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DIGITAL TRANSMISSION OF HIGH-QUALITY SOUND-PROGRAMME SIGNALS USING ONE, TWO OR THREE 64 kbit/s CHANNELS PER MONO SIGNAL (AND UP TO SIX PER STEREO SIGNAL)

ITU-T Recommendation J.52

(Previously "CCITT Recommendation")

FOREWORD

The ITU-T (Telecommunication Standardization Sector) is a permanent organ of the International Telecommunication Union (ITU). The ITU-T is responsible for studying technical, operating and tariff questions and issuing Recommendations on them with a view to standardizing telecommunications on a worldwide basis.

The World Telecommunication Standardization Conference (WTSC), which meets every four years, establishes the topics for study by the ITU-T Study Groups which, in their turn, produce Recommendations on these topics.

The approval of Recommendations by the Members of the ITU-T is covered by the procedure laid down in WTSC Resolution No. 1 (Helsinki, March 1-12, 1993).

ITU-T Recommendation J.52 was prepared by ITU-T Study Group 9 (1993-1996) and was approved under the WTSC Resolution No. 1 procedure on the 22nd of August 1994.

NOTE

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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Summary

If sound signals coded by new methods recommended by ITU-R Study Group 10 and standardized by ISO/IEC 11172-3 are to be transmitted in telecommunication networks, the properties of the network have to be taken into account. The N-ISDN allows the assembly of single channels with a bit rate of 64 kbit/s each. If the bit rate is greater than 64 kbit/s (2 times, 3 times), measures are necessary to maintain bit sequence. The equipment described in this Recommendation makes possible the transmission of source-coded sound signals on channels of N-ISDN or on connections with a 2048 kbit/s (or 1544 kbit/s) frame. Thereby, the ITU-T Recommendation H.221 is applied.

Recommendation J.52

DIGITAL TRANSMISSION OF HIGH-QUALITY SOUND-PROGRAMME SIGNALS USING ONE, TWO OR THREE 64 kbit/s CHANNELS PER MONO SIGNAL (AND UP TO SIX PER STEREO SIGNAL)

(Geneva, 1994)

The ITU-T,

considering

(a) that the ISO/IEC has approved the International Standard 11172-3 for the reduction of the bit rate of high-quality digital sound-programme signals;

(b) that ITU-R recommends for different applications of the transmission in a broadcast chain the same system for bit-rate reduction based on the ISO/IEC Standard 11172-3 and on extensive tests carried out by ITU-R Study Group 10;

(c) that with this system it is possible to transmit high-quality sound-programme signals with bit rates in the range of 64 to 192 kbit/s per mono channel;

(d) that ITU-T has approved some Recommendations which describe the structure and function of 64 kbit/s channels;

(e) that it is desirable to use such standardized 64 kbit/s channels for the transmission of high-quality sound signals;

(f) that the bit stream to be transmitted on 64 kbit/s channels should be protected optionally by means of error-correction measures,

recommends

that for transmission of high-quality sound-programme signals with reduced bit rate on one, two or three 64 kbit/s channels per mono signal (and up to six per stereo signal), the system given in Annex A should be used.

Annex A

(This annex forms an integral part of this Recommendation)

Introduction

If sound signals coded by ISO/IEC 11172-3 have to be transmitted in telecommunication networks, the properties of the networks are to be taken into account. The N-ISDN allows the assembly of single channels with the bit rate of 64 kbit/s. If the bit rate is greater than 64 kbit/s (two times, three times), measures are necessary to maintain the bit sequence. The equipment described in the following makes the transmission of source-coded sound signals on channels of N-ISDN or on connections with a 2048 kbit/s (or 1544 kbit/s) frame possible.

A.1 General features

A.1.1 Purpose of the equipment

The purpose of the equipment is to process a bit-rate reduced high-quality sound-programme signal for the transmission on standardized 64 kbit/s channels. For this purpose, it is appropriate that the source encoder and the multiplexer form a unit with an output to the line interface (see Figure A.1).

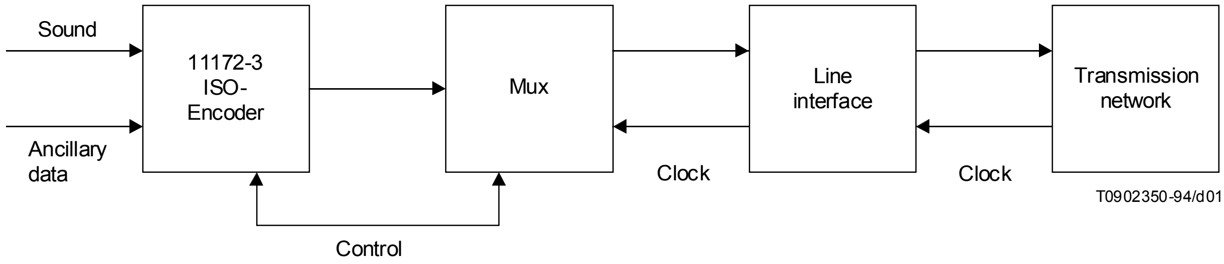

FIGURE A.1/J.52

Connection between encoder, multiplexer and line interface

A.1.2 Incoming signals (PCM level)

The source encoder provides a bit stream according to ISO Standard 11172-3 (Layer I, Layer II or Layer III). Layer I is not recommended for broadcast applications, the block structure of Layer II and Layer III with the block length 24 ms (in case of 48 kHz sampling frequency) is shown in Figure A.2.

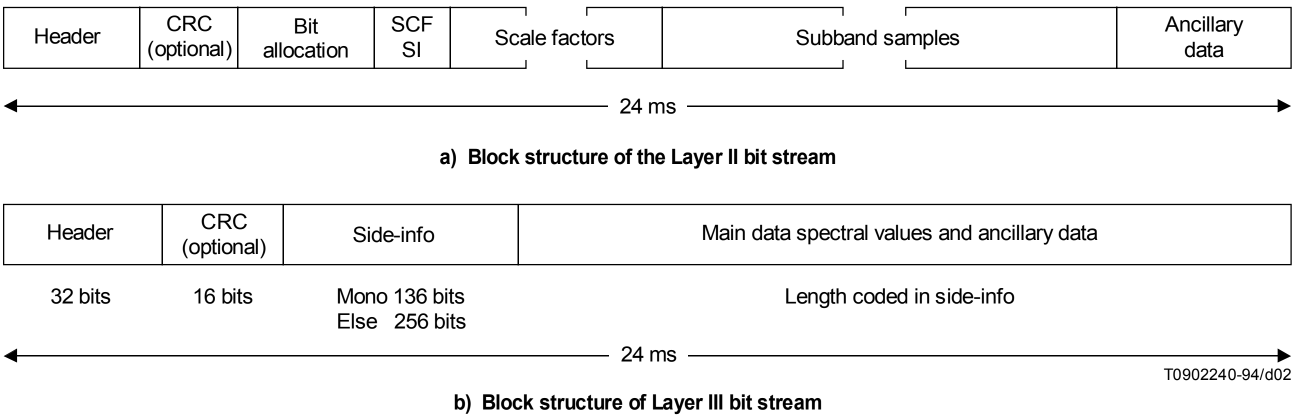

FIGURE A.2/J.52

If possible, this bit stream has to be transmitted unchanged. But in most cases this is impossible for the following reasons:

– additional framing bits are necessary for the transmission on the ISDN B-channels;

– if it is necessary to include error control in order to correct the bit errors, another coding hierarchy level has to be used.

Table A.1 gives all frame lengths defined in ISO 11172-3.

If the padding bit in the header is set, the frame contains an additional slot (1 byte for Layers II and III, 4 bytes for Layer I). The bit rate is encoded with 4 bit (bit-rate index). The coding of the bit rate is different for the three layers (see ISO 11172-3). Layer I and sampling frequency 44.1 kHz are not recommended for broadcast applications.

TABLE A.1/J.52

Possible frame lengths in bytes (without padding)

Bit rate (bit/s)	Available for Layer	Sampling frequency (Hz)		
		32 000	44 100	48 000
32 000	I, II, III	144	104	96
40 000	III	180	130	120
48 000	II, III	216	156	144
56 000	II, III	252	182	168
64 000	I, II, III	288	208	192
80 000	II, III	360	261	240
96 000	I, II, III	432	313	288
112 000	II, III	504	365	336
128 000	I, II, III	576	417	384
160 000	I, II, III	720	522	480
192 000	I, II, III	864	626	576
224 000	I, II, III	1008	731	672
256 000	I, II, III	1152	835	768
288 000	I	1296	940	864
320 000	I, II, III	1440	1044	960
352 000	I	1584	1149	1056
384 000	I, II	1728	1253	1152
416 000	I	1872	1358	1248
448 000	I	2016	1462	1344

A.2 Network interface formats

There is a distinction between dial-up connections (ISDN) and permanent connections. This distinction consists of the different routing of several B-channels constituting a virtual sound channel of more than 64 kbit/s, which requires a delay difference compensation in case of dial-up connections.

Two transmission formats are required:

– transmission of *one* mono/stereo signal using 1 to 6 64 kbit/s channels or a single H_0 channel in a switched ISDN network;

– transmission of *one or more* mono/stereo signals using, for example, H_0 or H_1 channels for permanent connections.[1)]

A.2.1 Dial-up connections (switched ISDN network)

According to Recommendation I.412 the following possibilities exist:

a) *B-channel interface structures:*

  i) basic interface structure: 2 B-channels + 1 D-channel (16 kbit/s);

  ii) primary rate B-channel interface structures:

    – 1544 kbit/s primary rate: 23 B + 1 D (64 kbit/s);

    – 2048 kbit/s primary rate: 30 B + 1 D (64 kbit/s);

b) *H-channel interface structures:*

  i) primary rate interface H_0-channel structures:

    – 1544 kbit/s primary rate: 4 H_0;
3 H_0 + D.

    – 2048 kbit/s primary rate: 5 H_0 + D.

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1) For transmission of *one* mono/stereo signal at a bit rate of 1 to 6 × 64 kbit/s on permanent connections, the X.21 leased circuit option can be used. In this case H.221 framing is optional.

For the following parameters, appropriate means have to be provided:

- clock synchronization between source coder and network;
- time synchronization (delay difference compensation) between several B-channels.

