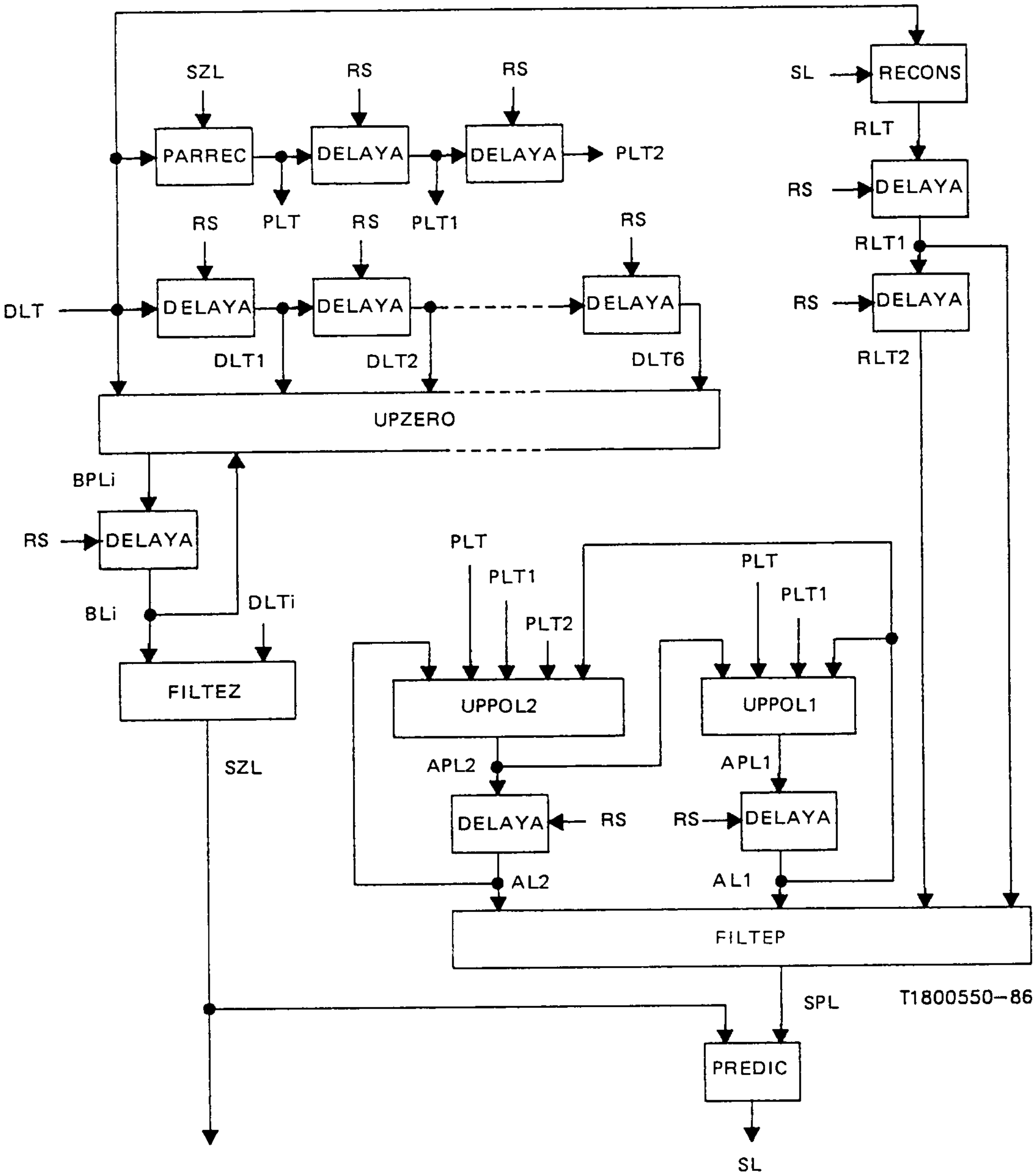


FIGURE 23/G.722

Adaptive predictor and reconstructed signal calculator in the lower sub-band

---

DELAYA

See § 6.2.1.3 for specification.

PARREC

Inputs: DLT, SZL

Output: PLT

Function: Compute partially reconstructed signal.

PLT = DLT + SZL

RECONS

Inputs: SL, DLT

Output: RLT

Function: Compute reconstructed signal for the adaptive predictor.

RLT = SL + DLT

UPZERO

Inputs: DLT, DLTi (i = 1 to 6), BLi (i = 1 to 6)

Output: BPLi (i = 1 to 6)

Function: Update sixth-order predictor (zero section) coefficients.

```
       ⎧    0,                    if DLT = = 0         |    Gain of zero
WD1 =  ⎨                                               |
       ⎩   128,                   if DLT ! = 0         |    Gain of 1/128
                                                       |
SG0 = DLT >> 15                                        |    Sign of DLT
```

Repeat the following computations for i = 1 to 6:

```
SGi = DLTi >> 15

       ⎧   WD1,                   if SG0 = = SGi       |    Sign of DLTi
WD2 =  ⎨                                               |
       ⎩   -WD1,                  if SG0 ! =   SGi     |    Attach sign to WD1
                                                       |
WD3 = BLi * 32640                                      |    Leak factor of 255/256
BPLi = WD2 + WD3                                       |    Update zero-section
                                                       |    coefficients
```

UPPOL2

Inputs: ALi (i = 1 and 2), PLT, PLTi (i = 1 and 2)

Output: APL2

Function: Update second predictor coefficient (pole section).

SG0 = PLT >> 15 | Sign of PLT
SG1 = PLT1 >> 15 | Sign of PLT1
SG2 = PLT2 >> 15 | Sign of PLT2

WD1 = AL1 + AL1 | Compute f (AL1)
WD1 = WD1 + WD1 | [Eq. (3-34) of § 3.6.3]

$$\text{WD2} = \begin{cases} 0 - \text{WD1} & \text{if SG0} == \text{SG1} \\ \text{WD1}, & \text{if SG0} \mathrel{!=} \text{SG1} \end{cases}$$

| Attach correct sign to f(AL1)

WD2 = WD2 >> 7 | Gain of 1/128

$$\text{WD3} = \begin{cases} 128, & \text{if SG0} == \text{SG2} \\ -128, & \text{if SG0} \mathrel{!=} \text{SG2} \end{cases}$$

| Attach sign to the constant of 1/128

WD4 = WD2 + WD3 | Compute gain factor
WD5 = AL2 ∗ 32512 | Leak factor of 127/128
APL2 = WD4 + WD5 | Update second pole section coefficient

$$\text{APL2} = \begin{cases} 12288, & \text{if APL2} > 12288 \\ -12288, & \text{if APL2} < -12288 \end{cases}$$

| Upper limit of +0.75
| Lower limit of -0.75

UPPOL1

Inputs: AL1, APL2, PLT, PLT1

Output: APL1

Function: Update first predictor coefficient (pole section).

SG0 = PLT >> 15 | Sign of PLT
SG1 = PLT1 >> 15 | Sign of PLT1

$$\text{WD1} = \begin{cases} 192, & \text{if SG0} == \text{SG1} \\ -192, & \text{if SG0} \mathrel{!=} \text{SG1} \end{cases}$$

| Gain 3/256

WD2 = AL1 ∗ 32640 | Leak factor of 255/256
APL1 = WD1 + WD2 | Update first pole section coefficient

WD3 = 15360 - APL2 | Compute $(1 - 2^{-4} - \text{APL2})$

$$\text{APL1} = \begin{cases} \text{WD3}, & \text{if APL1} > \text{WD3} \\ -\text{WD3}, & \text{if APL1} < -\text{WD3} \end{cases}$$

| Upper limit of APL1
| Lower limit of APL1

FILTEZ

Inputs: DLTi (i = 1 to 6), BLi (i = 1 to 6)

Output: SZL

Function: Compute predictor output signal (zero section).

WD1 = DLT1 + DLT1	Compute partial zero section output
WD1 = BL1 ∗ WD1	
WD2 = DLT2 + DLT2	
WD2 = BL2 ∗ WD2	
: : :	
: : :	
: : :	
WD6 = DLT6 + DLT6	
WD6 = BL6 ∗ WD6	
SZL = ((((WD6 + WD5) + WD4) + WD3) + WD2 + WD1	Sum the partial zero section outputs

FILTEP

Inputs: RLTi (i = 1 and 2), ALi (i = 1 and 2)

Output: SPL

Function: Compute predictor output signal (pole section).

WD1 = RLT1 + RLT1	Compute partial pole section output
WD1 = AL1 ∗ WD1	
WD2 = RLT2 + RLT2	Sum the partial pole Section outputs
WD2 = AL2 ∗ WD2	
SPL = WD1 + WD2	

PREDIC

Inputs: SPL, SZL

Output: SL

Function: Compute the predictor output value.

SL = SPL + SZL

6.2.1.5 *Reconstructed signal calculator for the decoder output in the lower sub-band (BLOCK 5L)*

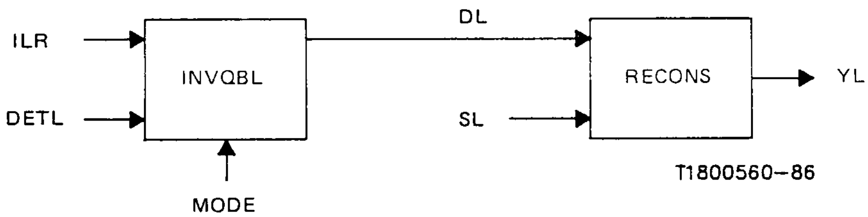


FIGURE 24/G.722

Reconstructed signal calculator for the decoder output in the lower sub-band

INVQBL

Inputs: ILR, DETL, MODE

Output: DL

Note - DL may be substituted by output signal (DLT) of sub-block INVQAL in the case of Mode 3.

