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OPERATIONS AND QUALITY OF SERVICE
AUDIOVISUAL SERVICE

AUDIOGRAPHIC CONFERENCE TELESERVICE FOR ISDN

ITU-T Recommendation F.711

(Previously "CCITT Recommendation")

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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, established 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 F.711 was prepared by the ITU-T Study Group 1 (1993-1996) and was approved under the WTSC Resolution No. 1 procedure on the 31st of August 1993.

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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INTRODUCTION

The Audiographic Conference Teleservice is a means for teleconferencing similar to a telephone conference or a videoconference. This service offers a conference facility of better quality and with more possibilities than for a telephone conference but different than a videoconference due to the lack of moving images. Whereas for videoconferencing there are moving images but not necessarily graphics, in audiographic conferencing high quality graphics are necessary and take precedence over all other information including voice.

The general aspects of an audiographic conferencing service are described in Recommendation F.710. This Recommendation applies three general aspects for the purpose of defining an international digital audiographic conference teleservice on the ISDN. The service and facilities offered are described and reference made to the need for interoperability with other related audiovisual services.

The Audiographic Conference Teleservice can be offered either as a point-to-point or as a multipoint service. Both aspects are covered by this Recommendation and the functional performance of the Multipoint Control Unit (MCU) required for multipoint operation is described. The associated in-band signalling and conference procedures are specified in Recommendations T.120 through T.124.

This Recommendation specifies, as a minimum, the use of one 64 kbit/s channel (B-channel) of the ISDN basic access. The channel is framed and carries both high quality speech and meeting aid data such as facsimile, still picture and image annotation/telewriting. In principle, the second B-channel of the ISDN basic access can also be used for more efficiently transmitting meeting aid information.

NOTE – For audiographic conferences requiring higher bandwidths, data rates up to 1 920 kbit/s, as defined in the H-Series of Recommendations, can be used.

Recommendation F.711

AUDIOGRAPHIC CONFERENCE TELESERVICE FOR ISDN

(Geneva, 1993)

1 Definition

The **Audiographic Conference Teleservice** is an international service, offered by recognized operating agencies (ROA), enabling participants to conduct a real-time teleconference in which audio signals are exchanged together with non-voice graphics information except for motion video.

Unless otherwise noted, the terms and definitions relating to the audiographic conference service used in this Recommendation are as defined in Recommendation F.710.

1.1 Additional references

ITU-T Rec. G.711 – Pulse code modulation (PCM) of voice frequencies

ITU-T Rec. G.722 – 7 kHz audio-coding within 64 kbit/s

ITU-T Rec. G.728 – Coding of speech at 16 kbit/s using low delay code excited linear prediction

ITU-T Rec. H.221 – Frame structure for a 64 kbit/s to 1920 kbit/s channel in audiovisual teleservices

ITU-T Rec. H.230 – Frame-synchronous control and indication signals for audiovisual systems

ITU-T Rec. H.231 – Multipoint control units for audiovisual systems using digital channels up to 2 Mbit/s

ITU-T Rec. H.242 – System for establishing communication between audiovisual terminals using digital channels up to 2 Mbit/s

ITU-T Rec. H.243 – Procedures for establishing communication between three or more audiovisual terminals using digital channels up to 2 Mbit/s

ITU-T Rec. I.430 – Basic user-network interface - layer 1 specification

ITU-T Rec. I.431 – Primary rate user-network interface - layer 1 specification

ITU-T Rec. Q.921 – ISDN user-network interface – data link layer specification

ITU-T Rec. Q.922 – ISDN data link layer specification for frame mode bearer services

ITU-T Rec. Q.930 – ISDN user-network interface - layer 3 - general aspects

ITU-T Rec. Q.931 – ISDN user-network interface layer 3 specification for basic call control

ITU-T Rec. T.4 – Standardization of group 3 facsimile apparatus for document transmission

ITU-T Rec. T.6 – Facsimile coding schemes and coding control functions for group 4 facsimile apparatus

ITU-T Rec. T.30 – Procedures for document facsimile transmission in the general switched telephone network

ITU-T Rec. T.62 – Control procedures for teletex and group 4 facsimile services

ITU-T Rec. T.70 – Network-independent basic transport service for the telematic services

ITU-T Rec. T.90 – Characteristics and protocols for terminals for telematic services in ISDN

ITU-T Rec. T.120 – Introduction to the audiographics and audiovisual control recommendations

ITU-T Rec. T.122 – Multipoint communications service for audiographics and audiovisual conferencing

ITU-T Rec. T.123 – Protocol stacks for audiographic and audiovisual teleconference applications

ITU-T Rec. T.124 – Generic conference control for audiovisual services

2 Description

2.1 General description

The general description of the Audiographic Conference Service is given in Recommendation F.710.