A.2.2 Permanent connections

2048 kbit/s or 1544 kbit/s signals are used. Framing is according to ITU-T Recommendation G.704.

The following parameters need consideration:

- clock synchronization between source coder and network;
- octet sequence integrity within the same frame;
- time-slot allocation.

A.3 Synchronization and frame alignment

A.3.1 Dial-up connections

For all 64 kbit/s channels of the basic interface structure in dial-up connections the framing according to Recommendation H.221 (in conjunction with Recommendation ITU-T H.242) should be used. With the use of the H.221 framing there is the possibility to achieve both clock synchronization and time synchronization between at maximum six B-channels. This is permanently provided, also in case of a change of the routing during the transmission.

Furthermore, the addition of H.221 framing to dial-up connections (H_0 or 1 to 6 B-channels) provides a control channel harmonized with other N-ISDN audiovisual equipments.

A.3.1.1 Basic interface structure

To achieve the synchronization according to Recommendation H.221, an overhead [consisting of a frame alignment signal (FAS) and bit-allocation signal (BAS)] of 1.6 kbit/s is required per 64 kbit/s channel or in the first time slot of an H_0 channel.

For the transmission of bit-rate reduced high-quality sound signals, Recommendation H.221 has to be extended. The details are indicated in Appendix I.[2)]

A.3.1.2 Realization of bit rates which are not explicitly listed in the bit-rate table of the ISO/IEC 11172-3 Standard

Figure A.3 describes the different sections of the transmission system. The network provides data channels with 64 kbit/s each. If H.221 framing and an additional error protection is applied, the full channel capacity cannot be used by the ISO 11172-3 encoder for audio data.

The ISO/IEC 11172-3 bit stream is sub-divided in audio frames corresponding to a sequence of 384 PCM audio samples for Layer I and 1152 PCM audio samples for Layer II and Layer III. The audio frame length (see Table A.1) is dependent on the bit-rate index, the sampling frequency and the status of the padding bit, an information which is given in the audio frame header. The ISO/IEC 11172-3 Standard allows for 14 different, explicitly listed bit rates, indicated by the so-called bit-rate index value. An additional format, the "free format", i.e. a user-defined audio frame length, can be chosen by using the bit-rate index "0000".

Three methods may be used in order to realize those bit rates which are not explicitly listed in the bit-rate tables of the ISO/IEC 11172-3 Standard. These methods are:

- free format;
- using the ancillary data field;
- dynamic bit-rate switching.

2) The extension of Recommendation H.221 is the responsibility of ITU-T Study Group 15. If these proposals are accepted by ITU-T Study Group 15, Appendix I can be deleted.

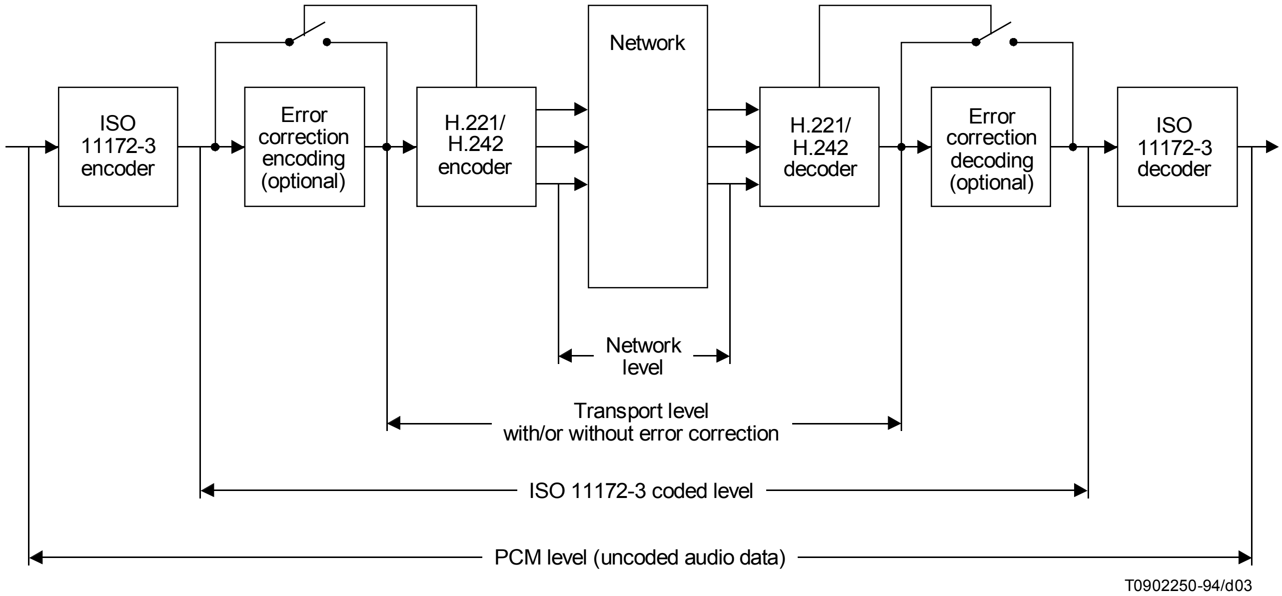

FIGURE A.3/J.52

Transmission system with transport format and optional error control

Table A.2 gives as an example (for a 48 kHz sampling frequency) the details for an adaption of the explicitly listed bit rates according to ISO/IEC 11172-3 to the available channel bit rate for the methods "Using the ancillary data field" and "Dynamic bit-rate switching". The contents of this table are:

Channels: Number of used 64 kbit/s channels

Mode:

- Inband: H.221 framing is used within each subchannel
- Inband 56k H.221 framing is used within 56 kbit/s channel
- Split Only parts of the subchannel are used for audio data transmission and/or H.221 framing
- Unframed Only one subchannel is used and no H.221 framing is used
- Unframed 56k Unframed mode for 56 kbit/s subchannel
- Permanent No H.221 framing is used within the subchannels

Useful data rate: Data rate available for ISO 11172-3 audio encoder.

The following values are given in Table A.2:

- L Nominal frame length (as defined in ISO/IEC 11172-3)
- L1 Nominal frame length (as defined in ISO/IEC 11172-3) for a short frame
- L2 Nominal frame length for a long frame
- dL Average frame length
- P Switching period of frame length
- R Number of reserved bytes per frame
- dR Average number of bytes per frame necessary for H.221
- I1 Number of frames using L1
- I2 Number of frames using L2

TABLE A.2/J.52

Adaption of the explicitly listed bit rates according to ISO/IEC 11172-3 to the available channel bit rate

			No protection								
Channels	Mode	Useful data rate	Dynamic bit-rate switching						Using the ancillary data field		
			L1 (Byte)	I1	L2 (Byte)	I2	P	dL	L (Byte)	R (Byte)	dR (Byte)
1	inband	62 400	168	1	192	4	5	187.2	192		
2	inband	124 800	336	1	384	4	5	374.4	384		
3	inband	187 200	480	3	576	17	20	561.6	576		
4	inband	249 600	672	1	768	4	5	748.8	768		
5	inband	312 000	768	1	960	7	8	936	960		
6	inband	374 400	–	–	–	–	–	–	1152		
1	inband 56 k	54 400	144	1	168	4	5	163.2	168		
1	split	32 000	96	1					96		
1	split	40 000	120	1					120		
1	split	48 000	144	1					144		
1	unframed 56 k/split	56 000	168	1					168		
1	unframed	64 000	192	1					192		
2	permanent	80 000	240	1					240		
2	permanent	96 000	288	1					288		
2	permanent	112 000	336	1					336		
2	permanent	128 000	384	1					384		
3	permanent	160 000	480	1					480		
3	permanent	192 000	576	1					576		
4	permanent	224 000	672	1					672		
4	permanent	256 000	768	1					768		
5	permanent	320 000	960	1					960		
6	permanent	384 000	1152	1					1152		

A.3.1.2.1 Free format

The so-called "free format" which is described for all layers in subclause 2.4.2.3 of ISO/IEC 11172-3 can be used to adapt the bit rate to every desired value fully in accordance with the Standard. Using this method, neither special formatters nor reformatters are needed. The free format is indicated by the bit-rate index in the header of an ISO/IEC 11172-3 audio frame. The length of the frame in bytes can be calculated by:

$$n \text{ bytes} = (1152/\text{fs}) \ (\text{bir}/8)$$

with

fs is the sampling frequency in kHz; and

bir is the bit rate in kbit/s.

In the encoder, the length of the audio frame has to be calculated according to the formula given above. After start-up, the decoder has to determine the distance between consecutive sync words. Then a flywheel synchronization process can be applied as in the case of the 14 predefined bit rates in the ISO/IEC Standard.

If a bit rate of for example, 124.8 kbit/s has to be realized with a sampling frequency of 48 kHz, the length of the audio frame will be:

$$(1152/48) \ (124.8/8) = 374.4 \text{ bytes}$$

For those bit rates which result in a non-integer number of bytes in a frame, the required bit rate can be realized by using padding. The details how padding should be applied, are described in subclause 2.4.2.3 of ISO/IEC 11172-3.

A.3.1.2.2 Using the ancillary data field

One of the 14 explicitly listed bit rates can be chosen which can be either equal or higher than the required bit rate. By constraining the bit allocation in the encoder, a certain amount of bits can be reserved for ancillary data. The length of this field is completely flexible. The number of bytes (R, Table A.2) necessary to achieve the synchronization according to Recommendation H.221 can be taken from the ancillary data field.

In case of error-control modes 1, 2 and 3, R is the sum of bytes used for H.221 coding and error control.

For Layer I and Layer II, this ancillary data field is always located at the very end of the audio frame, just ahead of the next sync word.

If for example, a bit rate of 128 kbit/s according to the bit-rate index of the ISO/IEC Standard is selected, only 124.8 kbit/s are available for the encoded audio data and Programme Associated Data (PAD). A bit rate of 3.2 kbit/s, i.e. 76.8 bits in average (with a sampling frequency of 48 kHz), has to be reserved in the ISO audio frame. The bit allocation of the encoder has to be set to reserve *n* bytes at the end of the audio frame, with *n* equal to an integer number of bytes, providing the capacity for the required bit rate for H.221 framing. This integer number of bytes does not vary in time. The bits within these bytes not used for H.221 framing shall be set to zero.