Function: Compute quantized difference signal for the decoder output in the lower sub-band.

RIL = ILR SIL and IL6 are obtained from Table18//G.722 using RIL Use IL6 as an address for QQ6 in Table 14/G.722. WD1 QQ6(IL6) << 3	- if MODE = = 1	6-bit codeword Scale table constant
RIL IRL >>> 1 SIL and IL5 are obtained from Table 19/G.722 using RIL. Use IL5 as an address for QQ5 in Table 14/G.722 WD1 = QQ5(IL5) << 3	- if MODE = = 2	5-bit codeword Scale table constant
RIL = IRL >>> 2 SIL and IL4 are obtained from Table 17/G.722 using RIL. Use IL4 as an address for QQ4 in Table 14/G.722 WD1 = QQ4(IL4) << 3	- if MODE = = 3	4-bit codeword Scale table constant

$$WD2 = \begin{cases} WD1, & \text{if } SIL == 0 \\ -WD1, & \text{if } SIL == -1 \end{cases}$$

Attach sign

DL = DETL $*$ WD2

RECONS

See § 6.2.1.4 for specification. Substitute DL for DLT as input, YL for RLT as output.

6.2.1.6 *Reconstructed signal saturation in the lower sub-band (BLOCK 6L)*

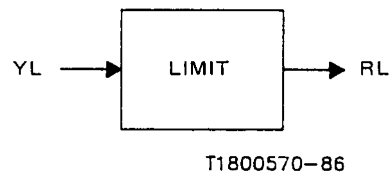


FIGURE 25/G.722

Reconstructed signal saturation in the lower sub-band

LIMIT

Inputs: YL

Output: RL

Function: Limit the output reconstructed signal.

RL = YL

$$RL = \begin{cases} 16383, & \text{if } YL > 16383 \quad \text{Upper limit} \\ -16384, & \text{if } YL < -16384 \quad \text{Lower limit} \end{cases}$$

6.2.2 *Description of the higher sub-band ADPCM*

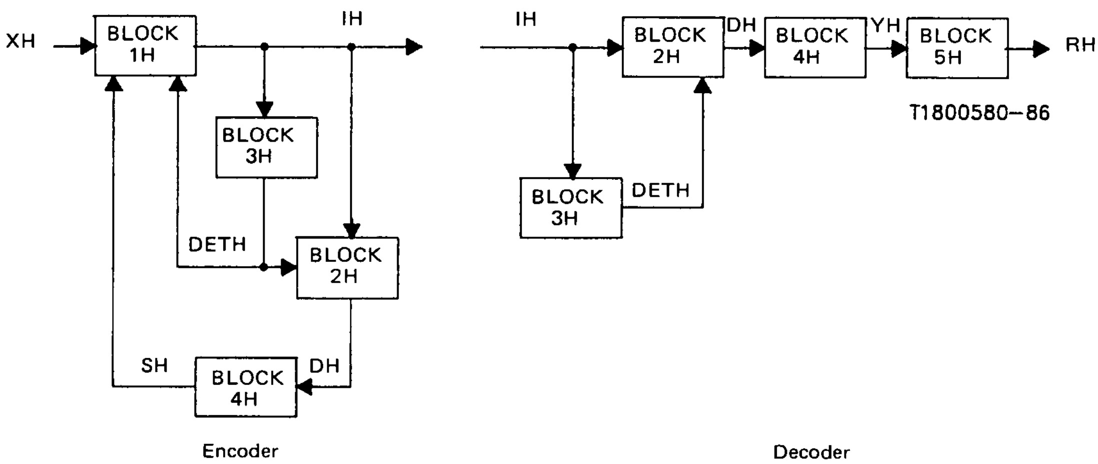


FIGURE 26/G.722

Higher sub-band ADPCM encoder and decoder

6.2.2.1 *Difference signal computation and quantization in the higher sub-band (BLOCK 1H)*

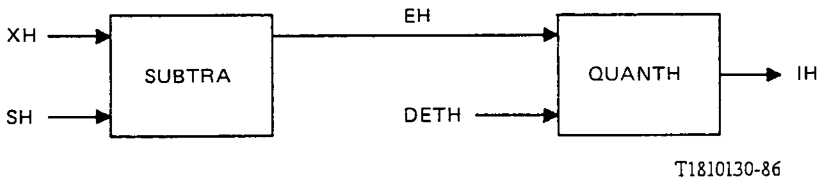


FIGURE 27/G.722

Difference signal computation and quantization in the higher sub-band

---

SUBTRA

See § 6.2.1.1 for specification. Substitute XH for XL and SH for SL as inputs, and EH for EL as output.

QUANTH

Inputs: EH, DETH

Output: IH

Note - If WD falls exactly on a higher decision level, HDU, the larger adjacent MIH is used.

Function: Quantize the difference signal in the higher sub-band.

SIH = EH >> 15		|	Sign of EH
WD = { EH,	if SIH = = 0	|	Magnitude of EH
32767, - EH&32767	if SIH = = - 1	|	(Magnitude of EH) - 1

Quantizer decision levels and corresponding MIH values:

WD		MIH
Lower decision level (HDL)	Higher decision level (HDU)	
0	(Q6 (1) << 3) ∗ DETH	1
otherwise		2

Q2 is obtained from Table 14/G.722.

IH is obtained from Table 20/G.722 using SIH and MIH.

6.2.2.2 *Inverse quantization of the difference signal in the higher sub-band (BLOCK 2H)*

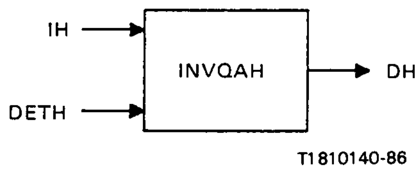


FIGURE 28/G.722

Inverse quantization of the difference signal in the higher sub-band

---

INVQAH

Inputs: IH, DETH

Output: DH

Function: Compute the quantized difference signal in the higher sub-band.

SIH and IH2 are obtained from Table 21/G.722 using IH. Use 1H2 as an address for QQ2 in Table 14/G.722	Derive sign of DH
WD1 = QQ2(IH2) << 3	Scale table constant
$WD2 = \begin{cases} WD1, & \text{if } SIH == 0 \\ -WD1, & \text{if } SIH == -1 \end{cases}$	Attach sign

DH = DETH ∗ WD2

6.2.2.3 *Quantizer scale factor adaptation in the higher sub-band (BLOCK 3H)*

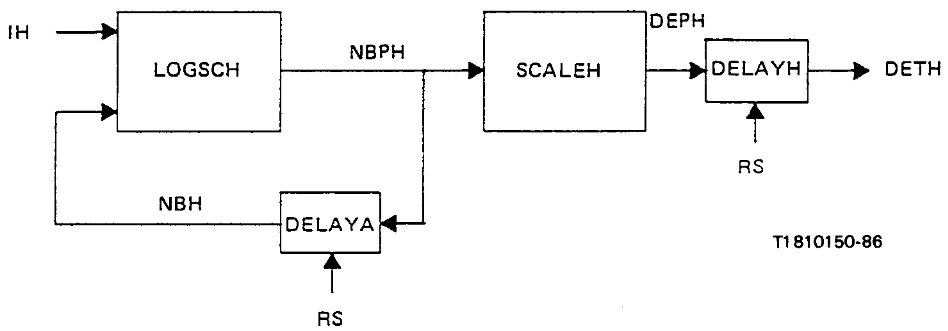


FIGURE 29/G.722

Quantizer scale factor adaptation in the higher sub-band

LOGSCH

Inputs: IH, NBH

Output: NBPH

Function: Update the logarithmic quantizer scale factor in the higher sub-band.

IH2 is obtained from
Table 21/G.722 in using IH.
Use IH2 as an address for WH
in Table 14/G.722.

```
WD = NBH * 32512                          |  Leakage factor of 127/128
NBPH = WD + WH (IH2)                      |  Add scale
                                          |  factor
                                          |  multiplier
```

$$NBPH = \begin{cases} 0, & \text{if } NBPH < 0 \\ 22528, & \text{if } NBPH > 22528 \end{cases}$$

```
| Lower limit of 0
|
| Upper limit of 11
```

DELAYA

See § 6.2.1.3 for specification.

SCALEH

Input: NBPH

Output: DEPH

Note - Either Method 1 or Method 2 is used.

Function: Compute the quantizer scale factor in the higher sub-band.