2.2 Specific terminology

2.2.1 Basic facilities

The basic facilities consist of the primary means for audiovisual communication, namely an audio channel together with a transmission capability for imagery and additional control information. The service is bidirectional via ISDN and provides for the interconnection of two or more terminals.

NOTE – The terminals may not be equal because of different terminal capabilities. The interchange of information concerning these capabilities will take place at conference set-up through terminal negotiation.

2.2.1.1 Audio facilities

By the design of the room and/or the equipment, participants should be allowed to listen and speak simultaneously. Speech transmission is nominally of 7 kHz bandwidth, although this may be reduced to 3.1 kHz for short durations when additional bandwidth is required to achieve image transmission quality objectives, or when the conference has been opened to participants served by a non-ISDN network.

2.2.1.2 Still image facilities

As for videoconferencing, an image capture device can be used to capture pictorial objects such as charts/diagrams, documents, solid objects. For still images indefinite retention and display of the still image at the remote end is necessary.

2.2.1.3 Conference control facility

Conference control messages may be provided for governing call establishment and call terminations, controlling conference modes and audio/image signals; and for transmission of conference control signals, such as floor requests, tokens, etc.

2.2.1.4 Data transmission

For further study.

2.2.1.5 User-to-user messages

When provided by the network, the user-to-user signalling (UUS) supplementary service may be used for transmission of user-to-user messages over the D-channel.

2.2.2 Other facilities

A wide range of activities may be carried out using an audiographic teleconferencing system. This Recommendation therefore seeks to provide a range of options, leaving to the user the choice as to which should be implemented.

Subjectively, the quality of the audio and conference graphics should not be greatly affected by providing these meeting aids. Ergonomic considerations suggest that the number of display screens be reduced to a minimum.

NOTE – More detailed information on human factor aspects is essential, but can only be obtained in the relevant context during a considerable period of experience.

The following list of facility options illustrates the variety of choice which a potential user may consider.

2.2.2.1 Still image TV systems

As for videoconferencing, a display camera can be used to capture pictures of objects (charts/diagrams, documents, solid objects). The nature of a still picture TV codec is such that indefinite retention and display of the still picture at the remote end is possible.

2.2.2.2 Facsimile

Facsimile documents may be transmitted during the conference, using standardized facsimile equipment located either within or external to the audiographic conferencing unit.

2.2.2.3 Still picture of participants

In the case where a subscriber possesses a videoconference or videophone type terminal, this may be used for audiographic input transmitting still pictures only.

2.2.2.4 Additional facilities

For further study.

2.2.3 Audiographic conference terminal

The general features of an audiographic conference terminal are as specified in Recommendation F.710. Application of these general features for specific use in the ISDN is as illustrated in Figure 1.

2.2.4 Conference control

Conference control is provided within the data stream through which control and indication information is conveyed. This information may be used for the following purposes:

a) At the start of an audiographic teleconference to send initialization information concerning configuration of the terminals and terminal conferencing capabilities between connected terminals or between terminals and the MCU(s).

b) During the audiographic teleconference to communicate requests and acknowledgements between terminals for the use of meeting aids.

c) In addition to the active teleconference, messages may be sent selectively between participants using a keyboard, or other input device, and the output display.

2.2.5 Multipoint Control Unit (MCU)

The MCU is a piece of equipment which can be connected via access ports to an audiographic conference terminal or to another MCU. The purpose of the MCU is to permit the transmission of audio signals and supplementary information between a number of separated audiographic conference terminals.

2.2.6 Reservation centre

Provision should be made for a reservation centre which will be available to all users.

3 Procedures

3.1 Normal procedures

3.1.1 Phases in the conference process

The following phases apply to the overall process of an audiographic conference and are described in further detail in Recommendation F.710.

- Call set-up;
- Conference set-up;
- Conference session;
- Recovery and reconfiguration;
- Call release (disconnection).

Formal reservation procedures are normally required as a pre-condition for establishing an audiographic conference.