See Figure A.4.

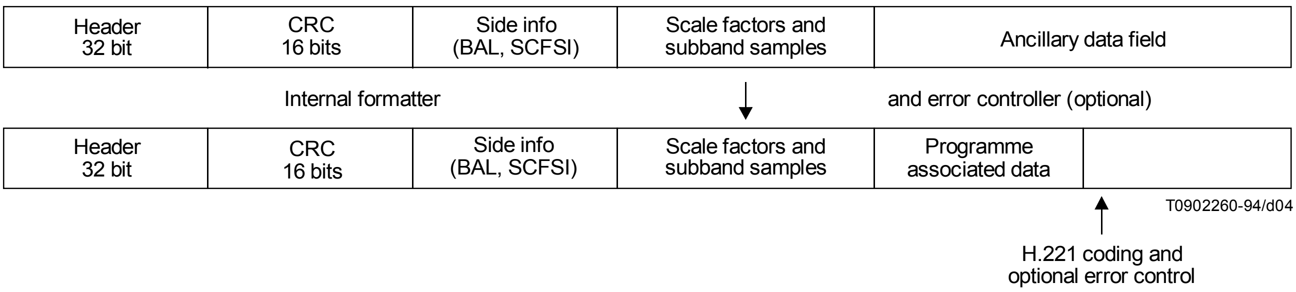

FIGURE A.4/J.52

ISO/IEC 11172-3 Layer II frame with defined ancillary data field for programme associated data (PAD) and H.221 coding inclusive optional error control

The ancillary data bits provided for the H.221 coding and the optional error control should be removed before the H.221 encoder or skipped inside the H.221 encoder before transmission. If these bits have been removed, a corresponding number of dummy bits has to be inserted at the end of an audio frame by the H.221 decoder before decoding by an ISO/IEC 11172-3 decoder.

The nominal length of the frame L, as defined by the bit-rate indices in ISO/IEC 11172-3 and the number of reserved bytes R, necessary for H.221 coding, are given in Table A.2. If additional error control is used, the error-control bits are inserted at the very end of the frame, i.e. after the bits needed for H.221 coding. In this mode R is the sum of the bytes used for the H.221 coding and the error control.

This method is the preferred method for Layer II.

A.3.1.2.3 Dynamic bit-rate switching

For Layer III the audio frame length can be changed dynamically from frame to frame. Using this method, on average, additional bit rates which are not listed in the bit-rate table of the ISO/IEC 11172-3 Standard are supported. The sequence of frames with different bit rates called switching period (P) which is necessary to realize the required bit rate has to be determined in the encoder. Hereby the following formula has to be considered:

$$\text{Bit rate}_{/\text{kbit/s}} = \text{Average frame length}_{/\text{bits}} \times \text{Sampling frequency}_{/\text{kHz}}/1152$$

For example, for a bit rate of 62.4 kbit/s and a sampling frequency of 48 kHz a sequence of one frame containing 168 bytes = 1344 bit (corresponding to a bit rate of 56 kbit/s) and four frames containing 192 bytes (corresponding to a bit rate of 64 kbit/s) has to be used. This sequence results in an average frame length of 187.2 bytes = 1497.6 bits. The switching period (P), the length of a short frame (L1) and a long frame (L2) is shown in Table A.2. I1 and I2 are the number of frames with the length L1 or L2 respectively within a switching period P. In case of error-control modes 1, 2 and 3, R is the sum of the bits used for the H.221 coding and the error control.

If this method is used for Layer I and Layer II, it has to be considered that the decoder implementation shall allow for accepting a change of the bit-rate index and an additional buffer is required.

Layer III has a built-in buffer. If a long frame is used some bits can be stored in this internal buffer and used for the next frame if necessary to achieve a constant average bit rate for the encoding.

Example: average bit rate 62.4 kbit/s, 48 kHz sampling frequency.

Bit-rate index	*Bits used for encoding*	*Bits put into/ taken from the buffer*
"64 000"	1497	39
"64 000"	1497	39
"64 000"	1498	38
"64 000"	1498	38
"56 000"	1498	–154

This method is the preferred method for Layer III.

A.3.2 Transmission on permanent connections

The structure of a 2048 kbit/s frame consists of 32 time slots (TS) with a capacity of 64 kbit/s each. The structure of a 1544 kbit/s consists of 24 TS, also with the capacity of 64 kbit/s per TS.

TS 0 is used for frame alignment. For 2 Mbit/s frames, TS 16 is reserved for signalling and other network purposes. In case of channel-associated signalling, transmission of low-speed data channels (associated with sound channels) in TS 16 is possible.

In case of error-control modes 1, 2, 3 capacity for redundancy is necessary. This capacity either

- is taken from the ancillary data field (preferred at Layer II); or
- is provided by dynamic bit-rate switching (preferred at Layer III).

In 2 Mbit/s (or 1.5 Mbit/s) connections there are two operation modes, a flexible multiplex mode and a fixed multiplex mode. In the flexible multiplex mode only time slot TS 1 has the H.221 framing, all other TS are free of framing and can transmit the full capacity of 64 kbit/s. In the fixed multiplex mode all TS can transmit the full capacity of 64 kbit/s without H.221 framing. The channels are allocated in a fixed order.

A.3.2.1 Flexible multiplex

In this flexible multiplex mode the Multiple Byte Extension (MBE) is used (see Appendix I)[3]. TS 1 has a frame structure according to Recommendation H.221. It is not to be used for the transmission of sound signals, but it contains FAS and BAS and it can be used for the transmission of one G.722 channel and/or low-speed data (LSD). The audio signals are transmitted in the TS 2 to 15 and 17 to 31 for 32 TS systems, and in the TS 2 to 23 for 24 TS systems.

Each TS except TS 1 is an unframed 64 kbit/s channel with the full capacity, which means it is possible to transmit:

- 64 kbit/s in one TS;
- 128 kbit/s in two TS;
- 192 kbit/s in three TS, etc.

[3] Further study on the use of MBE is required.

In case of transmission of more than 64 kbit/s, two or more TS constitute a virtual channel.

In Appendix I the details for control and signalling purposes (capability exchange, command exchange) are indicated according to the extension of Recommendation H.221.

A.3.2.2 Fixed multiplex

The fixed multiplex mode is established via 2048 kbit/s frames according to Recommendations G.704, G.735 and G.737.

The times slots could be allocated individually (according to Recommendation G.704), but if the frame is to be shared with J.41 encoded channels, the following TS allocations are recommended:

- in case of transmission of more than 64 kbit/s, two or more TS constitute a virtual channel.

In this case the following distribution is recommended:

- unidirectional 128 kbit/s access. The TS allocation is given in Table A.3.

TABLE A.3/J.52

	A	B	C	D	E
1	1 – 17	4 – 20	7 – 23	10 – 26	13 – 29
2	2 – 18	5 – 21	8 – 24	11 – 27	14 – 30
3	3 – 19	6 – 22	9 – 25	12 – 28	15 – 31
NOTE – The fifteen possible 128 kbit/s channels in a 2048 kbit/s stream are numbered A1 to E3.					

- unidirectional 192 kbit/s access. The TS allocation is given in Table A.4.

TABLE A.4/J.52

	A	B	C	D	E
a	1 – 2 – 3	4 – 5 – 6	7 – 8 – 9	10 – 11 – 12	13 – 14 – 15
b	17 – 18 – 19	20 – 21 – 22	23 – 24 – 25	26 – 27 – 28	29 – 30 – 31
NOTE – The ten possible 192 kbit/s channels in a 2048 kbit/s stream are numbered Aa to Eb. Preferably the channel pairs Aa-Ab and Ba-Bb and Ca-Cb and Da-Db and Ea-Eb should be used for stereophonic transmission.					

A.3.3 Synchronization of the sampling frequencies to the clock frequency

In real-time transmission systems, proper synchronization of encoder and decoder is a very important topic. A synchronous or an asynchronous operating mode can be used.

A.3.3.1 Synchronous operating mode

Usually, the transmission system provides the master clock at both sides, i.e. the sampling clock at the input and output of the system is synchronized by the transmission clock (see Figure A.5). Therefore, with digital audio inputs, either the audio source has to be synchronized to the transmission clock or a sample rate converter has to be used.

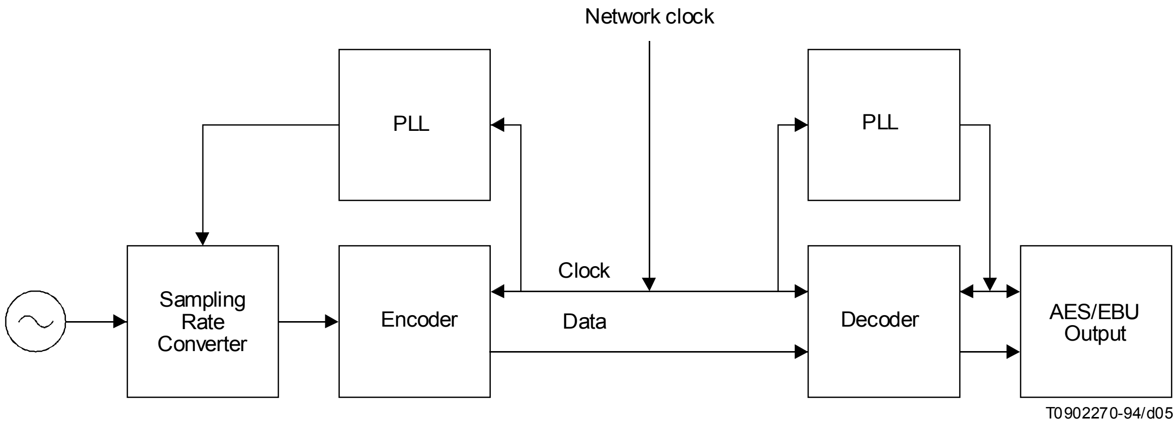

FIGURE A.5/J.52

Synchronous operating mode

A.3.3.2 Asynchronous operating mode

The sampling clock of the audio input is asynchronous to the network clock. The codec has to adapt the ratio between sampling frequency and bit rate to the network clock.