Method 1 (using 353-entry table)

```
WD = (NBPH >> 6) & 511                    |  Compute table address for
                                          |  ILA
```

Use WD as an address for ILA
in Table 15/G.722.

```
DEPH = (ILA(WD) + 1) << 2                 |  Scaling by
                                          |  2-bit shift
```

Method 2 (using 32-entry table)

```
WD1 = (NBPH >> 6) & 31                    |  Fractional part of NBPH
WD2 = NBPH >> 11                          |  Integer part of NBPH
```

Use WD1 as an address for ILB
in Table 15/G.722.

```
WD3 = ILB (WD1) >> (10 - WD2)             |  Scaling with
                                          |  integer part
DEPH = WD3 << 2                           |  Scaling by 2-bit shift
```

DELAYH

Inputs: x, RS

Output: y

Function: Memory block. For the input x, the output is given by:

$$y(n) = \begin{cases} x(n-1), & \text{if } RS == 0 \\ 8, & \text{if } RS == 1 \end{cases} \quad \Big| \quad \text{Reset to minimum value}$$

6.2.2.4 *Adaptive predictor and reconstructed signal calculator in the higher sub-band (BLOCK 4H)*

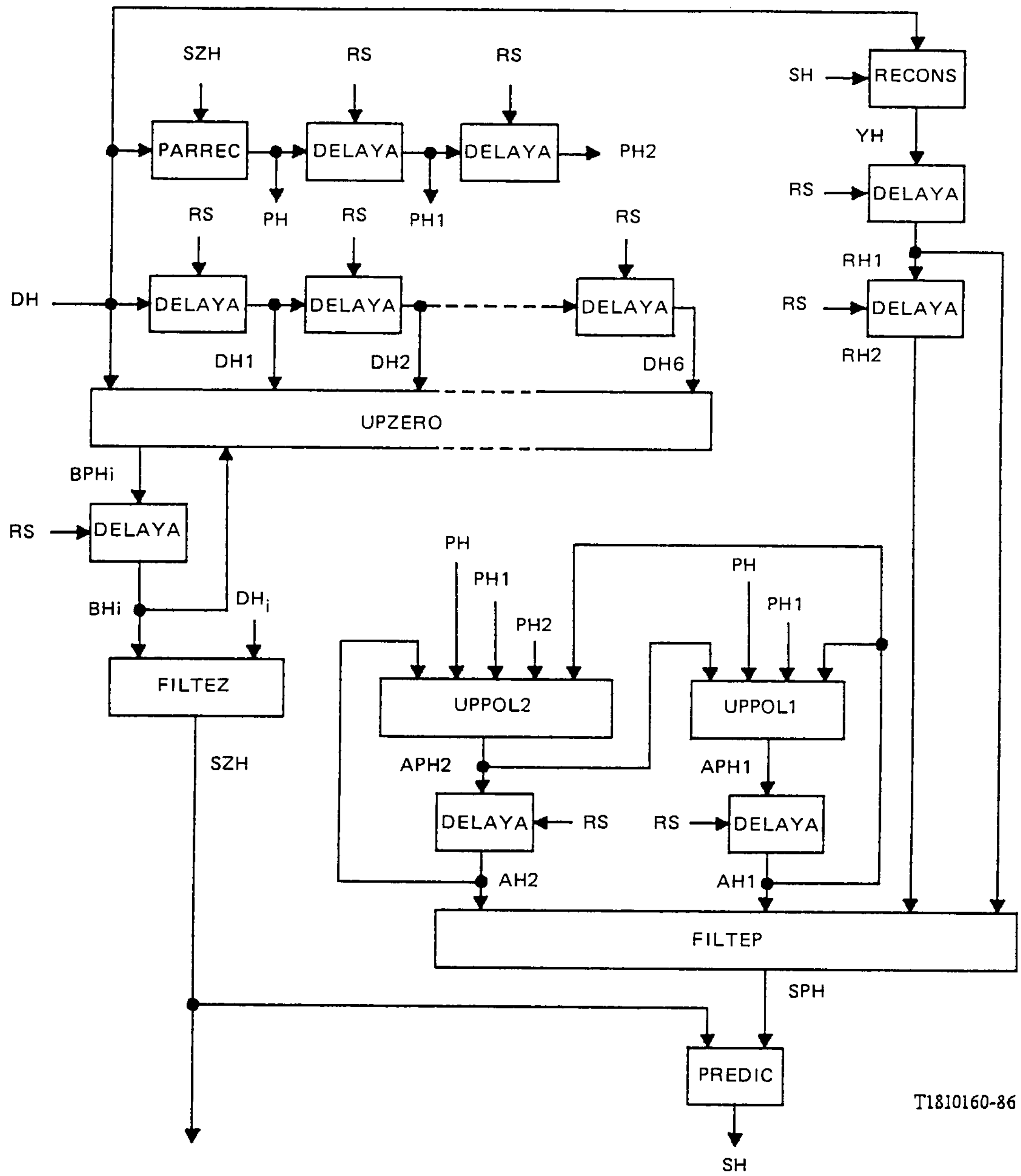


FIGURE 30/G.722

Adaptive predictor and reconstructed signal calculator in the higher sub-band

---

DELAYA

See § 6.2.1.3 for specification.

PARREC

See § 6.2.1.4 for specification. Substitute DH for DLT and SZH for SZL as inputs, and PH for PLT as output.

RECONS

See § 6.2.1.4 for specification. Substitute SH for SL and DH for DLT as inputs, and YH for RLT as output.

UPZERO

See § 6.2.1.4 for specification. Substitute DH for DLT, DHi for DLTi, and BHi for BLi as inputs, and BPHi for BPLi as outputs.

UPPOL2

See § 6.2.1.4 for specification. Substitute AHi for ALi, PH for PLT and PHi for PLTi as inputs, and APH2 for APL2 as output.

UPPOL1

See § 6.2.1.4 for specification. Substitute AH1 for AL1, APH2 for APL2, PH for PLT and PH1 for PLT1 as inputs, and APH1 for APL1 as output.

FILTEZ

See § 6.2.1.4 for specification. Substitute DHi for DLTi and BHi for BLi as inputs, and SZH for SZL as output.

FILTEP

See § 6.2.1.4 for specification. Substitute RHi for RLTi and AHi for ALi as inputs, and SPH for SPL as output.

PREDIC

See § 6.2.1.4 for specification. Substitute SPH for SPL and SZH for SZL as inputs, and SH for SL as output.

6.2.2.5 *Reconstructed signal saturation in the higher sub-band (BLOCK 5H)*

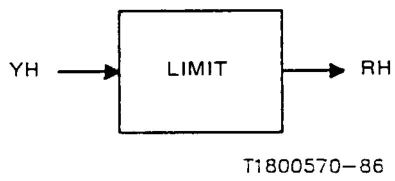


FIGURE 31/G.722

Reconstructed signal saturation in the higher sub-band

LIMIT

See § 6.2.1.6 for specification. Substitute YH for YL as input, and RH for RL as output.

APPENDIX I

(to Recommendation G.722)

Networking aspects

The purpose of this Appendix is to give a broad outline of the interaction of 64 kbit/s (7 kHz) audio coding with other parts of the digital network. Some general guidance is also offered.

The establishment of the connection is beyond the scope of this Recommendation.

I.1 *Network characteristics*

This Recommendation is applicable to systems operating in networks which exhibit each of the following characteristics:

i) availability of network octet timing at the terminals;

Note - Octet timing may also be derived from control signals within the frame structure defined in Recommendation G.725;

ii) plesiochronous networking where the reference clocks meet the timing requirements given in Recommendation G.811, or synchronous networking;

iii) 64 kbit/s connection types having either of the following characteristics:

- full 64 kbit/s transparency,
- pulse density restriction as described in Recommendation G.802.

Note - 64 kbit/s (7 kHz) audio coding can also operate in networks where there is substitution of a signalling bit for the 8th bit of the octet as described in Recommendation G.704, § 3.1 or where there is 56 kbit/s transparency only. However, a reduction of the audio bit rate and auxiliary data channel capacity occurs and only two modes of operation, denoted 1 *bis* (unframed) and 3 *bis,* are possible as follows:

- Mode 1 *bis* : 56 kbit/s for audio coding and no data channel;
- Mode 3 *bis*: 48 kbit/s for audio coding, a 6.4 kbit/s data channel and 1.6 kbit/s for service channel framing and mode control.

I.2 *Integration into the telecommunications network*

It is foreseen that the 64 kbit/s (7 kHz) audio coding system will be used for point-to-point, multipoint and broadcast applications. Examples of particular uses are: commentary quality channels for broadcasting purposes and high quality speech for audio and video conferencing applications.

The coding system can operate over any 64 kbit/s bearer channel (see § I.1), e.g. the public switched telephone network, leased circuits or over an ISDN.

Processes such as digital speech interpolation, echo control and digital pads must be disabled for the transmission of 64 kbit/s (7 kHz) audio coding. The disabling protocol is not the subject of this Recommendation.

It should be noted however that signal processing may occur in a multipoint conference unit (see § I.7).

I.3 *Audio performance of the 64 kbit/s (7 kHz) audio coding system*

I.3.1 *Speech*

The speech performance of the 64 kbit/s (7 kHz) audo coding system has been quantified in terms of Q_w-values, where Q_w is a measure of the signal-to-correlated noise ratio of the wideband system, measured in dB. Detailed information on *Q*-value measurements may be found in Recommendation P.81. This Recommendation, although primarily intended for telephony bandwidth applications, has been used for the evaluation of wideband systems - signified by the subscript *W* - by use of an appropriate filter (50-7000 Hz).

