



INTERNATIONAL TELECOMMUNICATION UNION

ITU-T

TELECOMMUNICATION
STANDARDIZATION SECTOR
OF ITU

J.52

(07/96)

SERIES J: TRANSMISSION OF SOUND-PROGRAMME
AND TELEVISION SIGNALS

Digital transmission of sound-programme signals

**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")

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TRANSMISSION OF SOUND-PROGRAMME AND TELEVISION SIGNALS

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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 revised by ITU-T Study Group 9 (1993-1996) and was approved under the WTSC Resolution No. 1 procedure on the 11th of July 1996.

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

The system described in this Recommendation makes possible the transmission of sound signals, coded by new methods recommended by ITU-R Study Group 10 and standardized by ISO/IEC 11172-3, on 1 to 6 B-channels of the N-ISDN or on permanent connections with a 2048 kbit/s (or 1544 kbit/s) frame. In the case of transmission via the N-ISDN, Recommendation H.221 is used to aggregate up to six B-channels while maintaining the bit sequence integrity. The revised text contains clarifications that reflect experience gained in implementing the current version of this Recommendation, and a new chapter on transmission of program-associated data in the ancillary data field of the ISO frame.

INTRODUCTION

The ISO and IEC have approved the international standard 11172-3 for the reduction of the bit rate of high-quality digital sound-programme signals. With this system, it is possible to transmit high-quality mono sound-programme signals with bit rates in the range of 64 to 192 kbit/s. Stereo signals are encoded in single bit streams with bit rates in the range 128 to 384 kbit/s. 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 ITU-T has approved some Recommendations which describe the structure and function of 64 kbit/s channels for audiovisual applications (Recommendation H.200 contains a list of such Recommendations).

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 etc.), measures are necessary to maintain the bit sequence. The equipment described in the following enables 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.

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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)

(revised in 1996)

0 Scope

The ITU-R Recommendation BS.1115 recommends, based on extensive tests carried out by ITU-R Study Group 10 for different applications of the transmission in a broadcast chain (contribution, distribution, emission, commentary), the bit rate reduction system according to ISO/IEC 11172-3. The encoding layers and bit rates (per mono channel) recommended by ITU-R SG 10 depend on the application and are in the range of 120 - 180 kbit/s using Layer II, except for commentary where 60 kbit/s using Layer III may be used. For details, see ITU-R BS.1115.

For transmission of high-quality sound programmes and associated data it is desirable to use standardized 64 kbit/s channels or multiples thereof. This Recommendation defines formatter and reformatter for the transport of the bit rate reduced audio signals according to ISO/IEC 11172-3 and associated data on these standardized 64 kbit/s channels. The bit stream to be transmitted on 64 kbit/s channels could be protected optionally by means of error-correction measures.

The ITU-R has not (before the end of 1995) made any recommendation for the use of the ISO/IEC 13818-3 *lower sampling frequencies*. **If and only if** the extension to *lower sampling frequencies* is to be implemented, the provisions in Annex A shall apply. The implementation of the *lower sampling frequency* modes is in no way mandatory as they inherently limit the bandwidth, and would therefore not permit the high-quality criteria to be met.

1 General features

1.1 Purpose of the equipment

The purpose of the equipment is to process a bit rate reduced high-quality sound-programme signal for 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 1).

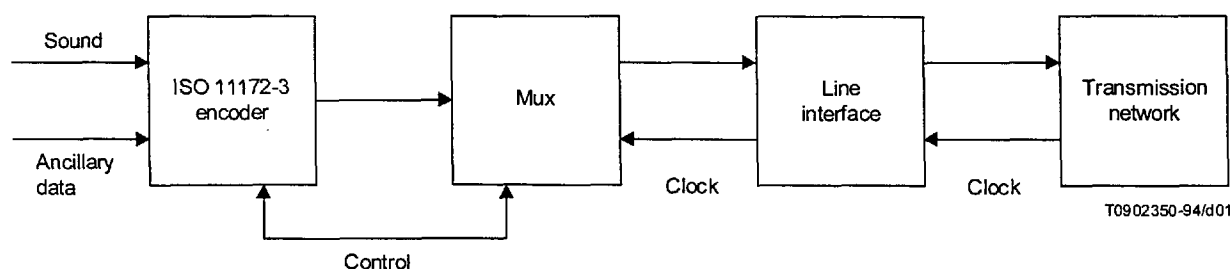


FIGURE 1/J.52

Connection between encoder, multiplexer and line interface

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 Layers II and III with the block length 24 ms (in case of 48 kHz sampling frequency) is shown in Figure 2.