With Layer III, the use of any data rate is possible. Therefore, a completely asynchronous operation is feasible. The encoder counts the number of data bits that have been transmitted within a certain time interval and compares it with the target number (the nominal data rate, multiplied by the time interval). If the actual number of data bits is too small, the encoder has to use a frame with a smaller data rate for the next frame and vice versa. Using a similar approach, the decoder controls its sampling frequency to set it to the same value as in the encoder (see Figure A.6).

If H.221 framing is applied, asynchronous operating mode is also possible for Layer II using padding (except the data rate 320 kbit/s).

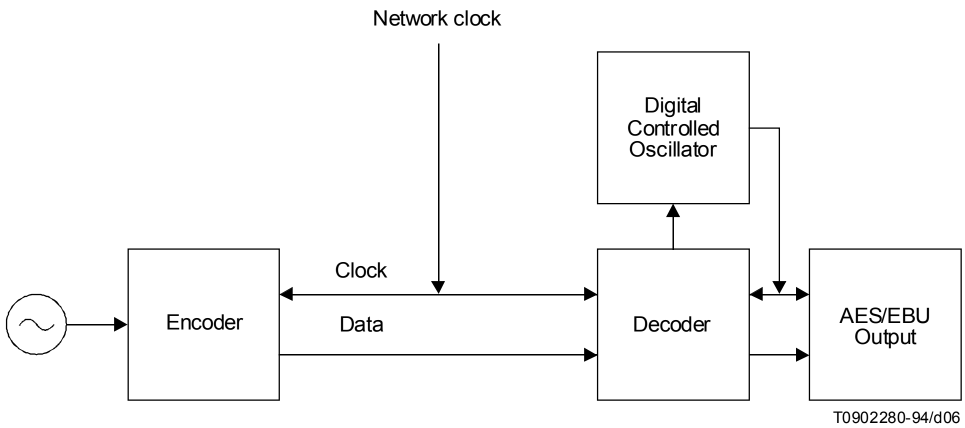

FIGURE A.6/J.52

Asynchronous operating mode

A.4 Error control

The use of the ISO CRC in the source coded audio signal is mandatory for the transmission application. (The CRC bits allow error detection in the header and in the most important parts of the side information.)

If Recommendation H.221 is applied, the CRC 4 procedure has to be used. Both ISO CRC and CRC 4 can provide to detect errors, but not correct them. If necessary, error correction can be applied in addition.

The following four error-control modes are provided:

Mode 0:	only ISO CRC	
Mode 1:	unequal error control	typical redundancy $r \approx 1.0\%$
Mode 2:	low equal error control	typical redundancy $r \approx 2.5\%$
Mode 3:	high equal error control	typical redundancy $r \approx 10\%$

The error control is performed by:

- error correction by a Reed-Solomon code; and
- error concealment with the CRC 16.

If the Reed-Solomon code (RS code) is overloaded, then the error concealment, based on the ISO CRC 16, should be applied. The error-control modes 1, 2, 3 use the same coding scheme.

Symbol length:	m = 8 bits (1 byte)
Code word length:	N byte (variable)
Code dimension:	K = N – 4 bytes
Field generator polynomial:	$f(x) = x^8 + x^4 + x^3 + x^2 + 1$
Code generator polynomial:	$g(x) = \Pi_{i=1}^{4} \left(x + \alpha^{125+i}\right)$
	$= x^4 + \alpha^{201}x^3 + \alpha^{246}x^2 + \alpha^{201}x + 1$

A.4.1 Unequal error correction

This error-control mode is currently only specified for Layer II. The application for Layer I and Layer III needs further consideration.

In any frame, error correction applies to:

- bits 16 ... 31 of the header,
- bits of CRC check,
- bits of allocation,

and the maximum number of bits in:

- scale factor select information,
- scale factors.

Other parts of the signal are not protected.

Single channel signals always use one Reed-Solomon code word per frame. Stereo and dual channel signals at low bit rates use one and at high bit rates two Reed-Solomon code words per frame (see Tables A.5, A.6 and A.7).

If there are two code words, they are interleaved byte by byte according to the following scheme: The first transmitted byte is the first byte of the code-word one, the second transmitted byte is the first byte of the code-word two, the third transmitted byte is the second byte of code-word one, and so on. This means that the information itself is not interleaved.

Redundancy of code $r_{code} = \frac{4}{N} 100\ (\%)$

Redundancy per frame $r_{frame} = 4 \frac{\text{code words per frame}}{\text{byte per frame}} 100\ (\%)$

TABLE A.5/J.52

Code parameters for a sampling frequency of 48 kHz
(unequal error-control mode – mode 1 – for Layer II only)

Bit rate (kbit/s)	Byte per frame	Single channel				Stereo or dual channel			
		No. code words	K (byte)	r_{code} (%)	r_{frame} (%)	No. code words	K (byte)	r_{code} (%)	r_{frame} (%)
32	96	1	28	12.5	4.2	–	–	–	–
48	144	1	28	12.5	2.8	–	–	–	–
56	168	1	83	4.6	2.4	–	–	–	–
64	192	1	83	4.6	2.1	1	51	7.3	2.1
80	240	1	83	4.6	1.7	–	–	–	–
96	288	1	83	4.6	1.4	1	51	7.3	1.4
112	336	1	83	4.6	1.2	2	81	4.7	2.4
128	384	1	83	4.6	1.0	2	81	4.7	2.1
160	480	1	83	4.6	0.8	2	81	4.7	1.7
192	576	1	83	4.6	0.7	2	81	4.7	1.4
224	672	–	–	–	–	2	81	4.7	1.2
256	768	–	–	–	–	2	81	4.7	1.0
320	960	–	–	–	–	2	81	4.7	0.8
384	1152	–	–	–	–	2	81	4.7	0.7

TABLE A.6/J.52

**Code parameters for a sampling frequency of 44.1 kHz
(unequal error-control mode – mode 1 – for Layer II only)**

		Single channel				Stereo or dual channel			
Bit rate (kbit/s)	Byte per frame	No. code words	K (byte)	r_{code} (%)	r_{frame} (%)	No. code words	K (byte)	r_{code} (%)	r_{frame} (%)
32	104.5	1	28	12.5	3.8	–	–	–	–
48	156.7	1	28	12.5	2.6	–	–	–	–
56	182.9	1	83	4.6	2.2	–	–	–	–
64	209.0	1	83	4.6	1.9	1	51	7.3	1.9
80	261.2	1	83	4.6	1.5	–	–	–	–
96	313.5	1	91	4.2	1.3	1	51	7.3	1.3
112	365.7	1	91	4.2	1.1	2	81	4.7	2.2
128	418.0	1	91	4.2	1.0	2	81	4.7	1.9
160	522.4	1	91	4.2	0.8	2	81	4.7	1.5
192	626.9	1	91	4.2	0.6	2	89	4.3	1.3
224	731.4	–	–	–	–	2	89	4.3	1.1
256	835.9	–	–	–	–	2	89	4.3	1.0
320	1044.9	–	–	–	–	2	89	4.3	0.8
384	1253.9	–	–	–	–	2	89	4.3	0.6

TABLE A.7/J.52

Code parameters for a sampling frequency of 32 kHz
(unequal error-control mode – mode 1 – for Layer II only)

		Single channel				Stereo or dual channel			
Bit rate (kbit/s)	Byte per frame	No. code words	K (byte)	r_{code} (%)	r_{frame} (%)	No. code words	K (byte)	r_{code} (%)	r_{frame} (%)
32	144	1	39	9.3	2.8	–	–	–	–
48	216	1	39	9.3	1.9	–	–	–	–
56	252	1	83	4.6	1.6	–	–	–	–
64	288	1	83	4.6	1.4	1	74	5.1	1.4
80	360	1	83	4.6	1.1	–	–	–	–
96	432	1	91	4.2	0.9	1	74	5.1	0.9
112	504	1	91	4.2	0.8	2	81	4.7	1.6
128	576	1	91	4.2	0.7	2	81	4.7	1.4
160	720	1	91	4.2	0.6	2	81	4.7	1.1
192	864	1	91	4.2	0.5	2	89	4.3	0.9
224	1008	–	–	–	–	2	89	4.3	0.8
256	1152	–	–	–	–	2	89	4.3	0.7
320	1440	–	–	–	–	2	89	4.3	0.6
384	1728	–	–	–	–	2	89	4.3	0.5

A.4.2 Equal error correction

The RS code defined above is used for protecting a complete frame. The more efficient error control of mode 3 is gained by using more and shorter code words per frame. The L code words per frame are interleaved, in order to increase the error-correction capability in case of burst errors. By this simple coding scheme only one encoder implementation, one decoder implementation and a flexible interleaver is required for all applications considered in this subclause. See Figure 7 for examples.

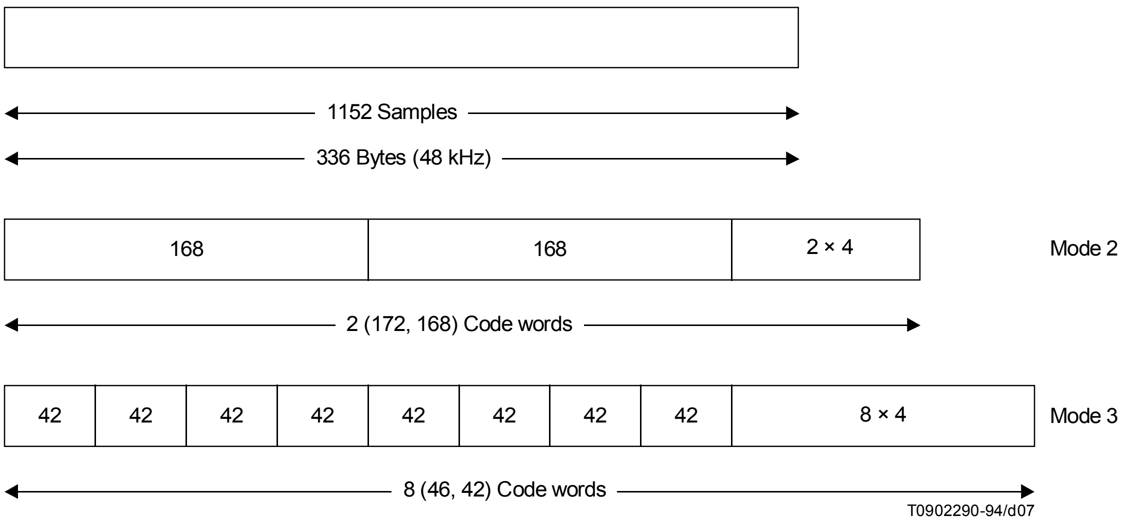

FIGURE A.7/J.52

Examples for the coding scheme before interleaving

For the sampling frequencies 48 kHz, 44.1 kHz and 32 kHz and for different bit rates, the code length N and the number L of code words per frame are given in Tables A.8, A.9 and A.10.