For guidance purposes only, a Q_w value of 38 dB corresponds approximately to a 128 kbit/s (7 kHz) PCM system (sampling frequency 16 kHz, coding law as in Recommendation G.711), whereas a Q_w value of 45 dB is approximately equivalent to the audio parts of the coder interconnected without the intermediate SB-ADPCM coding process.

Table I-1/G.722 indicates the relative performance in Q_w values for nominal input values.

TABLE I-1/G.722

Relative levels of speech performance (Q_w values)

Mode of operation	Transcodings		
	1	4 Analogue according to Fig. I-1/G.722	4 Digital according to Fig. I-2/G.722
1 (64 kbit/s)	45	38	41
1 (56 kbit/s)	43	36	38
1 (48 kbit/s)	38	29	34

The performance of the 64 kbit/s (7 kHz) audio coding system has been found to be substantially unaffected by randomly distributed errors at BER levels as high as $1 \cdot 10^{-4}$. High error ratios approaching $1 \cdot 10^{-3}$ produce perceptible degradation which may be considered tolerable in certain applications.

No particular problems have been experienced in the multiple talker condition and hence correct operation under normal conference conditions can safely be assumed.

The performance under conditions of mode mismatch (i.e. where the variant used in the decoder for a given octet does not correspond to the mode of operation) is considered in § I.5.

I.3.2 *Music*

Although primarily designed for speech, no significant distortions may be expected when coding a wide range of music material in Mode 1. Further study on the effects on music signals is a matter of Study Group CMTT.

I.4 *Audio performance when interconnected with other coding systems on an analogue basis*

I.4.1 *64 kbit/s PCM*

Informal subjective tests carried out over a path consisting of an analogue interconnected combination of a 64 kbit/s PCM link conforming to Recommendation G.711 and a 64 kbit/s (7 kHz) audio coding link has indicated that no interworking problems will occur. However, the performance of the combination will not be better than that of 64 kbit/s PCM.

Interconnection of the two coding systems on a digital basis is the subject of § I.8.

I.4.2 *32 kbit/s ADPCM*

An analogue interconnected combination of a 32 kbit/s ADPCM link conforming to Recommendation G.721 and a 64 kbit/s (7 kHz) audio coding link is not expected to pose any interworking problems. However, the performance of the combination will not be better than that of 32 kbit/s ADPCM.

Interconnection of the two coding systems at a digital level is the subject of further study.

I.5 *Audio performance under mode switching*

It is recommended that mode switching should be performed synchronously between the transmitter and the receiver to maximize the audio performance. However, asynchronous mode switching may be considered since the condition of mode mismatch will probably be of limited duration and hence the corresponding performance is likely to be acceptable. Although not desirable, operation under permanent mode mismatch may be contemplated in exceptional circumstances. Table I-2/G.722 indicates the relative performance under all mode mismatch combinations for nominal input levels.

TABLE I-2/G.722

Relative speech performance under mode mismatch (Q_W values)

Bit rate used for audio reception	Bit rate used for audio transmission	
	56 kbit/s	48 kbit/s
64 kbit/s	41	35
56 kbit/s	-	36

Note - The bits not used for audio coding have been replaced by bits of a pseudorandom sequence.

I.6 *Auxiliary data channel performance*

The available combinations of audio and data channel bit rates depends on the connection types described in § I.1 iii).

The data channel is unaffected by the characteristics of the audio signal since the audio and data channels are effectively decoupled. The transparency of the data channel is limited only by the choice of signalling sequences which could be used to derive the terminal identification. If these sequences are chosen to be of a suitable format, the possibility of their simulation by audio or data bits can be made extremely low. Hence, for all practical purposes, the data channel may be assumed to be transparent.

The control of the data channel capacity is considered in Recommendation G.725.

Although the format of the data channel is not part of this Recommendation, it may be noted that the use of two completely independent 8 kbit/s data channels when the total data channel capacity is 16 kbit/s is not prohibited by the algorithm.

Under transmission error conditions the data channel is not subject to error multiplication due to the audio coding algorithm.

Note - It might be possible to obtain additional data channel capacity by substituting data for the two bits normally allocated to the higher sub-band with the consequent penalty of a reduction in the audio bandwidth. However, such an approach is likely to require a more stringent specification for the receive filter characteristics in order to minimize aliasing effects.

I.7 *Multi-point conference configuration*

The specific features of a multipoint conference arrangement including control of the data channel, echo control, and handling of control messages between terminals, are beyond the scope of this Recommendation. However, the audio coding algorithm has been chosen to maintain maximum flexibility for multipoint conference arrangements which are likely to emerge. There are a number of general guidelines which should be noted:

- To maximize audio performance, the highest audio bit rate possible, consistent with the transmitted data channel bit rate requirement, should be used for transmission into and out of the signal summing facility of the multipoint conference unit.

  Note - The signal summation must be carried out on a linear representation of the signals.

- The transmit and receive modes of a terminal or port of a multipoint conference unit do not necessarily have to be the same.

- Signal summing at the sub-band uniform PCM level is preferred for the following reasons:

  i) the hardware is minimized in the multipoint conference unit (MCU) by eliminating the need for quadrature mirror filters,

  ii) signal quality is maximized and additional signal delay is eliminated by avoiding additional filtering,

  iii) echo control is likely to be simpler to perform at the sub-band level.

Figure I-3/G.722 indicates a possible arrangement at the multipoint conference bridge with signal summing at the sub-band level;

- For reasons of audio performance, the number of tandem connected multipoint conference units interconnected with 64 kbit/s (7 kHz) audio coding is limited to three, see Figure I-4/G.722).
- In the case where the multipoint conference unit includes 64 kbit/s PCM ports, digital transcoding principles equivalent to that described in § I.8 should be used to derive the higher and lower sub-band signals.

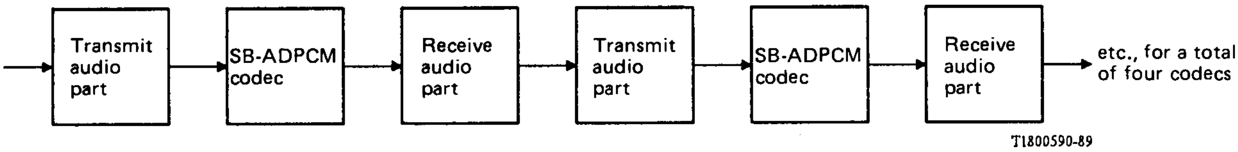


FIGURE I-1/G.722

Four transcodings (analogue interconnection)

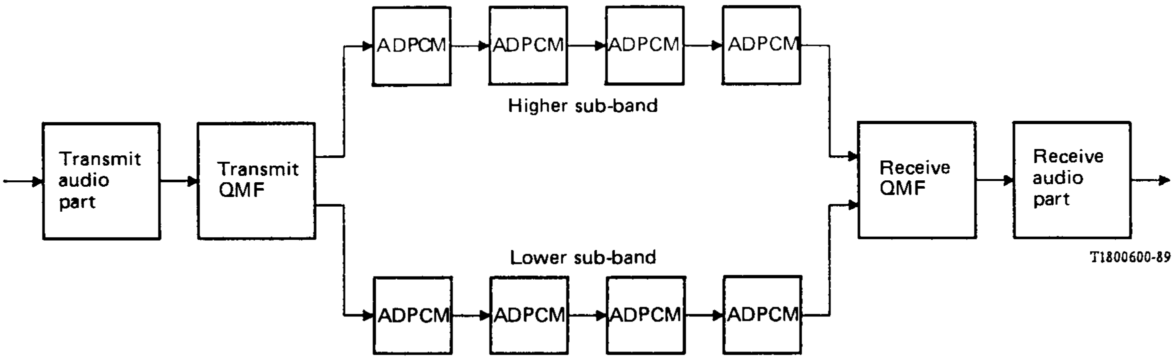


FIGURE I-2/G.722

Four transcodings (digital interconnection)

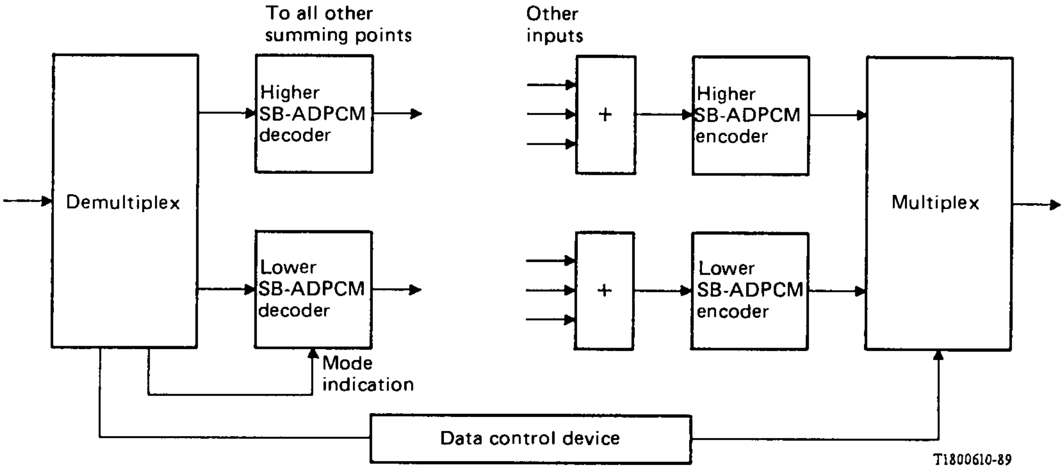


FIGURE I-3/G.722

Possible arrangement at a multipoint conference unit

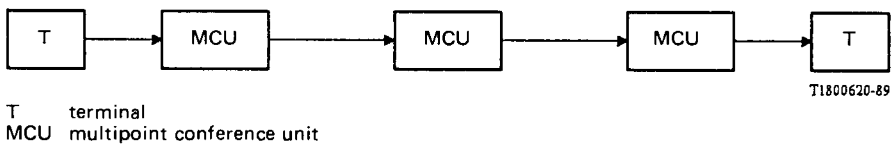


FIGURE I-4/G.722

Tandem connected multipoint conference units

I.8 *Digital transcoding between the 64 kbit/s (7 kHz) audio coding system and 64 kbit/s PCM*

Figure I-5/G.722 indicates the method recommended for the digital interconnection of the 64 kbit/s (7 kHz) audio coding system and 64 kbit/s PCM to Recommendation G.711.