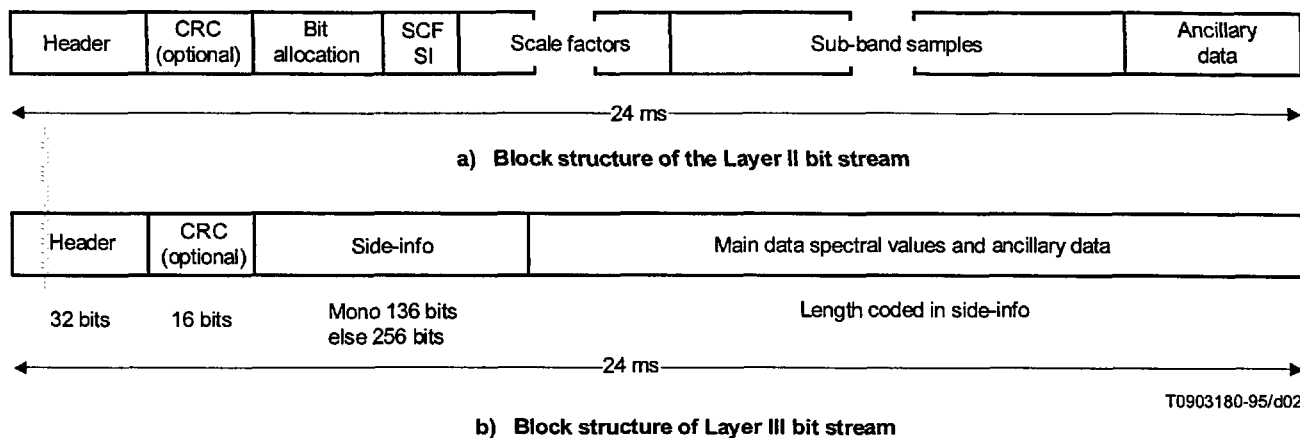


FIGURE 2/J.52

If possible, this bit stream has to be transmitted unchanged. 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 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 bits (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.

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 one to six 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⁾

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.

TABLE 1/J.52

Possible frame lengths in bytes (without padding)

Sampling Frequency (Hz)	32 000			44 100			48 000		
Layer	I	II	III	I	II	III	I	II	III
Bit rate (bit/s)									
32 000	48	144	144	32	104	104	32	96	96
40 000	–	–	180	–	–	130	–	–	120
48 000	–	216	216	–	156	156	–	144	144
56 000	–	252	252	–	182	182	–	168	168
64 000	96	288	288	68	208	208	64	192	192
80 000	–	360	360	–	261	261	–	240	240
96 000	144	432	432	104	313	313	96	288	288
112 000	–	504	504	–	365	365	–	336	336
128 000	192	576	576	136	417	417	128	384	384
160 000	240	720	720	172	522	522	160	480	480
192 000	288	864	864	208	626	626	192	576	576
224 000	336	1008	1008	240	731	731	224	672	672
256 000	384	1152	1152	276	835	835	256	768	768
288 000	432	–	–	312	–	–	288	–	–
320 000	480	1440	1440	348	1044	1044	320	960	960
352 000	528	–	–	380	–	–	352	–	–
384 000	576	1728	–	416	1253	–	384	1152	–
416 000	624	–	–	452	–	–	416	–	–
448 000	672	–	–	484	–	–	448	–	–

NOTE – A dash “–” in the table means that the combination of sampling frequency, bit rate and layer is not specified in ISO/IEC 11172-3.

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 H₀-channel structures:
 - 1544 kbit/s primary rate: 4 H₀;
3 H₀ + D.
 - 2048 kbit/s primary rate: 5 H₀ + D.

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.

2.2 Permanent connections

2048 kbit/s or 1544 kbit/s signals are used. Framing is according to 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.