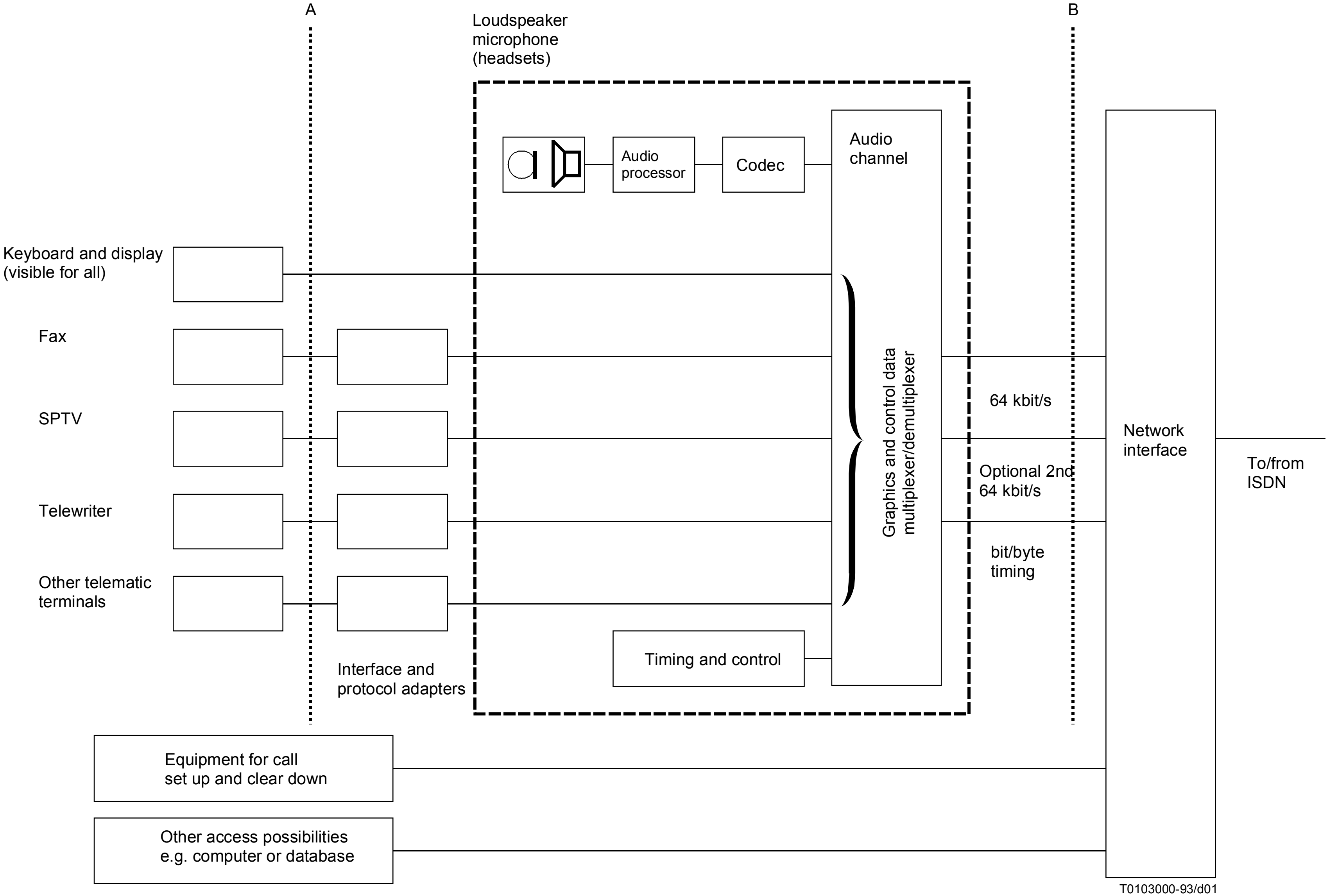

FIGURE 1/F.711

Audiographic conference terminal

3.2 Exceptional procedures

3.2.1 Mode switching

The ITU-T has defined various combinations of bit rate for wideband speech and data path capability within a 64 kbit/s channel. These combinations of bit rate are known as modes and are defined in the following manner:

Mode 0 – 64 kbit/s PCM speech (Recommendation G.711);

Mode 1 – 64 kbit/s wideband speech (Recommendation G.722);

Mode 2 – 56 kbit/s wideband speech plus 6.4 kbit/s data capability;

Mode 3 – 48 kbit/s wideband speech plus 14.4 kbit/s data capability (6.4 + 8 kbit/s).

