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.

H.T. [T24.722]

TABLE I-1/G.722

Relative levels of speech performance

(

Q

| *fBvalues)*

Mode of operation	Transcodings 1 { {		
1 (64 kbit/s)	45	38	41
2 (56 kbit/s)	43	36	38
3 (48 kbit/s)	38	29	34

Table I-1/G.722 [T24.722], p.

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 1×10^{D}lF261^{4}. High error ratios approaching 1×10^{D}lF261^{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.

H.T. [T25.722]

TABLE I-2/G.722

Relative speech performance under mode mismatch

(

Q

| fBvalues)

{	{	
	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.

Table I-2/G.722 [T25.722], p.

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 transmisssion 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, p.

Figure I-2/G.722, p.

Figure I-3/G.722, p.

Figure I-4/G.722, p.

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.

Figure I-5/G.722, p.

Figure I-6/G.722, p.

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 *Configuratins 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.

H.T. [T26.722]

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

Tableau II-1/G.722 [T26.722], p. 9

Figure II-1/G.722, p. 10

Figure II-2/G.722, p. 11

Figure II-3/G.722, p. 12

FIGURE II-4/G.722, p. 13

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, p.

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. [Formula Deleted]