The principle of transcoding from 64 kbit/s PCM to 64 kbit/s (7 kHz) audio coding involves the conversion from A-law or μ-law PCM to uniform PCM and the insertion of interleaved alternate samples of zero amplitude to the 8 kHz sampled uniform PCM signal to form a 16 kHz sampled signal. This signal is then passed through a digital low pass filter sampled at 16 kHz which does not significantly modify the baseband frequency response up to 3.4 kHz and which attenuates the frequency components above 4.6 kHz. The resulting signal is then applied to the sub-band ADPCM encoder as shown in Figure I.3/G.722.

It should be noted that the use of the lower sub-band alone to carry the information in a signal emanating from a 64 kbit/s PCM link to Recommendation G.711 should be avoided.

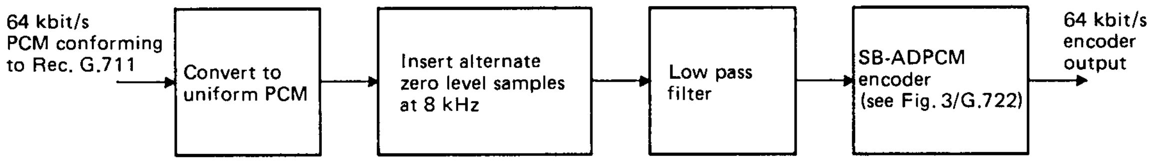
An alternative method of deriving two sub-band signals from a 64 kbit/s PCM signal using the low pass (LP) and high pass (HP) QM filter designs already employed for the 64 kbit/s (7 kHz) audio coding scheme is given in Figure I-6/G.722. The objective is to generate a higher sub-band signal which will eventually cancel the aliasing distortion introduced into the lower sub-band signal. The 64 kbit/s PCM signal is converted to uniform PCM and upsampled to 16 kHz by inserting alternate zero-valued samples. The factor 2 multiplier is inserted to preserve unity gain. The lower sub-band signal is derived by two identical stages of HP QM filtering following by 2:1 subsampling. The higher sub-band signal is derived by two filtering stages, HP followed by LP, a factor 1/2 gain reduction, sign inversion, followed by 2:1 subsampling. When these two signals are input to the QM synthesis filter of Recommendation G.722, an appropriate 7 kHz form of the original PCM is obtained.

Note that the upsampling and subsampling process should be synchronized so that instants of sample deletion correspond to the instants of zero-sample insertion.

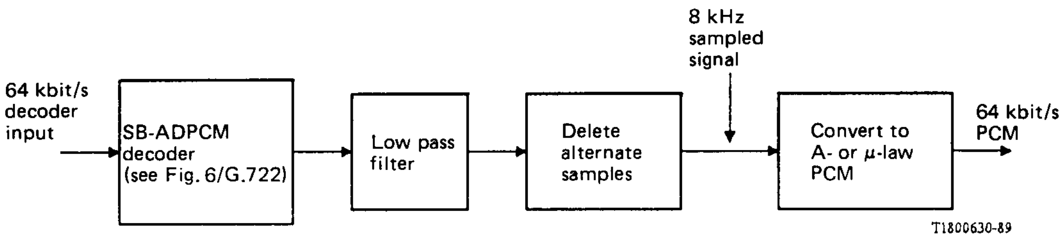
Transcoding from 64 kbit/s (7 kHz) audio coding to 64 kbit/s PCM can be achieved by taking the output signal from the sub-band ADPCM decoder and performing the following three processes in turn:

- digital low pass filtering (16 kHz sampling), which does not significantly modify the baseband frequency response up to 3.4 kHz and which attenuates the frequency components above 4.6 kHz;
- the deletion of alternate samples from the resulting 16 kHz sampled signal;
- conversion from the resulting 8 kHz sampled uniform PCM signal to A-law or μ-law PCM.

Note - The derivation of a 64 kbit/s PCM signal solely from the lower sub-band of the 64 kbit/s (7 kHz) signal is subject to further study.



a) 64 kbit/s PCM to 64 kbit/s (7 kHz) audio coding



b) 64 kbit/s (7 kHz) audio coding to 64 kbit/s PCM

FIGURE I-5/G.722

Digital transcoding between the 64 kbit/s (7 kHz) audio coding system and 64 kbit/s PCM conforming to Recommendation G.711

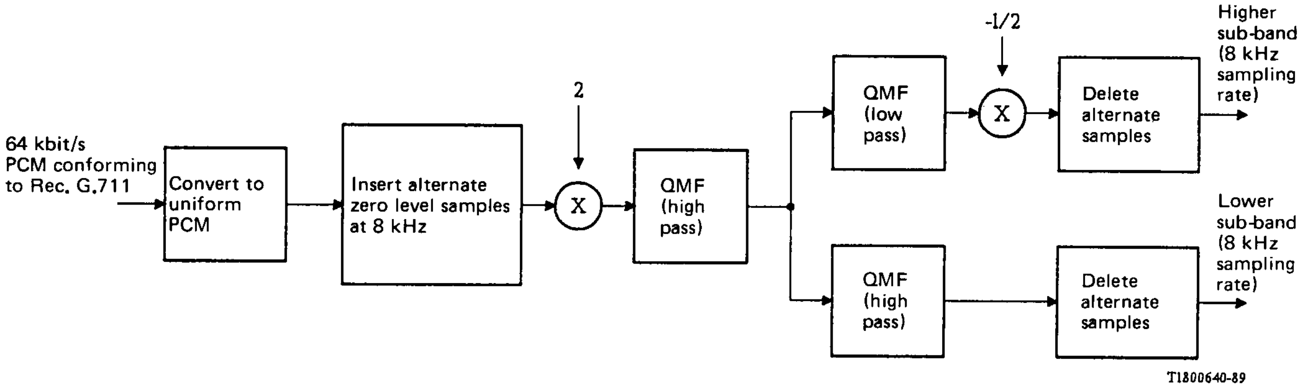


FIGURE I-6/G.722

An alternative method for digital transcoding between 64 kbit/s PCM conforming to Rec. G.711 and 64 kbit/s (7 kHz) audio coding

APPENDIX II

(to Recommendation G.722)

Digital test sequences

This Appendix gives information concerning the digital test sequences which should be used to aid verification of implementations of the ADPCM codec part of the wideband coding algorithm. Copies of the sequences are available on flexible disks (see § II.4).

II.1 *Input and output signals*

Table II-1/G.722 defines the input and output signals for the test sequences. It contains some signals (indicated by #) peculiar to these test sequences in order to facilitate the interface between the test sequence generator/receiver and the encoder/decoder. 16-bit word formats for these input and output signals are shown in Figures II-1/G.722, II-2/G.722 and II-3/G.722.

II.2 *Configurations for the application of test sequences*

Two configurations (Configuration 1 and Configuration 2) are appropriate for use with test sequences. In both configurations, a TEST signal is used to make the encoder and decoder ready to be tested with the digital test sequences. When the TEST signal is provided, the QMFs are by-passed and the test sequences are applied directly to the ADPCM encoders or decoders. An RSS signal is extracted from the input test sequences X # (I # in decoder) and results in a reset signal RS for the encoder and decoder. The RS signal will be used to initialize state variables (those indicated by * in Table 13/G.722 to zero or specific values.

II.2.1 *Configuration 1*

Configuration 1 shown in Figure II-4/G.722 is a simplified version of Figures 4/G.722 and 5/G.722. The encoder input signals, XL and XH, are described in Table 12/G.722. These input signals are directly fed to the respective lower and higher sub-band ADPCM encoders, by-passing the QMF. The encoder output signals, IL and IH, are defined in the sub-block QUANTL and QUANTH, respectively.

This sequence is used for testing the quantizer/predictor feedback loop in the encoder.

