5.Computational details for QMF

5.1Input 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.

5.2Description 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 the internal variables, XA, XB and WD, must be equal to or greater than 24 bits (see Note 1). 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 minimum or maximum representable value in case of underflow or overflow, respectively,

- denotes arithmetic subtraction with saturation control which forces the result to 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 right-hand variable for left-hand one.

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.

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

Attenuation/frequency response of the input antialiasing filter

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

(43)

(44)

4.2Quantizer adaptation

See s[s] 3.5.

4.3Adaptive prediction

4.3.1Predicted value computation

See s[s] 3.6.1.

4.3.2Reconstructed signal computation

See s[s] 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:

$$r_L(n) = S_L(n) + d_L(n) \qquad (40)$$

4.3.3Pole section adaptation

See s[s] 3.6.3.

4.3.4Zero section adaptation

See s[s] 3.6.4.

4.4Receive QMF

A 24-coefficient QMF is used to reconstruct the output signal, $x_{out}(j)$, 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:

(41)

(42)

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_{Lr}, 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, $_L(n)$:

$$d_L(n) = QL6^{-1}(I_{Lr}(n))._L(n).\mathrm{sgn}\,(I_{Lr}(n)) \qquad (39)$$

where $\mathrm{sgn}\,(I_{Lr}(n))$ is derived from the sign of $I_L(n)$ defined in equation (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 s^s 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 s^s 3.4.2.

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, $r_{Lt}(n)$ 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 s^s 3.6.2.

where sgn ($I_H(n)$) is derived from the sign of $e_H(n)$ defined in equaltion (10), and where $Q2^{-1}(I_H(n))$ 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.

3.5Quantizer adaptation

This block defines $__L(n)$ and $__H(n)$, 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, $__L(n)$ and $__H(n)$, are updated as follows:

$$_L(n) = B._L(n-1)÷W_L(I_{Lt}(n-1)) \quad (13)$$

$$_H(n) = B._H(n-1)÷W_H(I_H(n-1)) \quad (14)$$

where $W_L(.)$ and $W_y(.)$ are logarithmic scaling factor 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:

$$0 _ _L(n) _ 9 \quad (15)$$

$$0 _ _H(n) _ 11 \quad (16)$$

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

$$__L(n) = 2(_L(n)÷2)_min \quad (17)$$

$$__H(n) = 2 H(n)_min \quad (18)$$

where _min is equal to half the quantizer step size of the 14 bit analogue-to-digital converter.

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

mH	$Q2^{-1}$	WH
1	0.39453	-0.10449
2	1.80859	0.38965

TABLE 7/G.722

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

TABLE 8/G.722

TABLE 6/G.722

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

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

TABLE 5/G.722

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

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".

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 codeword, $I_{Lt}(n)$, 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, $_L(n)$:

$$d_{Lt}(n) = QL4^{-1}(I_{Lt}(n))._L(n).sgn(I_{Lt}(n)) \qquad (11)$$

where sgn $(I_{Lt}(n))$ is derived from the sign of $e_L(n)$ defined in equation 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
$Q_L4^{-1}(m_L)$ by reference to Table 7/G.722.

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, _H(n):

$$d_H(n) = Q2^{-1}(I_H(n))._H(n).sgn(I_H(n)) \qquad (12)$$

$$+ \mathrm{ILN}(m_L)\ ,\ \text{if } e_L(n) < 0$$

$$I_H(n) = \begin{cases} + \mathrm{IHP}(m_H)\ ,\ \text{if } e_H(n) _\ 0 \\ + \\ + \mathrm{IHN}(m_H)\ ,\ \text{if } e_H(n) < 0 \end{cases} \qquad (10)$$

for the lower and higher sub-bands respectively.

TABLE 4/G.722

Transmit and receive QMF coefficient values

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, LL6, LU6, HL and HU, are scaled by computed scale factors, $__L(n)$ and $__H(n)$ (see s[s] 3.5). Indices, m_L and m_H, are then determined to satisfy the following:

$$LL6(m_L).__L(n) _ e_L(n) < LU6(m_L).__L(n) \qquad (7)$$

$$HL(m_H).__H(n) _ e_H(n) < HU(m_H).__H(n) \qquad (8)$$

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:

$$I_L(n) = \begin{cases} + \\ + \end{cases} \begin{matrix} \\ ILP(m_L) \text{ , if } e_L(n) _ 0 \end{matrix} \qquad (9)$$

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.

(3291)

TABLE 3/G.722

Variables used in the SB-ADPCM encoder and decoder descriptions

Note - Variables used exclusively within one section are not listed. Subscripts L and H refer to

$$x_B = \sum_{i=0}^{11} \eta_{2\iota+1}.\xi_{\iota\nu}(\varphi - 2\iota - 1) \qquad (4)$$

3.2Difference signal computation

The difference signal, $e_L(n)$ 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 $x_H(n)$:

$$e_L(n) = x_L(n) - s_L(n) \qquad (5)$$

$$e_H(n) = x_H(n) - s_H(n) \qquad (6)$$

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 7 000 Hz and a level of +6 dBmO 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 dBmO. The measurement should be made with no signal at test point A, but with the test point correctly terminated.