3 Synchronization and frame alignment

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 H.242, should be used. With the use of the H.221 framing there is a possibility to achieve both clock synchronization and time synchronization between up to 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 equipment.

The procedure for **initialization** of the communication at the start of a call shall be according to Recommendation H.242, except that, if there is no requirement for intercommunication with 3 kHz speech terminals, it is not necessary to transmit G.711 audio during the period until Sequence A is completed. The capabilities to be transmitted during Sequence A include appropriate values from Table A.2/H.221, and a single command from that table introduces a mode switch to the required sound transmission mode.

Particular attention is drawn to clause 13/H.242 for the cases of transmission within a restricted network (multiple 56 kbit/s channels) and of interworking between unrestricted and restricted networks. Terminals on both types of network shall transmit within their capability sets the relevant values from attribute (110) of Table A.1/H.221.

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 Rate 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 been extended.

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 3 describes the different sections of the transmission system. The network provides data channels each with 64 kbit/s. 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 Layers II and III. The audio frame length (see Table 1) is dependent on the bit rate index, the sampling frequency and the status of the padding bit, an information that 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”.

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

- using the ancillary data field (recommended for Layers I and II);
- dynamic bit rate switching (recommended for Layer III).

The use of free format is not recommended.

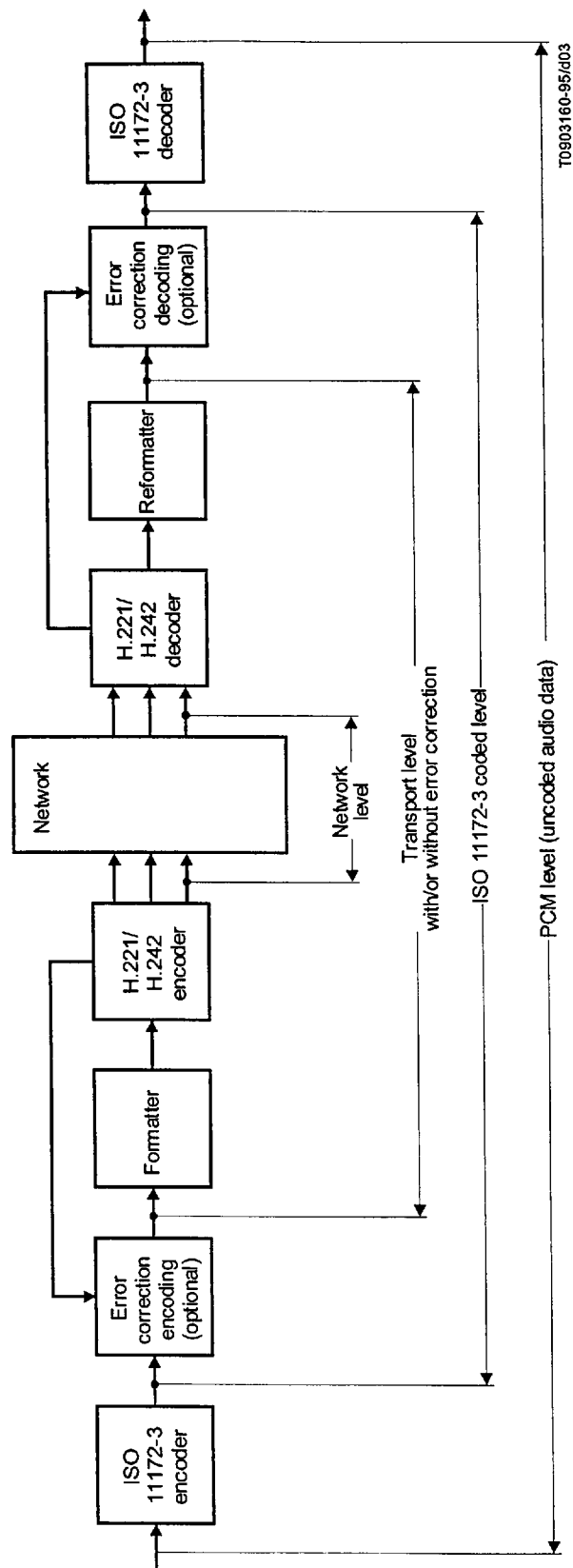


FIGURE 3/J.52
Transmission system with transport format and optional error control