In many cases two different code lengths N and N – 1 are necessary in each frame. The number of code words of length N is denoted by L_N and the number of code words of length N – 1 is denoted by L_{N-1}, where $L = L_N + L_{N-1}$. The redundancy

$$r = \frac{4L}{\text{No. of bytes per frame}}$$

considers an additional prefix byte, which includes the information about the chosen coding level.

If the number of bytes per frame is not an integer (see Table A.9), the following modifications periodically occur:

for $L_{N-1} > 0$:		$L_N \leftarrow L_N + 1$	$L_{N-1} \leftarrow L_{N-1} - 1$, or
for $L_{N-1} = 0$	$N \leftarrow N + 1$	$L_N \leftarrow 1$	$L_{N-1} \leftarrow L - 1$

For details of the performance, see Appendix II.

A.4.2.1 Interleaving

Block interleaving is proposed, where the first transmitted byte is the first byte of the first code word, the second transmitted byte is the first byte of the second code word, etc. The interleaving scheme is shown in Figure A.8. Interleaving is accomplished such that byte No. $(i + k \cdot L)$ of each frame ($i = 1,2, ... L$, and $k = 0,1, ... N – 5$) is the $(k + 1)^{th}$ unaltered information byte of the i^{th} code word. The encoding of each frame starts with encoding of the L_N code words of length N byte.

TABLE A.8/J.52

Code parameters for a sampling frequency of 48 kHz

Sampling frequency 48 kHz													
Bit rate (bit/s)	No. bit per frame	No. byte per frame	N	No. code words			r (%)	N	No. code words			r (%)
				L	L_N	L_{N-1}			L	L_N	L_{N-1}	
32 000	768	96	100	1	1	0	4.17	52	2	2	0	8.33
40 000	960	120	124	1	1	0	3.33	44	3	3	0	10.00
48 000	1 152	144	148	1	1	0	2.78	52	3	3	0	8.33
56 000	1 344	168	172	1	1	0	2.38	46	4	4	0	9.52
64 000	1 536	192	196	1	1	0	2.08	43	5	2	3	10.42
80 000	1 920	240	244	1	1	0	1.67	44	6	6	0	10.00
96 000	2 304	288	148	2	2	0	2.78	46	7	1	6	9.72
112 000	2 688	336	172	2	2	0	2.38	46	8	8	0	9.52
128 000	3 072	384	196	2	2	0	2.08	47	9	6	3	9.38
160 000	3 840	480	164	3	3	0	2.50	44	12	12	0	10.00
192 000	4 608	576	196	3	3	0	2.08	46	14	2	12	9.72
224 000	5 376	672	172	4	4	0	2.38	44	17	9	8	10.12
256 000	6 144	768	158	5	3	2	2.60	45	19	8	11	9.09
320 000	7 680	960	164	6	6	0	2.50	44	24	24	0	10.00
384 000	9 216	1 152	169	7	4	3	2.43	44	29	21	8	10.07
			Mode 2					Mode 3				

TABLE A.9/J.52

Code parameters for a sampling frequency of 44.1 kHz

Sampling frequency 44.1 kHz												
Bit rate (bit/s)	No. bit per frame	No. byte per frame	N	No. code words			r (%)	N	No. code words			r (%)
				L	L_N	L_{N-1}			L	L_N	L_{N-1}	
32 000	835.9183673	104.4897959	108	1	1	0	3.83	56	2	2	0	7.66
40 000	1 044.897959	130.6122449	134	1	1	0	3.06	48	3	1	2	9.19
48 000	1 253.877551	156.7346939	160	1	1	0	2.55	43	4	4	0	10.21
56 000	1 462.857143	182.8571429	186	1	1	0	2.19	50	4	2	2	8.75
64 000	1 671.836735	208.9796918	212	1	1	0	1.91	46	5	3	2	9.57
80 000	2 089.795918	261.2244898	135	2	1	1	3.06	48	6	3	3	9.19
96 000	2 507.755102	313.4693878	161	2	1	1	2.55	44	8	1	7	10.21
112 000	2 925.714286	365.7142857	187	2	1	1	2.19	45	9	5	4	9.84
128 000	3 343.673469	417.9591836	213	2	1	1	1.91	46	10	7	3	9.57
160 000	4 179.591837	522.4489796	178	3	1	0	2.30	45	13	2	11	9.95
192 000	5 015.510204	626.9387755	161	4	2	2	2.55	46	15	11	4	9.57
224 000	5 851.428571	731.4285714	187	4	3	1	2.19	45	18	11	7	9.84
256 000	6 687.346939	835.9183673	171	5	5	0	2.39	44	21	16	5	10.05
320 000	8 359.183673	1 044.897959	178	6	6	0	2.30	45	26	4	22	9.95
384 000	10 031.020410	1 253.877551	161	8	5	3	2.55	45	31	13	18	9.89
			Mode 2					Mode 3				

TABLE A.10/J.52

Code parameters for a sampling frequency of 32 kHz

Sampling frequency 32 kHz													
Bit rate (bit/s)	No. bit per frame	No. byte per frame	N	No. code words			r [%]	N	No. code words			r [%]
				L	L_N	L_{N-1}			L	L_N	L_{N-1}	
32 000	1 152	144	148	1	1	0	2.78	52	3	3	0	8.33
40 000	1 440	180	184	1	1	0	2.22	49	4	4	0	8.89
48 000	1 728	216	220	1	1	0	1.85	48	5	1	4	9.26
56 000	2 016	252	130	2	2	0	3.17	46	6	6	0	9.52
64 000	2 304	288	148	2	2	0	2.78	46	7	1	6	9.72
80 000	2 880	360	184	2	2	0	2.22	44	9	9	0	10.00
96 000	3 456	432	148	3	3	0	2.78	44	11	3	8	10.19
112 000	4 032	504	172	3	3	0	2.38	46	12	12	0	9.52
128 000	4 608	576	196	3	3	0	2.08	46	14	2	12	9.72
160 000	5 760	720	184	4	4	0	2.22	44	18	18	0	10.00
192 000	6 912	864	177	5	4	1	2.31	46	21	3	18	9.72
224 000	8 064	1 008	172	6	6	0	2.38	45	25	8	17	9.92
256 000	9 216	1 152	169	7	4	3	2.43	44	29	21	8	10.07
320 000	11 520	1 440	164	9	9	0	2.50	44	36	36	0	10.00
384 000	13 824	1 728	162	11	1	10	2.55	45	43	8	35	9.95
			Mode 2					Mode 3				

The effect of this block interleaving scheme and the systematic encoding is that the order and the values of the transmitted information bytes are not affected by the interleaving process and the encoding process, respectively. The additional 4 L redundant bytes, derived in the RS-encoder, are transmitted at the end of each frame.

Examples for this interleaving scheme are given in Figure A.8.

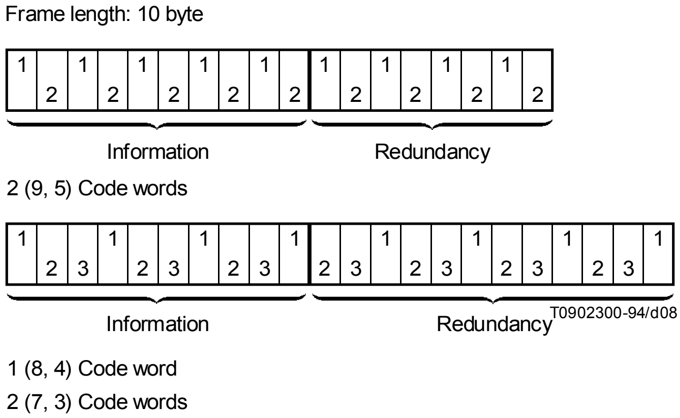

FIGURE A.8/J.52

Interleaving scheme

A.4.3 Position of redundancy

In all cases, redundancy for error correction is inserted just before the header and corresponds to the following frame.

Appendix I

Extension of Recommendation H.221 to include ISO/MPEG audio programme transmission over ISDN and 2 Mbit/s channels

(This appendix does not form an integral part of this Recommendation)

A new application of Recommendation H.221 is the exchange of audio mono/stereo broadcasting programmes between radio reporters/studios or studio/studios or studio/ transmitters using the ISDN or 2 Mbit/s channels.

The mono or stereo audio signals are coded according to ISO/IEC Standard 11172-3.

Two transmission formats are required:

– Transmission of *one* mono/stereo signal using 1 to 6 of 64 kbit/s channels or a single H_0 channel in a switched ISDN network.

– Transmission of *one or more* mono stereo signals using for example, H_0 or H_1 channels for permanent connections.

It is proposed:

– for the first application to add an escape table for MPEG audio; and

– to use the MBE (multiple byte extension) format for the second application.

Tables A.1/H.221 to A.7/H.221 and Figures 5h/H.221 to 5o/H.221 given in this appendix are extensions the information given in Recommendation H.221, 1993.