TABLE II-1/G.722

Description of input and output signals for test sequence

Name	Description
XL	15-bit uniformly quantized input signal to the lower sub-band encoder
XH	15-bit uniformly quantized input signal to the higher sub-band encoder
X #	Input test sequence with 16-bit word format as shown in Figure II-1/G.722
IL	6-bit lower sub-band ADPCM codeword
ILR	Received 6-bit lower sub-band ADPCM codeword
IH	2-bit higher sub-band ADPCM codeword
I #	Output (in Configuration 1) and Input (en Configuration 2) test sequence with 16-bit word format as shown in Figure II-2/G.722
RL	15-bit uniformly quantized output signal from the lower sub-band decoder
RH	15-bit uniformly quantized output signal from the higher sub-band decoder
RL #	Output test sequence with 16-bit word format as shown in Figure II-3/G.722
RH#	Output test sequence with 16-bit word format as shown in Figure II-3/G.722
RSS	Reset/synchronization signal
VI	Valid data indication signal

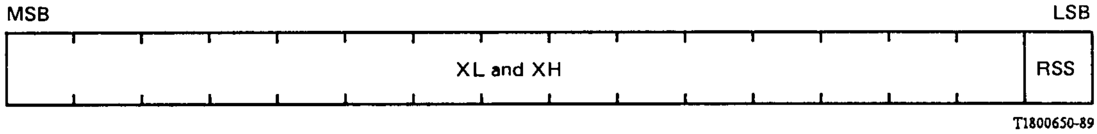
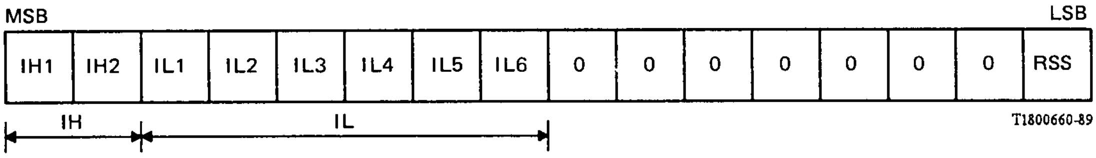


FIGURE II-1/G.722

Word format of X #



Note 1 — IH1 and IL1 are MSBs of IH and IL, respectively.

Note 2 — IL is read as ILR in Configuration 2.

FIGURE II-2/G.722

Word format of I #

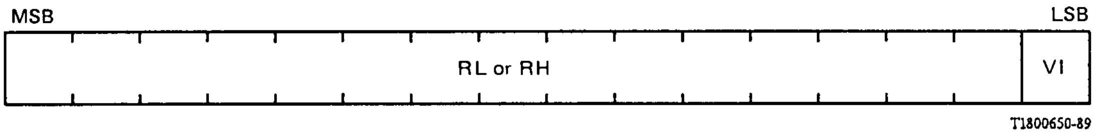


FIGURE II-3/G.722

Word format of RL# and RH#

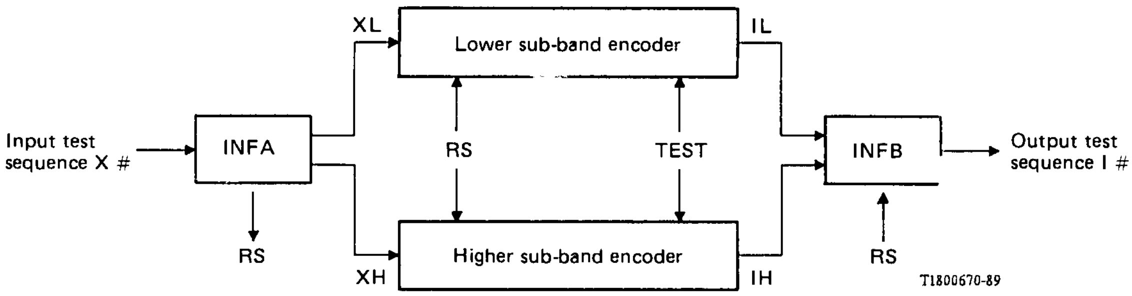


FIGURE II-4/G.722

Configuration 1 – encoder only

II.2.2 *Configuration 2*

Configuration 2 shown in Figure II-5/G.722 is a simplified version of Figures 7/G.722 and 8/G.722. The test signals, ILR and IH, and the MODE signal are described in Table 12/G.722. The corresponding decoder output signals, RL and RH, are defined in the sub-blocks LIMIT in §§ 6.2.1.6 and 6.2.2.5. For the lower sub-band, the ADPCM decoder output signals are derived for three basic modes of operation (Modes 1, 2 and 3). By-passing the QMF, the output signals, RL and RH, are separately obtained from the lower and higher sub-band ADPCM decoders, respectively.

Configuration 2 is used for testing the inverse quantizer operation and the predictor adaptation without a quantizer/predictor feedback loop in the decoder.

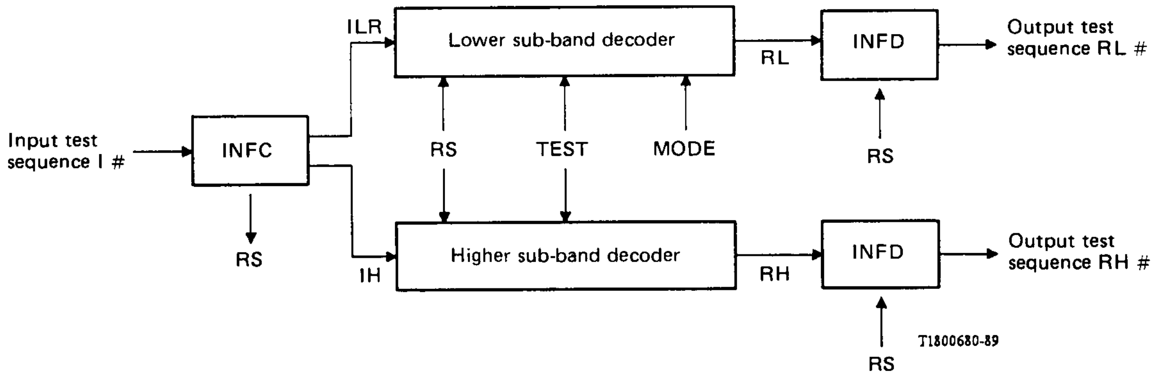


FIGURE II-5/G.722

Configuration 2 – decoder only
(RL and RL# are derived for Modes 1, 2 and 3)

II.2.3 *Reset/synchronization signal (RSS) and valid data indication (VI)*

All memory states in the two test configurations must be initialized to the exact states specified in this Recommendation prior to the start of an input test sequence in order to obtain the correct output values for the test.

In Configuration 1, the input test sequence, X#, is composed of encoder input test signals and the reset/synchronization signal (RSS) as shown in Figure II-1/G.722. The RSS signal is located at the first LSB of the input sequence. If RSS is "1", the lower and higher sub-band encoders are initialized, and the outputs of the encoders are set to "0", i.e., IH = "0" and IL = "0". This normally forbidden output code is used to indicate "non-valid data" of the outputs. After the RSS signal goes to "0", the input test sequence will be valid and the ADPCM algorithm begins to operate.

In Configuration 2, the input test sequence, I #, is composed of the first 8 bits of lower and higher sub-band decoder input codewords, and the last 8 bits consists of 7-bit zeroes and "RSS" in the LSB as shown in Figure II-2/G.722. The RSS signal has the same role as in Configuration 1. That is, if the RSS signal equals "1", the lower and higher sub-band decoders are initialized. After the RSS signal goes to "0", the ADPCM algorithm will be in the operational state. The output test sequences, RL# and RH#, are made up of a decoder output signal of 15 bits and a valid data indication signal (VI) as shown in Figure II-3/G.722. While the RSS signal to the decoder is "1", the signal "VI" is set to "1" and the decoder output set to "0", which indicates "non-valid data" of the output. When "VI" is "0", the output test sequence is valid.

In order to establish the connection between the test sequence generator/receiver and the encoder/decoder, four sub-blocks, INFA, INFB, INFC, INFD in Figures II-4/G.722 and II-5/G.722 are provided. A detailed expansion of these sub-blocks is described below using the same notations specified in § 6.2.

INFA

Input: X#

Outputs: XL, XH, RS

Function: Extract reset/synchronization signal and input signals to lower and higher sub-band ADPCM encoder.

```
RS = X# & 1        |   Extract RSS signal
XL = S#  >> 1      |   Lower sub-band input signal
XH = XL            |   Higher sub-band input signal
```

INFB

Inputs: IL, IH, RS

Outputs: I#

Function: Create an output test sequence by combining lower and higher sub-band ADPCM encoder output signals and the reset/synchronization signal.

$$I = \begin{cases} (IH <<< 6) + IL & \text{if } RS == 0 \quad | \quad \text{Combine IH and IL} \\ 0 & \text{if } RS == 1 \quad | \quad \text{Set output to zero} \end{cases}$$

```
I# = (I <<< 8) + RS    |   Add RSS signal
```

INFC

Input: I#

Outputs: ILR, IH, RS

Function: Extract reset/synchronization signal and input signals to lower and higher sub-band ADPCM decoder.

```
RS = I# & 1                |   Extract RSS signal
ILR = (I # >>> 8) & 63     |   Lower sub-band ADPCM input
IH = I # >>> 14            |   Higher sub-band ADPCM input
```

INFD

Inputs: RL (RH in higher sub-band), RS

Output: RL#(RH# in higher sub-band)

Function: Create output test sequence by combining lower (higher) sub-band ADPCM decoder output signal and the valid data indication signal.

$$RLX = \begin{cases} RL << 1 & \text{if } RS == 0 \quad | \quad \text{Scaling by 1-bit shift} \\ & \quad | \\ 0 & \text{if } RS == 1 \quad | \quad \text{Set output to zero} \end{cases}$$

```
RL# = RLX + RS    |   Add VI signal
```

II.3 *Test sequences*

II.3.1 *Input sequences for Configuration 1*

For Configuration 1, two types of input test sequences are provided:

1) sequence containing tones, d.c. and white noise,

2) sequence for testing overflow controls in the ADPCM encoders.