2.6Transcoding 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 1.

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 $s^{s}s^{s}$ 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-l) 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-l) 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.

3.1Transmit 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:

$$x_L(n) = x_A + x_B \qquad (1)$$

$$x_H(n) = x_A - x_B \qquad (2)$$

where

$$x_A = \sum_{i=0}^{11} \eta_{2\iota}.\xi_{\iota\nu}(\varphi - 2\iota) \qquad (3)$$

should satisfy the limits of the mask shown in Figure 12/G.722. The nominal reference frequency is 1 000 Hz and the test level for the in-band characteristic is -10 dBmO. 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.

2.5.3Group 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.

2.5.4Idle noise for the receive audio part

The unweighted noise power of the receive audio part measured in the frequency range 50 to 7 000 Hz with a 14-bit all-zero signal at its input should not exceed -75 dBmO.

2.5.5Signal-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 7 000 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.

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

With a sine wave signal at a level of -10 dBmO 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 7 000 Hz at test point B should satisfy the limits of the mask shown in Figure 15/G.722.

2.5.7Variation of gain with input level

With a sine wave signal at the nominal reference frequency of 1 000 Hz, but excluding the submultiple 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 dBmO measured selectively at test point B, should satisfy the limits of the mask shown in Figure 16/G.722.

2.5.8Intermodulation

Under study.

2.5.9Go/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 7 000 Hz and at a level of +6 dBmO applied to test point A, the crosstalk level measured selectively at test point B should not exceed -64 dBmO. The

The nominal 3 dB bandwidth is 50 to 7 000 Hz.

2.4.2Attenuation/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 1 000 Hz and the test level is -10 dBmO.

2.4.3Absolute group delay

The absolute group delay, defined as the minimum group delay for a sinewave signal between 50 and 7 000 Hz, should not exceed 4 ms. The test level is -10 dBmO.

2.4.4Idle noise

The unweighted noise power measured in the frequency range 50 to 7 000 Hz with no signal at the input port (test point A) should not exceed -66 dBmO. When measured in the frequency range 50 Hz to 20 kHz the unweighted noise power should not exceed -60 dBmO.

2.4.5Single frequency noise

The level of any single frequency (in particular 8 000 Hz, the sampling frequency and its multiples), measured selectively with no signal at the input port (test point A) should not exceed -70 dBmO.

2.4.6Signal-to-total distortion ratio

Under study.

2.5Transmission 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 s^S 2.4 with the measurement configuration of Figure 9b/G.722.

2.5.1Attenuation/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 1 000 Hz and the test level for the in-band characteristic is -10 dBmO. Appropriate measurements should be made to check the out-of-band characteristic taking into account the aliasing due to the 16 kHz sampling.

2.5.2Attenuation/frequency response of the output reconstructing filter

The in-band and out-of-band attenuation/frequency response of the output reconstructing filter

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 $\pm 50.10^{-6}$.

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 output 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 dBmO $\pm$ 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 1 000 Hz is indicated below, the actual frequency should be chosen equal to 1 020 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

tized 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 $d_{L,4}$.

1.5.3Higher 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 s^s 1.4.3, the output being the reconstructed signal, r_H.

1.5.4Receive 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.

1.6Timing requirements

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.

1.4.4Multiplexer

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.

1.5Functional 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 $s^s s^s$ 1.5.1 to 1.5.4.

1.5.1Demultiplexer

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.

1.5.2Lower 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.

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 s^s 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 quan-

The mode control strategy could be derived from the auxiliary data channel protocol (see draft 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 s^s^s 1.4.1 to 1.4.4.

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 8 000 Hz into two sub-bands: the lower sub-band (0 to 4 000 Hz) and the higher sub-band (4 000 to 8 000 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 differencesignal 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 s^s 1.3 without causing misoperation in the decoder. Use of a 60- level quantizer (instead of 64-level) ensures the pulse density requirements as described in Recommendation G.802 are met under all conditions and in all modes of operation.

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.

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 1 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.

s^{s} 1.2 contains a functional description of the transmit and receive audio parts, s^{s} 1.3 describes the modes of operation and the implication of inserting data bits on the algorithms, whilst s^{s}s^{s} 1.4 and 1.5 provide the functional descriptions of the SB-ADPCM encoding and decoding algorithms respectively. s^{s} 1.6 deals with the timing requirements. s^{s} 2 specifies the transmission characteristics of the 64 kbit/s (7 kHz) audio codec and of the transmit and receive audio parts, s^{s}s^{s} 3 and 4 give the principles of the SB- ADPCM encoder respectively whilst s^{s}s^{s} 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: "Systems aspects for the use of the 7 kHz audio-codec within 64 kbit/s" contains specifications for in-channel handshaking procedures for terminal identification and for mode control strategy, including interworking with existing 64 kbit/s PCM terminals.

1.2Functional 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 s^{s} 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 s^{s} 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.

1.3Possible modes of operation and implications of inserting data

- 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.

10.Recommendation G.722:

7 kHz AUDIO-CODING WITHIN 64 KBIT/S

1. General

1.1Scope 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;