An unpartioned 64 kbit/s ISDN-channel corresponds either to Mode 1 or to a PCM speech channel. Therefore, to provide general compatibility between an audiographic conference and terminals using an unpartioned channel, the audiographic conference terminal can work in any of the following three configurations:

a) Mode 2 or Mode 3 with frame alignment and data capability;

b) Mode 1 (Recommendation G.722 speech) with no data capability;

c) Mode 0 (Recommendation G.711 speech) with no data capability.

At the beginning of a session it is necessary that all terminals assume the same initial mode in order to enable different types of terminals to be interconnected. For the Audiographic Conference Teleservice the default mode is Mode 2.

It is possible that the subsequent call set-up be adjusted manually. However, it is preferred that this be performed automatically according to the method specified in Recommendation H.242. Mode switching during the conference is signalled in the BAS word in accordance with Recommendation H.221.

3.2.2 MCU procedures

For multipoint communications all terminals must be connected to the MCUs. For point-to-point communications it is not necessary for the terminals to be connected to MCUs.

Although particular attention must be paid to network topology in the case of satellite transmission, the basic functions of the MCU for both a terrestrial and a satellite network are similar.

The MCU in the Audiographic Conference Teleservice provides the following:

a) user access and network interface to the Audiographic Conference;

b) management of framing structure: multiplexing and demultiplexing;

c) mixing or switching of audio signals;

d) processing of the sub-channels;

e) analysis of control messages;

f) routing of signals to audiographic terminals or other MCUs;

g) handling of encrypted signals;

h) terminal interconnection.

All terminals in a multipoint conference shall be interconnected by a network of one or more MCUs. The MCUs may be cascaded when required by a particular conference configuration.

3.3 Alternative procedures

3.3.1 Reservation facility

Provision should be made for a centralized and/or decentralized facility for audiographic conference resources. A reservation facility may be part of the MCU or located separately.

Resources that should be considered for the reservation facility include:

– terminal equipment;

– communication paths;

– bandwidth.

3.3.2 Allocation of various information signals

For the basic service, the information signals are multiplexed within a single 64 kbit/s stream. For special cases (e.g. fast image transfer, high speed facsimile, etc.) a second 64 kbit/s channel can be used. It is possible to use the second 64 kbit/s channel either continuously, temporarily on a demand basis, or during the conference when an optional feature is desired. The D-channel may be used for user-to-user messages (see 2.2.1.5).

4 Network capabilities for charging

This Recommendation does not cover charging principles. Future Recommendations are expected to address this subject.

5 Interworking requirements

The possibilities for interworking/intercommunications with other services are in Table 1.

TABLE 1/F.711

Interworking/intercommunication requirements

Interworking/intercommunication of an Audiographic Conference Terminal with other audiographic terminals	Level of interworking/intercommunication
7 kHz ISDN telephone terminals	Wideband speech G.722, see Note
3.1 kHz ISDN telephone terminals	A-law/μ-law speech G.711, see Note
3.1 kHz PSTN telephone terminals	A-law/μ-law speech G.711, see Note
Videotelephone terminals	A-law/μ-law speech or wideband speech if featured by the VP. See Note
Videoconference terminals	Wideband speech
Other terminals	For further study
NOTE – Both A-law and μ-law must be supported.	

Point-to-point Audiographic Conferences between a terminal on an ISDN network and a terminal on a non-ISDN network must be interconnected through a network interface. An example of a simple point-to-point connection between different networks is shown in Figure 2.

6 Applicability of supplementary services to the AGCS

In principle, supplementary services provided in the switched service are the same as for telephony.

7 Attributes/values

NOTE – This Attribute list only deals with Attributes for the AGCS using one B-channel.

It is foreseen that the future AGCS may use two or more 64 kbit/s channels. Attributes for additional channels need further study.