TABLE A.1/H.221

BAS numerical values

	(000) Audio command	(001) Transfer rate command	(010) Other commands	(011) LSD/MLP command	(100) Audio transfer rate capability	(101) Data/video video capability	(110)	(111) Escape
[0]	Neutral	64	Video-off	LSD off	Neutral	Var-LSD	(R)	Class (R)
[1]	(R)	2 × 64	H.261	300	A-law	300	(R)	Class (R)
[2]	(R)	3 × 64	Vid-imp (R)	1200	μ-law	1200	(R)	Class (R)
[3]	(R)	4 × 64	Video-ISO	4800	G.725-T1	4800	(R)	Class (R)
[4]	A-law, OU	5 × 64	AV-ISO	6400	G.725-T2	6400	(R)	Class (R)
[5]	μ-law, OU	6 × 64	(R)	8000	∅G.728	8000	(R)	Class (R)
[6]	G.722, m1	384	Encrip-on	9600	Au-ISO	9600	(R)	Class (R)
[7]	Au-off, U	2 × 384	Encrip-off	14 400	∅SM-comp	14 400	(R)	Class (R)
[8]	(Note 2)	3 × 384	(R)	16k	128	16k	(R)	Familly (R)
[9]	(Note 2)	4 × 384	(R)	24k	192	24k	(R)	Familly (R)
[10]	(R)	5 × 384	(R)	32k	256	32k	(R)	Familly (R)
[11]	(R)	1536	(R)	40k	(R)	40k	(R)	Familly (R)
[12]	(R)	1920	(R)	48k	512	48k	(R)	Familly (R)
[13]	Au-ISO-64	128	(R)	56k	768	56k	(R)	∅Agg-C1
[14]	Au-ISO-128	192	(R)	62.4k	(R)	62.4k	(R)	∅Agg-C2
[15]	Au-ISO-192	256	(R)	64k	1152	64k	(R)	∅Agg-C3
[16]	Au-ISO-156	∅320	Freeze-pic	MLP-off	1B	MLP-4k	(R)	∅HI-rates A2
[17]	Au-ISO-384	Loss i.c.	Fast-update	MLP-4k	2B	MLP-6.4k	(R)	H.230
[18]	A-law, OF	Chan. # 2	Au-loop	MLP-6.4k	3B	Var-MLP	(R)	Date-app. A3
[19]	μ-law, OF	Chan. # 3	Vid-loop	Var-MLP	4B	∅dummy	(R)	SBE numbers
[20]	(R)	Chan. # 4	Dig-loop	(R)	5B	QCIF	(R)	SBE characters
[21]	(R)	Chan. # 5	Loop-off	DTI-1 (R)	6B	CIF	(R)	SBE(R)
[22]	(R)	Chan. # 6	(R)	DTI-2 (R)	Restrict	1/29.97	(R)	SBE(R)
[23]	(R)	512	∅SM-comp	DTI-3 (R)	6B-H_0-comp	2/29.97	(R)	SBE(R)
[24]	G.722, m2	768	∅Not-SM-comp	(R)	H_0	3/29.97	(R)	Cap-mark
[25]	G.722, m3	(R)	6B-H_0-comp	(R)	$2H_0$	4/29.97	(R)	Start-MBE
[26]	Au-40k (R)	1152	Not-6B-H_0	(R)	$3H_0$	V-Imp(R)	(R)	(R)
[27]	Au-32k (R)	(R)	Restrict	(R)	$4H_0$	Video-ISO	(R)	(R)
[28]	Au-24k (R)	(R)	Derestrict	(R)	$5H_0$	AV-ISO	(R)	(R)
[29]	∅G.278	1472	(R)	(R)	1472	ESC-CF(R)	(R)	(R)
[30]	Au<16k (R)	(R)	(R)	(R)	H_{11}	Encryp.	(R)	ns-cap
[31]	Au-off, F	(R)	(R)	Var-LSD	H_{12}	MBE-cap.		na-comm

∅ Denotes proposed new values

NOTES

1 Shaded cells denote those values which are unallocated or could be released for other uses in due course.

2 These codes are listed in Recommendation G.725 with reference to an "application channel"; such a channel has not been defined, the concept having been superseded by that of LSD/MLP, therefore these codes should not be used.

TABLE A.2/H.221

Hi-rates numerical values

	(000) – ∅ More MLP commands	(001) – ∅ Au-ISO commands	(010)	(011) HSD/H-MLP commands	(100) – ∅ Au-ISO capabilities	(101) HSD/H-MLP capabilities	(110)	(111) Forbidden
[0]	MLP-14.4k	Au-off		HSD off				
[1]	MLP-22.4k	Au-32k		Var-HSD	Au-1B	Var-HSD		
[2]	MLP-30.4k	Au-40k		H-MLP-62.4	Au-2B	H-MLP-62.4		
[3]	MLP-38.4k	Au-48k		H-MLP-64k	Au-3B	H-MLP-84k		
[4]	MLP-46.4k	Au-56k		H-MLP-128k	Au-4B	H-MLP-128k		
[5]		Au-62.4k		H-MLP-192k	Au-5B	H-MLP-192k		
[6]		Au-64k		H-MLP-258k	Au-6B	H-MLP-258k		
[7]		Au-80k		H-MLP-320k		H-MLP-320k		
[8]	MLP-16k	Au-96k		H-MLP-384k		H-MLP-384k		
[9]	MLP-24k	Au-112k						
[10]	MLP-32k	Au-124.8k						
[11]	MLP-40k	Au-128k						
[12]		Au-160k						
[13]		Au-187.2k		Var-H-MLP		Var-H-MLP		
[14]		Au-192k		H-MLP-off				
[15]		Au-224k						
[16]		Au-249.6k						
[17]		Au-256k		HSD-64k		HSD-64k		
[18]		Au-288k		HSD-128k		HSD-128k		
[19]		Au-312k		HSD-192k	CorrMode 1	HSD-192k		
[20]		Au-320k		HSD-256k	CorrMode 2	HSD-256k		
[21]		Au-352k		HSD-320k	CorrMode 3	HSD-320k		
[22]		Au-374.4k		HSD-384k		HSD-384k		
[23]				HSD-512k		HSD-512k		
[24]				HSD-768k	AsyncMode	HSD-768k		
[25]		Error off		HSD-1152k	Au-Layer-I	HSD-1152k		
[26]		Error 1		HSD-1536k	Au-Layer-II	HSD-1538k		
[27]		Error 2			Au-Layer-III			
[28]		Error 3			Sample-32k			
[29]					Sample-44.1k			
[30]					Sample-48k			
[31]								

NOTE – Escape table reached by BAS (111) [16].

Au-ISO commands (16) – For bit position illustrations see Figure A.2/H.221

Au-off	ISO: audio switched off (Audio off, F or U in Table A.1/H.221 should turn off all types of audio).
Au-32k	1B-mode: audio data at 32 kbit/s in initial channel in bit 3 ... 6.
Au-40k	1B-mode: audio data at 40 kbit/s in initial channel in bit 3 ... 7.
Au-48k	1B-mode: audio data at 48 kbit/s in initial channel in bit 1 ... 6.
Au-56k	1B-mode: audio data at 56 kbit/s in initial channel in bit 1 ... 7.
Au-62.4k	1B-mode: audio data at 62.4 kbit/s in initial channel in bit 1 ... 7 and in octets 17 ... 80 of SC (service channel).
Au-64k	1B-mode: audio data at 64 kbit/s in initial channel in bit 1 ... 8.
Au-80k	2B-mode: audio data at 80 kbit/s in initial channel in bit 5, 6, in octets 41 ... 56 of SC and in complete second channel excluding FAS and BAS.
Au-96k	2B-mode: audio data at 96 kbit/s in initial channel in bit 3 ... 6, in octets 41 ... 56 of SC and in complete second channel excluding FAS and BAS.
Au-112k	2B-mode: audio data at 112 kbit/s in initial channel in bit 1 ... 6, in octets 41 ... 56 of SC and in complete second channel excluding FAS and BAS.
Au-124.8k/126.4k[4]	2B-mode: audio data at 124.8 kbit/s in initial channel in bit 1 ... 7, in octets 17 ... 80 of SC and in complete second channel excluding FAS and BAS.
Au-128k	3B-mode: audio data at 128 kbit/s in initial channel in octets 41 ... 73 of SC and in complete second and third channel excluding FAS and BAS.
Au-160k	3B-mode: audio data at 160 kbit/s in initial channel in 3 ... 6, in octets 17 ... 80 of SC and in complete second and third channel excluding FAS and BAS.
Au-187.2k/190.4k[4]	3B-mode: audio data at 187.2 kbit/s in initial channel in bit 1 ... 7, in octets 17 ... 80 of SC and in complete second and third channel excluding FAS and BAS.
Au-192k	4B-mode: audio data at 192 kbit/s in initial channel, in octets 25 ... 73 of SC and in complete second, third and fourth channel excluding FAS and BAS.
Au-224k	4B-mode: audio data at 224 kbit/s in initial channel, in bits 3 ... 6, octets 25 ... 73 of SC and in complete second, third and fourth channel excluding FAS and BAS.
Au-249.6k/254.4k[4]	4B-mode: audio data at 249.6 kbit/s in initial channel in bit 1 ... 7, octets 17 ... 80 of SC and in complete second, third and fourth channel excluding FAS and BAS.
Au-256k	5B-mode: audio data at 256 kbit/s in initial channel, octets 17 ... 80 of SC and in complete second, third, fourth and fifth channel excluding FAS and BAS.
Au-288k	5B-mode: audio data at 288 kbit/s in initial channel in bit 3 ... 6, octets 17 ... 80 of SC and in complete second, third, fourth and fifth channel excluding FAS and BAS.

[4] The higher bit rate is reached during an H_0 call by issuing the command together with the <not-6B-H_0-comp> command. The bit positions will be all of the corresponding number of time slots, except FAS and BAS (in TS 1 only).

Au-312k/318.4k[5]	5B-mode: audio data at 312 kbit/s in initial channel in bit 1 ... 7, octets 17 ... 80 of SC and in complete second, third, fourth and fifth channel excluding FAS and BAS.
Au-320k	6B-mode: audio data at 320 kbit/s in initial channel in bit 5 and in complete second, third, fourth, fifth and sixth channel excluding FAS and BAS.
Au-352k	6B-mode: audio data at 352 kbit/s in initial channel in bit 1 ... 5 and in complete second, third, fourth, fifth and sixth channel excluding FAS and BAS.
Au-373.4k/382.4k[5]	6B-mode: audio data at 374.4 kbit/s in initial channel in bit 1 ... 7, octets 17 ... 80 of SC and in complete second, third, fourth, fifth and sixth channel excluding FAS and BAS.