The first input sequence contains tones with various frequencies, DC and white noise with two levels. The signal segments and lengths are given in Table II-2/G.722.

The tones are used to move the predictor poles over their operating range and to test the stability control. Although the second pole coefficients are settled only in the vicinity of their lower limit for tone inputs, the upper limit is examined at the beginning of the d.c.-positive input. d.c. and white noise are used to vary the quantizer scale factors over their entire range.

The second input sequence permits testing of frequent overflows. The signal segments and lengths are given in Table II-3/G.722.

The sequences produces large prediction errors, so it is used to check the overflow controls in pole and zero section output computations.

In Configuration 1, the coefficient values of the zero predictor do not move to the range limits of -2 and + 2.

II.3.2 *Input sequences for Configuration 2*

For Configuration 2, these types of input test sequences are provided:

1) The sequence generated by the encoder is used when applying the input test sequence described in Table II-2/G.722;

2) The sequence generated by the encoder is used when applying the input test sequence described in Table II-3/7G.722;

3) An artificial sequence containing consecutive sub-sequences is used that would not ordinarily emanate from an encoder.

The third test sequence, consisting of 16384 values, is described below.

TABLE II-2/G.722

Sequence of tones, d.c. and white noise

Signal segments	Length (16 bits words)
3504 Hz tone	1 024
2054 Hz tone	1 024
1504 Hz tone	1 024
504 Hz tone	1 024
254 Hz tone	1 024
1254 Hz tone	1 024
2254 Hz tone	1 024
3254 Hz tone	1 024
4000 Hz tone	512
d.c, positive, low level	512
d.c., value of zero	512
d.c., negative, low level	512
White noise, low level	3 072
White noise, high level	3 072
Total length of sequence	16 384

TABLE II-3/G.722

Overflow test sequence

Signal segments	Length (16 bits words)
-16 384, +16 383; repeated	639
0, -10 000, -8192	3
-16 384, +16 383, -16 384; repeated	126
Total length of sequence	768

II.3.2.1 *Lower sub-band ADPCM codewords*

The 6-bit lower sub-band decoder input sequence consists of an MSB sequence and a distinct sequence of the remaining 5 bits. The MSB sequence consists of eight artificial sub-sequences, each 2048 bits in length, as follows:

(1) 0 0 1 0 0 1 0 0 1 0 0 1 0 0 1 0 0........
(2) 1 1 1 1 0 0 0 0 1 1 1 1 0 0 0 0 1........
(3) 1 1 1 1 1 1 1 1 1 1 1 1 1 1 1 1 1........
(4) 1 1 0 0 1 1 0 0 1 1 0 0 1 1 0 0 1........
(5) 1 0 1 0 1 0 1 0 1 0 1 0 1 0 1 0 1........
(6) 0 0 0 0 0 1 0 0 0 0 0 0 0 1 0 0 0........
(7) 0 0 1 0 1 0 0 1 0 1 0 0 1 0 1 0 0........
(8) 1 1 0 0 0 1 1 0 0 0 1 1 0 0 0 1 1........

These MSB sequences are used to force the coefficients of the zero predictor to vary across the entire range of ± 2.

The associated 5-bit word sequence consists of 64 concatenated artificial sub-sequences, each 256 values long, as described in Table II-4/G.722. This 5-bit word sequence was chosen to exercise the logarithmic quantizer scale factor over its entire range, and the log-to-linear conversion.

The composite sequence of ILR also tests the pole predictor and varies its coefficients over their allowable range. The sequences from sub-sequence numbers (56) to (64) test the conversion from the suppressed codewords, which can occur due to transmission errors, to specified quantizer intervals.

II.3.2.2 *Higher sub-band ADPCM codewords*

The 2-bit higher sub-band decoder input sequence consists of an MSB sequence and a distinct LSB sequence.

The MSB sequence consists of eight artificial sub-sequences, identical to those used in the MSB sequence for the lower sub-band ADPCM.

The LSB sequence consists of 8 concatenated artificial sub-sequences, each 2048 bits long, as follows:

(1) 1 1 1 1 1 1......
(2) alternating sixteen 1s, sixteen 0s
(3) 0 0 0 0 0 0......
(4) alternating eight 1s, eight 0s
(5) 0 0 0 0 0 0......
(6) alternating four 1s, four 0s
(7) 1 1 1 1 1 1......
(8) alternating two 1s, two 0s.

The role of the composite sequence formed by appending the l-bit LSB to the l-bit MSB is equivalent to that for the lower sub-band ADPCM codeword described in § II.3.2.1.

II.4 *Format for test sequence distribution*

II.4.1 *Disk interface and format*

Copies of the digital test sequences, on three $5^1/_4''$ flexible disk cartridge, are available from the ITU.

The operating system is the PC-DOS or MS-DOS (Version 2.0 or greater). MS-DOS $5^1/_4''$ disk format is used. The following format is used:

- 2 sides $5^1/_4''$ flexible disk
- 40 tracks per side
- 9 sectors per track
- 512 bytes per sector.

The files are written in ASCII text in order to be dumped, listed or edited easily.

TABLE II-4/G.722

Sequence of last 5 bits of ILR

Repetitive pattern, each 256 values long	
(1) 31 31 31 31 31 31	(33) 15 15 15 15 15 15 . . . .
(2) alternating sixteen 31’s, sixteen 30’s	(34) alternating sixteen 15’s, sixteen 14’s
(3) 30 30 30 30 30 30.	(35) 14 14 14 14 14 14 . . . .
(4) alternating sixteen 30’s, sixteen 29’s	(36) alternating sixteen 14’s, sixteen 13’s
(5) 29 29 29 29 29 29	(37) 13 13 13 13 13 13 . . . .
(6) alternating sixteen 29’s, sixteen 28’s	(38) alternating sixteen 13’s, sixteen 12’s
(7) 28 28 28 28 28 28	(39) 12 12 12 12 12 12 . . . .
(8) alternating sixteen 28’s, sixteen 27’s	(40) alternating sixteen 12’s, sixteen 11’s
(9) 27 27 27 27 27 27	(41) 11 11 11 11 11 11 . . . .
(10) alternating sixteen 27’s, sixteen 26’s	(42) alternating sixteen 11’s, sixteen 10’s
(11) 26 26 26 26 26 26	(43) 10 10 10 10 10 10 . . . .
(12) alternating sixteen 26’s, sixteen 25’s	(44) alternating sixteen 10’s, sixteen 9’s
(13) 25 25 25 25 25 25	(45) 9 9 9 9 9 9 . . . .
(14) alternating sixteen 25’s, sixteen 24’s	(46) alternating sixteen 9’s, sixteen 8’s
(15) 24 24 24 24 24 24	(47) 8 8 8 8 8 8 . . . .
(16) alternating sixteen 24’s, sixteen 23’s	(48) alternating sixteen 8’s, sixteen 7’s
(17) 23 23 23 23 23 23	(49) 7 7 7 7 7 7 7 . . . .
(18) alternating sixteen 23’s, sixteen 22’s	(50) alternating sixteen 7’s, sixteen 6’s
(19) 22 22 22 22 22 22	(51) 6 6 6 6 6 6 . . . .
(20) alternating sixteen 22’s, sixteen 21’s	(52) alternating sixteen 6’s, sixteen 5’s
(21) 21 21 21 21 21	(53) 5 5 5 5 5 5 . . . .
(22) alternating sixteen 21’s, sixteen 20’s	(54) alternating sixteen 5’s, sixteen 4’s
(23) 20 20 20 20 20 20	(55) 4 4 4 4 4 4 . . . .
(24) alternating sixteen 20’s, sixteen 19’s	(56) alternating sixteen 4’s, sixteen 3’s
(25) 19 19 19 19 19 19	(57) 3 3 3 3 3 3 . . . .
(26) alternating sixteen 19’s, sixteen 18’s	(58) alternating sixteen 3’s, sixteen 2’s
(27) 18 18 18 18 18 18	(59) 2 2 2 2 2 2. . . . .
(28) alternating sixteen 18’s, sixteen 17’s	(60) alternating sixteen 2’s, sixteen 1’s
(29) 17 17 17 17 17 17 17	(61) 1 1 1 1 1 1 . . . .
(30) alternating sixteen 17’s, sixteen 16’s	(62) alternating sixteen 1’s, sixteen 0’s
(31) 16 16 16 16 16 16	(63) 0 0 0 0 0 0 . . . .
(32) alternating sixteen 16’s, sixteen 15’s	(64) alternating sixteen 0’s, sixteen 3’s

II.4.2 *Type of files provided*

The test sequences are arranged into 7 files. These 17 files are classified in 3 groups according to the following description:

Class T1: Source files to be input to the ADPCM codec. Class T1 includes 2 files to be used in Configuration 1 (encoder only) and 1 file to be used in Configuration 2 (decoder only).

Class T2: Combined source-comparison files. There are 2 files in class T2. Both are used for comparison purposes at the output of the encoder in Configuration 1. Also they are used as source files to test the decoder in Configuration 2.