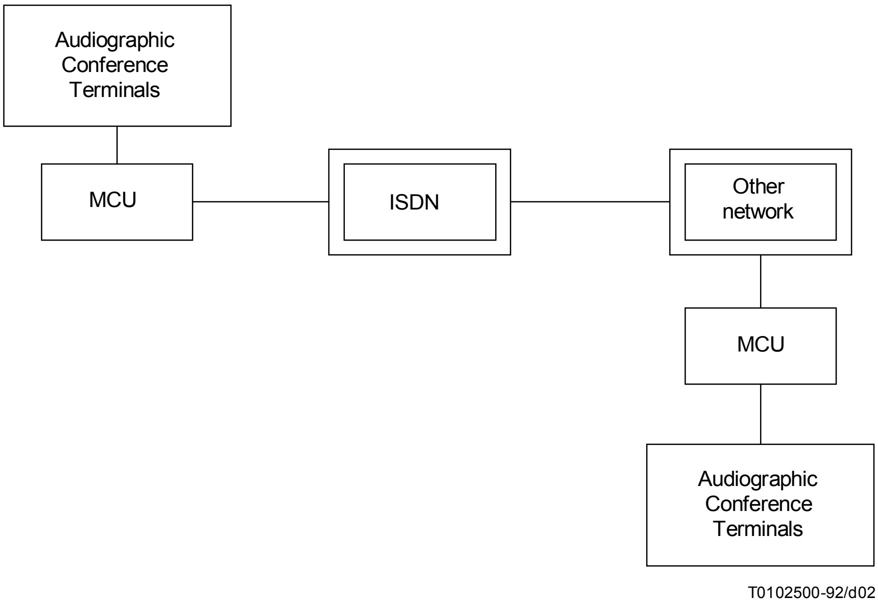

FIGURE 2/F.711

AGC interworking/intercommunication between the ISDN and other networks

7.1 Low Layer Attributes

Information transfer attributes

1)	Information transfer mode	Circuit
2)	Information transfer rate	64 kbit/s
3)	Information transfer capability	Unrestricted digital information with tone/announcements (UDI-TA)

NOTE – As an interim solution before circuit mode 64 kbit/s 8 kHz structured multi-use bearer service is available, the AGCS should use "unrestricted digital information" (UDI) as the transfer capability when calling other AGCS terminals.

4)	Structure	8 kHz structure
5)	Establishment of communication	On demand, by reservation
6)	Communication configuration	Point-to-point, multipoint
7)	Symmetry	Bidirectional symmetric

Access attributes

8)	Access type basis	ISDN
9)	Access channel and rate	64 kbit/s or 2 × 64 kbit/s, B (64 kbit/s) for user information and D (16 kbit/s) for signalling and user-to-user communications
10)	Info access structure	Multimedia (1 × 64 kbit/s)
11)	Signalling access protocol	
11.1)	Layer 1	I.430/I.431
11.2)	Layer 2	Q.922, Q.921
11.3)	Layer 3	Q.931, I.233, I.122, I.370
12)	Information access protocol	
12.1)	Layer 1	I.430, H.221
12.2)	Layer 2	- - -
12.3)	Layer 3	- - -

7.2 High layer attributes (Note)

13)	Type of user information	Audio, SPTV, X-Y devices, facsimile, telewriter, telematic data, user-to-user message, C & I
14)	Transport attribute (Layer 4)	None
15)	Session attribute (Layer 5)	T.122, H.230, H.231, H.242 and H.243
16)	Presentation attributes (Layer 6)	
16.1)	Audio	G.722, G.728, G.711 (for compatibility with telephony)
16.2)	Video	SPTV
16.3)	Auxiliary	T.30, T.4, T.6, T.62, T.70 and T.90 (for facsimile)
16.4)	Dialogue	Message channel
17)	Application attributes (Layer 7)	
17.1)	Audio	Microphone and loudspeaker, echo control device, headset
17.2)	Video	Object/document camera, person camera (for still pictures), monitor(s)
17.3)	Auxiliary	Group 3 and Group 4 facsimile
17.4)	Dialogue	Dedicated keyboard and display

NOTE – The mandatory status of the equipments for the basic service and the optional status of the auxiliary equipments are for further study.

7.3 General attributes

18)	ISDN supplementary services provided in circuit switched mode	Same as for telephony
19)	Quality of service	
19.1)	Transmission performance	G.821
19.2)	Routing	The choice of the routing, including routing via satellite, should ideally not influence the quality of service
19.3)	Confidentiality	Must be guaranteed
19.4)	Audio quality	The basic service provides for an audio channel of 7 kHz bandwidth. For the case of interworking with telephony and videotelephone, details are given in Table 1. The switchover between two different codings for speech should be either automatic or chosen by the user, and must not interrupt the normal flow of audio
19.5)	Data transmission	The data channel provided offers a bit error rate similar to that of the network used
20)	Operational, commercial	For further study

8 Dynamic description

In those instances when the requirement for graphic information transfer competes for the bandwidth required to achieve 7 kHz operation, the speech objective may be relaxed as long as there is no significant difference in speech quality from that available in the 64 kbit/s ISDN telephony service based on a bandwidth of 3.1 kHz.