Au-ISO capabilities (16)

Au-1B	Capability to operate in any of the MPEG audio modes listed in the corresponding command table, on a single B-channel[6].
Au-2B	Capability to operate in any of the MPEG audio modes listed in the corresponding command table, on one or two B-channels[6] (or TS 1).
Au-3B	Capability to operate in any of the MPEG audio modes listed in the corresponding command table, on one, two or three B-channels[6].
Au-4B	Capability to operate in any of the MPEG audio modes listed in the corresponding command table, on one to four B-channels[6].
Au-5B	Capability to operate in any of the MPEG audio modes listed in the corresponding command table, on one to five B-channels[6].
Au-6B	Capability to operate in any of the MPEG audio modes listed in the corresponding command table, on one to six B-channels[6].
Asynch.mode	Can decode audio data sampled asynchronous to the network clock.
Au-Layer I	Capable of decoding audio to ISO/IEC 11172-3 Layer I.
Au-Layer II	Capable of decoding audio to ISO/IEC 11172-3 Layer II.
Au-Layer III	Capable of decoding audio to ISO/IEC 11172-3 Layer III.
Sample 32k	Can decode audio sampled with 32 kHz clock frequency.
Sample 44.1k	Can decode audio sampled with 44.1 kHz clock frequency.
Sample 48k	Can decode audio sampled with 48 kHz clock frequency.
Correction Mode 1, 2 and 3	Can decode error-correction data of the ancillary data field of the ISO/IEC 11172-3 signal.

---

[5] The higher bit rate is reached during an H_0 call by issuing the command together with the <not-6B-H_0-comp> command. The bit positions will be all of the corresponding number of time slots, except FAS and BAS (in TS 1 only).

[6] Or the corresponding number of an H_0 channel, from TS 1 upwards.

Initial channel

	1	2	3	4	5	6	7	8
1			1	2	3	4		
2			5	6	7	8		
3			9	10	11	12		FAS
4			13	14	15	16		
5			17	18	19	20		
6			21	22	23	24		
7			25	26	27	28		
8			29	30	31	32		
9			33	34	35	36		
10			37	38	39	40		
11			41	42	43	44		BAS
12			45	46	47	48		
13			49	50	51	52		
14			53	54	55	56		
15			57	58	59	60		
16			61	62	63	64		
17			65	66	67	68		
18			69	70	71	72		
•			•	•	•	•		
•			•	•	•	•		
79			#	#	#	316		
80		3	#	#	319	320		

FIGURE (5h)/H.221

32 kbit/s ISO MPEG audio in 1B
(Extension to Recommendation H.221)

Initial channel

	1	2	3	4	5	6	7	8
1			1	2	3	4	5	
2			6	7	8	9	10	
3			11	12	13	14	15	
4			16	17	18	19	20	FAS
5			21	22	23	24	25	
6			26	27	28	29	30	
7			31	32	33	34	35	
8			36	37	38	39	40	
9			41	42	43	44	45	
10			46	47	48	49	50	
11			51	52	53	54	55	
12			56	57	58	59	60	BAS
13			61	62	63	64	65	
14			66	67	68	69	70	
15			71	72	73	74	75	
16			76	77	78	79	80	
17			81	82	83	84	85	
18			86	87	88	89	90	
•			•	•	•	•	•	
•			•	•	•	•	•	
79			#	#	#	#	395	
80			#	#	#	399	400	

FIGURE (5i)/H.221

40 kbit/s ISO MPEG audio in 1B
(Extension to Recommendation H.221)

Initial channel

	1	2	3	4	5	6	7	8
1	1	2	3	4	5	6		
2	7	8	9	10	11	12		
3	13	14	15	16	17	18		
4	19	20	21	22	23	24		FAS
5	25	26	27	28	29	30		
6	31	32	33	34	35	36		
7	37	38	39	40	41	42		
8	43	44	45	46	47	48		
9	49	50	51	52	53	54		
10	55	56	57	58	59	60		
11	61	62	63	64	65	66		
12	67	68	69	70	71	72		BAS
13	73	74	75	76	77	78		
14	79	80	81	82	83	84		
15	85	86	87	88	89	90		
16	91	92	93	94	95	96		
17	97	98	99	100	101	102		
18	103	104	105	106	107	108		
•	•	•	•	•	•	•		
•	•	•	•	•	•	•		
79	#	#	#	#	#	474		
80	#	#	#	#	479	480		

FIGURE (5j)/H.221

48 kbit/s ISO MPEG audio in 1B
(Extension to Recommendation H.221)

Initial channel

	1	2	3	4	5	6	7	8
1	1	2	3	4	5	6	7	
2	8	9	10	11	12	13	14	
3	15	16	17	18	19	20	21	
4	22	23	24	25	26	27	28	FAS
5	29	30	31	32	33	34	35	
6	36	37	38	39	40	41	42	
7	43	44	45	46	47	48	49	
8	50	51	52	53	54	55	56	
9	57	58	59	60	61	62	63	
10	64	65	66	67	68	69	70	
11	71	72	73	74	75	76	77	
12	78	79	80	81	82	83	84	BAS
13	85	86	87	88	89	90	91	
14	92	93	94	95	96	97	98	
15	99	100	101	102	103	104	105	
16	106	107	108	109	110	111	112	
17	113	114	115	116	117	118	119	
18	120	121	122	123	124	125	126	
•	•	•	•	•	•	•	•	
•	•	•	•	•	•	•	•	
79	#	#	#	#	#	#	553	
80	#	#	#	#	#	559	560	

FIGURE (5k)/H.221

56 kbit/s ISO MPEG audio in 1B
(Extension to Recommendation H.221)

Initial channel

	1	2	3	4	5	6	7	8
1	1	2	3	4	5	6	7	
2	8	9	10	11	12	13	14	
3	15	16	17	18	19	20	21	
4	22	23	24	25	26	27	28	FAS
5	29	30	31	32	33	34	35	
6	36	37	38	39	40	41	42	
7	43	44	45	46	47	48	49	
8	50	51	52	53	54	55	56	
9	57	58	59	60	61	62	63	
10	64	65	66	67	68	69	70	
11	71	72	73	74	75	76	77	
12	78	79	80	81	82	83	84	BAS
13	85	86	87	88	89	90	91	
14	92	93	94	95	96	97	98	
15	99	100	101	102	103	104	105	
16	106	107	108	109	110	111	112	
17	113	114	115	116	117	118	119	120
18	121	122	123	124	125	126	127	128
•	•	•	•	•	•	•	•	•
•	•	•	•	•	•	•	•	•
79	#	#	#	#	#	#	#	616
80	#	#	#	#	#	#	623	624

FIGURE (5l)/H.221

62.4 kbit/s ISO MPEG audio in 1B
(Extension to Recommendation H.221)

Initial channel

	1	2	3	4	5	6	7	8
1	1	2	3	4	5	6	7	8
2	9	10	11	12	13	14	15	16
3	17	18	19	20	21	22	23	24
4	25	26	27	28	29	30	31	32
5	33	34	35	36	37	38	39	40
6	41	42	43	44	45	46	47	48
7	49	50	51	52	53	54	55	56
8	57	58	59	60	61	62	63	64
9	65	66	67	68	69	70	71	72
10	73	74	75	76	77	78	79	80
11	81	82	83	84	85	86	87	88
12	89	90	91	92	93	94	95	96
13	97	98	99	100	101	102	103	104
14	105	106	107	108	109	110	111	112
15	113	114	115	116	117	118	119	120
16	121	122	123	124	125	126	127	128
17	129	130	131	132	133	134	135	136
18	137	138	139	140	141	142	143	144
•	•	•	•	•	•	•	•	•
•	•	•	•	•	•	•	•	•
79	#	#	#	#	#	#	#	632
80	#	#	#	#	#	#	639	640

FIGURE (5m)/H.221

64 kbit/s ISO MPEG audio in 1B
(Extension to Recommendation H.221)

	Initial channel								Second channel							
	1	2	3	4	5	6	7	8	1	2	3	4	5	6	7	8
1					1	2			3	4	5	6	7	8	9	
2					10	11			12	13	14	15	16	17	18	
3					19	20			21	22	23	24	25	26	27	
4					28	29		FAS	30	31	32	33	34	35	36	FAS
5					37	38			39	40	41	42	43	44	45	
6					46	47			48	49	50	51	52	53	54	
7					55	56			57	58	59	60	61	62	63	
8					64	65			66	67	68	69	70	71	72	
9					73	74			75	76	77	78	79	80	81	
10					82	83			84	85	86	87	88	89	90	
11					91	92			93	94	95	96	97	98	99	
12					100	101		BAS	102	103	104	105	106	107	108	BAS
13					109	110			111	112	113	114	115	116	117	
14					118	119			120	121	122	123	124	125	126	
15					127	128			129	130	131	132	133	134	135	
16					136	137			138	139	140	141	142	143	144	
17					145	146			147	148	149	150	151	152	153	154
18					155	156			157	158	159	160	161	162	163	164
•					•	•			•	•	•	•	•	•	•	•
41					385	386		387	388	389	390	391	392	393	394	395
•					•	•		•	•	•	•	•	•	•	•	•
56					550	551		552	553	554	555	556	557	558	559	560
•					•	•			•	•	•	•	•	•	•	•
•					•	•			•	•	•	•	•	•	•	•
79					#	#			#	#	#	#	#	#	#	790
80					#	#			#	#	#	#	#	#	799	800

FIGURE (5n)/H221

80 kbit/s ISO MPEG audio in 2B
(Extension to Recommendation H.221)