Class T3: Comparison files used to check the output of the decoder in different modes. There are 9 files in class T3 to test the lower sub-band decoder and 3 files in the same class to test the higher sub-band decoder. In class T3, the suffix H or L in the file name distinguishes the higher and lower sub-band. Also a number from 1 to 3 in the file name indicates the corresponding mode used for the test.

II.4.3 *Directory of the test sequence files*

This section gives the name and the content of each file provided for the digital test sequences. Figure II-6/G.722 shows which files are to be used in the different configurations of test.

Class T1 file names

T1C1.XMT: 16 416 test values (16-bit words) containing various frequencies, d.c., white noise for encoder test.

T1C2.XMT: 800 test values (16-bit words) containing the artificial sequence to test overflow in the encoder.

T1D3.COD: 16 416 test values (16-bit words) containing the artificial sequence for the decoder test. The most significant 8 bits contain the ADPCM code (IH, IL) and the least significant 8 bits contain the RSS information (reset/synchronisation signal).

Class T2 file names

T2R1.COD: 16 416 test values (16-bit words) containing the output code for the T1C1.XMT file. This file is also used as an input to test the decoder, and consequently has the same structure as the T1D3.COD file.

T2R2.COD: 800 test values (16 bit words) containing the output code for the T1C2.XMT file. This file is also used as source to test the decoder and consequently has the same structure as the T1D3.COD file.

Class T3 file names

T3L1.RC1 16 416 test values (1 6-bit words) containing the output of the lower sub-band decoder in Mode 1 when the file T2R1.COD is used as an input.

T3L1.RC2 same meaning as for T3L1.RC1 file but when Mode 2 is used.

T3L1.RC3 same meaning as for T3L1.RC1 file but when Mode 3 is used.

T3H1.RC0 16 416 test values (16-bit words) containing the output of the higher sub-band decoder when the file T2R1.COD is used as an input.

T3L2.RC1 800 test values (16-bit words) containing the output of the lower sub-band decoder in Mode 1 when the file T2R2.COD is used as an input.

T3L2.RC2 same meaning as for T3L2.RC1 file but when Mode 2 is used.

T3L2.RC3 same meaning as for T3L2.RC1 file but when Mode 3 is used.

T3H2.RC0 800 test values (16-bit words) containing the output of the higher sub-band decoder when the file T2R2.COD is used as an input.

T3L3.RC1 16 416 test values (16-bit words) containing the output of the lower sub-band decoder in Mode 1 when the file T1 D3.COD is used as an input.

T3L3.RC2 same meaning as for T3L3.RC1 file but when Mode 2 is used.

T3L3.RC3 same meaning as for T3L3.RC1 file but when Mode 3 is used.

T3H3.RC0 16 416 test values (16-bit words) containing the output of the higher sub-band decoder when the file T1D3.COD is used as an input.

Note - Mode indication must be set by the user of the digital test sequences.

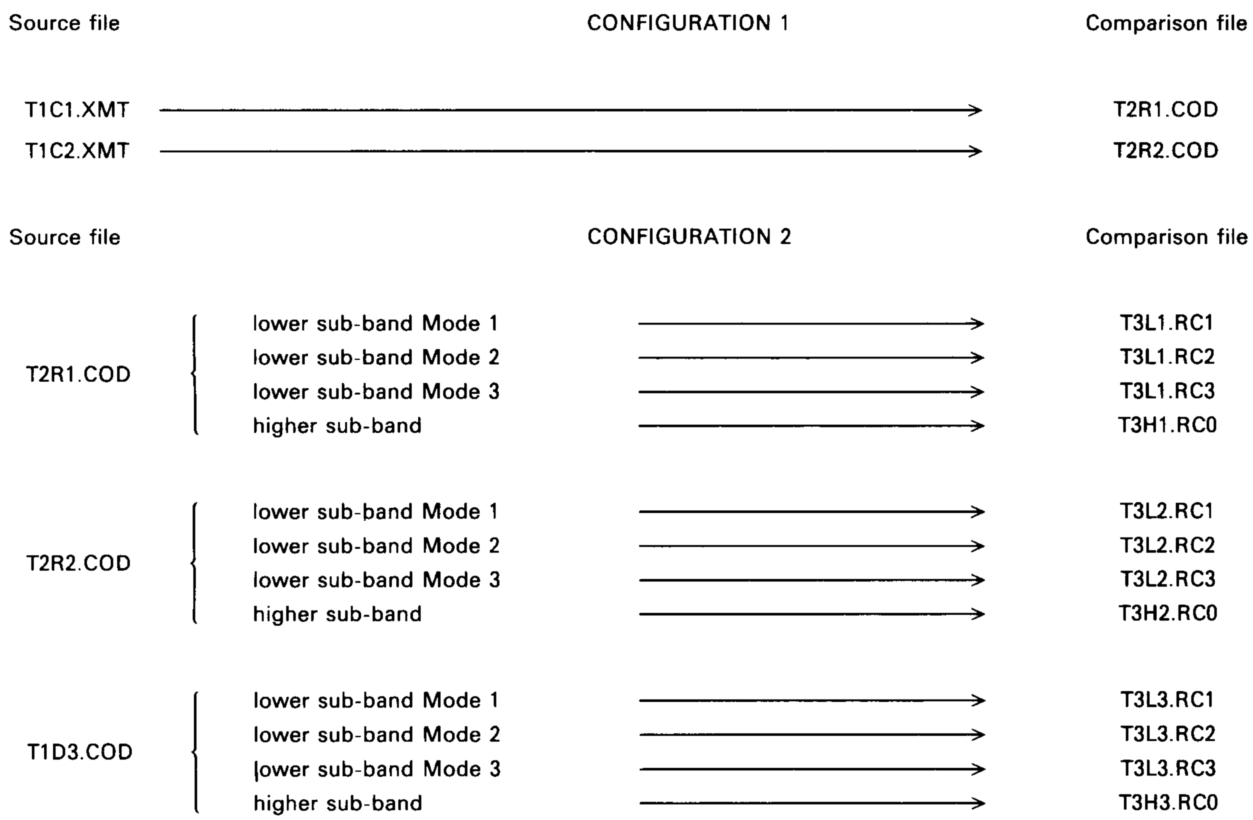


FIGURE II-6/G.722

Configuration of test

II.4.4 *File format description*

All the files are written in ASCII with a line structure. The first two lines of each file give some information on the file content. The following format is used for the two first lines:

/* CCITT 64 KBIT/S SB-ADPCM DIGITAL TEST SEQUENCE G.722 /*

/ * FILE NAME: xxxx.eee DATE: mm-dd-yy VERSION: V 1.0 */

For the subsequent lines of the file, 16 test values (16-bit words, 64 hexadecimal characters) are followed by a checksum on 1 byte (2 hexadecimal characters), a carriage return (ASCII code 0D in hexadecimal), and a line feed (ASCII code 0A in hexadecimal). These last two characters are non-printable.

The checksum is the two's complement of the least significant 8 bits of the summation of all the preceding characters (ASCII codes) in the line. If the least significant 8 bits of the summation are all zero, the corresponding two's complement is set all zero.

At the end of each file, a line of comment closes the file. This line is:

/* END OF FILE: xxxx.eee

II.4.5 *Internal line description*

II.4.5.1 *File with extension .XMT*

- 16 words of 16 bits with the LSB set to 1, all others set to zero (RSS = 1: reset mode);
- 16 384 or 768 words of 16 bits of digital test sequence with RSS = 0 (RSS is the LSB of the lower byte of the word);
- 16 words of 16 bits with the LSB set to 1, all others set to 0 (marks for end of test sequence).

II.4.5.2 *File with extension .COD*

- 16 words of 16 bits with the LSB set to 1, all others set to 0 (RSS = 1: reset mode and the ADPCM code set to 0);
- 16 384 or 768 words of 16 bits of digital test sequence with RSS = 0 (RSS is the LSB of the lower byte of the word and the upper byte is the ADPCM code);
- 16 words of 16 bits with the LSB set to 1, all others set to zero (marks for end of test sequence).

II.4.5.3 *File with extension .RCx*

- 16 words of 16 bits with the LSB set to 1, all others set to 0 (this means that these words are non-valid data);
- 16 384 or 768 words of 16 bits of digital test sequence with the LSB of the lower byte set to 0 to indicate valid data;
- 16 words of 16 bits with the LSB set to 1, all others set to 0 (marks for end of test sequence).

II.4.6 *Distribution of the CCITT digital test sequences*

The distribution of the digital test sequences comprises three $5^1/_4''$ MS-DOS flexible disks (2 sides, 360 K formatted). The directories of the disks are given in Table II-5/G.722.

TABLE II-5/G.722

Directory digital test sequence diskettes

	Directory		
	Filename	Extension	Number of bytes
Disk 1	T1C1	XMT	69 973
	T1C2	XMT	3 605
	T1D3	COD	69 973
	T2R1	COD	69 973
	T2R2	COD	3 605
Disk 2	T3L1	RC1	69 973
	T3L1	RC2	69 973
	T3L1	RC3	69 973
	T3H1	RC0	69 973
	T3L2	RC1	3 605
	T3L2	RC2	3 605
Disk 3	T3L2	RC3	3 605
	T3H2	RC0	3 605
	T3L3	RC1	69 973
	T3L3	RC2	69 973
	T3L3	RC3	69 973
	T3H3	RC0	69 973