	Initial channel								Second channel							
	1	2	3	4	5	6	7	8	1	2	3	4	5	6	7	8
1	1	2	3	4	5	6	7		8	9	10	11	12	13	14	
2	15	16	17	18	19	20	21		22	23	24	25	26	27	28	
3	29	30	31	32	33	34	35		36	37	38	39	40	41	42	
4	43	44	45	46	47	48	49	FAS	50	51	52	53	54	55	56	FAS
5	57	58	59	60	61	62	63		64	65	66	67	68	69	70	
6	71	72	73	74	75	76	77		78	79	80	81	82	83	84	
7	85	86	87	88	89	90	91		92	93	94	95	96	97	98	
8	99	100	101	102	103	104	105		106	107	108	109	110	111	112	
9	113	114	115	116	117	118	119		120	121	122	123	124	125	126	
10	127	128	129	130	131	132	133		134	135	136	137	138	139	140	
11	141	142	143	144	145	146	147		148	149	150	151	152	153	154	
12	155	156	157	158	159	160	161	BAS	162	163	164	165	166	167	168	BAS
13	169	170	171	172	173	174	175		176	177	178	179	180	181	182	
14	183	184	185	186	187	188	189		190	191	192	193	194	195	196	
15	197	198	199	200	201	202	203		204	205	206	207	208	209	210	
16	211	212	213	214	215	216	217		218	219	220	221	222	223	224	
17	225	226	227	228	229	230	231	232	233	234	235	236	237	238	239	240
18	241	242	243	244	245	246	247	248	249	250	251	252	253	254	255	256
•	•	•	•	•	•	•	•	•	•	•	•	•	•	•	•	•
•	•	•	•	•	•	•	•	•	•	•	•	•	•	•	•	•
79	#	#	#	#	#	#	#	#	#	#	#	#	#	#	#	1232
80	#	#	#	#	#	#	#	#	#	#	#	#	#	#	#	1248

FIGURE (5o)/H221

124.8 kbit/s ISO MPEG audio in 2B
(Extension to Recommendation H.221)

TABLE A.4/H.221

Start-MBE numerical values
(Extension to Recommendation H.221)

(000)	(001)	(010)	(011)	(100)	(101)	(111)
Au-COM (Note 3)				Au-MAP (Note 3)		

NOTES

1 Escape table reached by BAS (111) [25].

2 The column header gives the attribute designation as bits (b_0, b_1, b_2), the left-hand column gives the decimal value of bits (b_3 b_4, b_5, b_6, b_7).

3 The actual values to be assigned by ITU-T Study Group 15.

TABLE A.5/H.221

Start-MBE numerical values
(Extension to Recommendation H.221)

Capability exchange	
Task No.	Task
1	START- MBE
2	N = number of bytes = number of audio codecs + 1
3	Au-MAP = (Note 2)
4	Capability codec 1 (Note 3)
.	
.	
.	
N + 2	Capability last codec (Note 3)

NOTES

1 Escape table reached by BAS (111)/25/.

2 The actual values to be assigned by ITU-T Study Group 15.

3

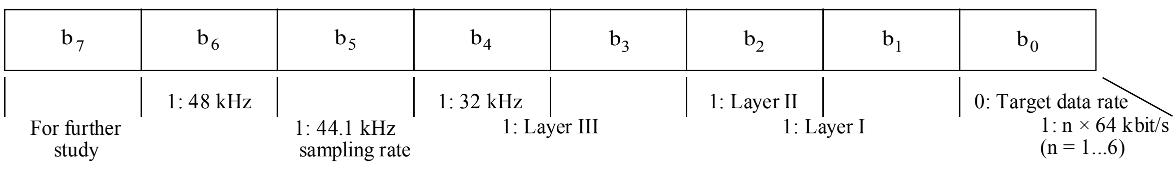

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TABLE A.6/H.221

Start-MBE numerical values
(Extension to Recommendation H.221)

Command exchange	
Task No.	Task
1	START- MBE
2	N = number of bytes = number of audio codecs + 1
3	Au-COM = (Note 2)
4	Command codec 1 (Note 3)
.	
.	
.	
N + 2	Command last codec (Note 3)

NOTES

1 Escape table reached by BAS (111)/25/.

2 The actual values to be assigned by ITU-T Study Group 15.

3

b_7	b_6	b_5	b_4	b_3	b_2	b_1	b_0
Number of Time Slots for one audio channel				Number of the first occupied Time Slot for an audio channel without Time Slot 1			

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Appendix II

Performance of the forward error correction

Within each code word, up to two (t < 2) arbitrarily distributed byte errors can be corrected. The burst-error-correction capability b within each frame depends on the number L of code words and the amount t of error correction:

$$b \le [(Lt - 1)\ 8 + 1] \text{ bits} \qquad \text{for } t > 0$$

In Figure II.1 the bit-error probability P_b after decoding is shown in dependence on the number t of corrected symbol errors, the code length N, and the bit-error probability BER of the transmission channel, where statistically independent bit errors are assumed. For burst errors with the same BER within a frame, the performance is much better than shown in Figure II.1.

The probability P_f of a falsely decoded word is shown in Figure II.2. The probability P_F of a falsely decoded frame (containing L code words) is given by:

$$P_F = 1 - \left(1 - P_f\right)^L$$

It must be mentioned that even for t = 2, a high amount of uncorrectable error patterns can be detected by the RS-decoder. The reliability of error detection can further be increased by reducing t.

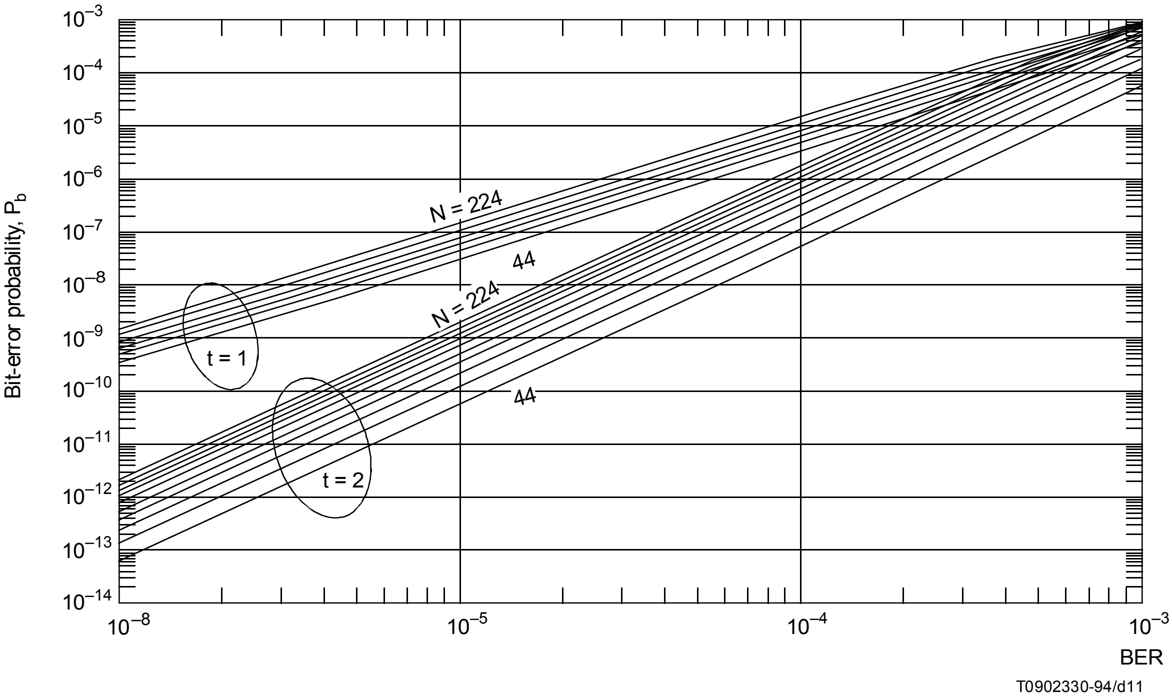

FIGURE II.1/J.52

Bit-error probability P_b after decoding, N = 44, 64, ..., 224

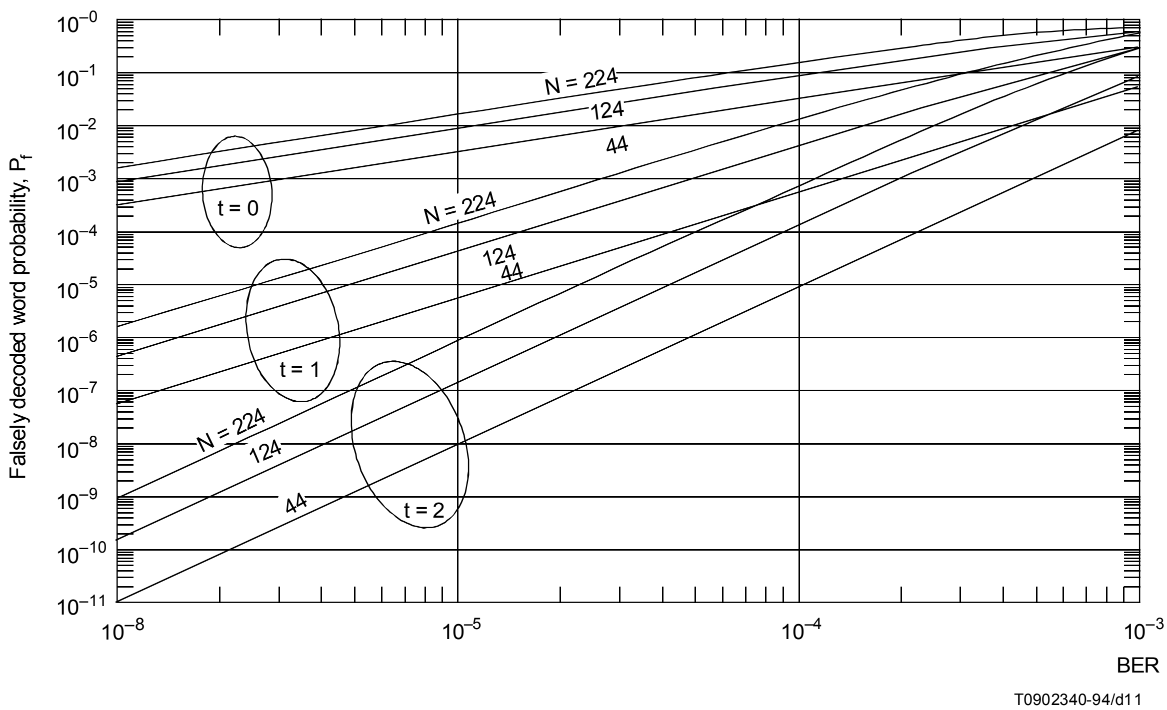

FIGURE II.2/J.52

Probability P_f of a falsely decoded code word

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