Recommendation T/N 33-01 (Copenhagen 1987)

INTERNATIONAL DIGITAL AUDIOGRAPHIC TELECONFERENCING

Recommendation proposed by Special Group Integration (GSI)

Text of the Recommendation adopted by the "Telecommunications" Commission:

"The European Conference of Posts and Telecommunications Administrations,

considering

that there is a potential demand for audiographic teleconference services, nationally and internationally; such services having basic audiovisual facilities and a variety of optional facilities according to customers' requirements, and

recognizing

the need for interconnectability between audiographic teleconference terminals in different countries and in different customers' premises, preserving not only the correct electrical operation but also satisfactory acoustic conditions,

recommends

to the member Administrations that they should implement an International audiographic teleconference service as described and defined in the attachment hereto."

Contents

1. SCOPE

1.1. Introduction

A potential market for Audiographic Teleconferencing Service is now in evidence, resulting from technological progress which makes possible satisfactory performance in a commercially viable way.

Audiographic Teleconference is a means for teleconferencing, like telephoneconference and videoconference. It offers a conference facility of better quality and more possibilities than telephoneconference although less than videoconference because of the lack of moving pictures, but is less expensive than videoconference.

Information on human factor aspects is essential, but can only be obtained in the relevant context during a considerable period of experience.

In order to open the new service on the basis of a European standard and to avoid any restraints on the service development the Recommendation set out here offers the basic elements for an international digital Audiographic Teleconference Service.

Reference is made in the Recommendation to compatibility with other related services, for example videoconferencing as described in CEPT Recommendation T/L 01-02, and visual telephony: at some time in the future it could be anticipated that distinctions between these three services will tend to disappear, and should then be covered by a single Recommendation covering a range of bitrate options.

This document presents the Recommendation, including definition and description of the services and facilities offered, technical specifications for equipment, and alignment and operating procedures. A number of sections are included as headings only for completeness of the structure of the Recommendation, but in fact require further study.

Although this Recommendation contains references to both point-to-point and multipoint applications, for the time being only point-to-point working has been specified. Also the envisaged message channel has not fully been specified yet and will be added to this Recommendation later.

1.2. Definition of audiographic teleconference

Real time conferencing among individuals or groups of individuals at separate locations by means of audioterminals together with the possibility of transmission of non-moving visual information, telematic information and control and indications (speaker identification, floor request, etc.).

2. DEFINITIONS AND ABBREVIATIONS

See Annex 1.

3. SERVICES AND FACILITIES

This section provides descriptive information concerning the Audiographic Teleconference Service from the point of view of customers and marketing organisations. Further details, including technical specifications, are considered in later sections of the Recommendation.

3.1. Service description

A service for real time communication between groups of users in different locations, combining a good audio facility with optional auxiliary facilities such as still pictures, facsimile, telewriting, character or graphic format, etc. The service is applicable to companies' private audiographic teleconference rooms, as well as to public-access audiographic teleconference rooms for hire on an occasional basis; it is applicable to a variety of types of audiographic teleconference terminals, including multi-purpose committee rooms used only part-time for audiographic teleconference as well as dedicated studios; it is applicable also to "mobile" or "site-transferable" audiographic teleconference facilities and even desk top terminals.

The service is bidirectional via public switched or private telecommunication networks, and provides for interconnection of two or more terminals on an equal basis. In the case of switched calls, the user must be assured of availability (the probability of successfully dialling up the connection must be high), and may also require an assurance of confidentiality.

3.2. **Basic facilities**

The basic facilities consist in the primary means for audiovisual communication, namely an audiochannel together with transmission capability for additional control and telematic data and messages.

3.2.1. *Audio System*

By the design of the room and/or the equipment participants should be allowed to listen and speak simultaneously; speech transmission is of 7 kHz bandwidth.

3.2.2. *Data Transmission*

The basic service provides for a data port for the use of the customer in any way he wishes.

3.2.3. *Additional data channel*

An additional data channel is permanently included which may be used to provide for the control and management of the audiographic teleconference system and for user messages (see section 5.2.).

3.3. **Other facilities**

Since practical experience of audiographic teleconferencing is still very limited, there is no clear consensus as to which auxiliary facilities must be provided; the audiographic teleconference system specification therefore seeks to avoid constraints upon the users by providing for a wide range of possibilities, leaving to the customer the choice as to which should be implemented. Here follows a (non-exhaustive) list of the facility options which a potential user may consider. Subjectively the quality of the audio should not be greatly affected by providing this meeting aids.

Ergonomic reasons suggest that the number of display screens be reduced to a minimum.

The following auxiliary facilities may be provided by use of a data transmission channel as in 3.2.2. above:

3.3.1. *Still Picture TV System*

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.

3.3.2. *Facsimile*

Facsimile documents may be transmitted at any time on a point-to-point call, using a digital facsimile machine.

3.3.3. *X-Y Devices (including Telewriter)*

Several proprietary devices are available which can be used for remote pointing (cursor function) or remote writing, or both. Input devices include writing tablet, light pen, mouse, ball and joystick. The resulting information can be displayed independently, or superimposed on another picture.

3.3.4. *Still Pictures of Participants*

In the case where a company possesses a videoconference the videoconference type terminal may be used for audiographic teleconferencing, transmitting only still pictures.

The "split-screen" allows the combination of the outputs from two cameras, each viewing up to three people, into a single video signal; similarly, at the remote end, the two half-height pictures can be separated for display on two adjacent screens.

3.4. **Audiographic teleconference Terminals**

In general, it is not possible to standardise completely all aspects of audiographic teleconference rooms and equipment: much will depend on the size and shape of the accommodation which the customer has at his disposal, and on the choice of optional facilities. Furthermore, considerable scope can be allowed to equipment, suppliers in the styling and detailed facilities of their products, which will not materially affect the international service itself.

To ensure correct interworking of terminals on the international service, it is only necessary to ensure complete adherence to the electrical interface specifications and observance of the audio and video alignment procedures, as described in this Recommendation and in the Videoconference Recommendation T/L 01-02.

3.5. Controls and indications

In general, the controls to be operated by the user are kept to a minimum. Apart from call set up, audio mute and volume control, further controls will depend on the optional facilities present.

Normally one person ("the conductor") at each terminal may be appointed to operate these controls, though this is not essential. The system itself does not require the appointment of a conductor but provides the facilities whereby this may he done.

3.6. Quality of service

3.6.1. *Transmission Performance*

The worst-case transmission assumed in the above paragraphs is of an error rate of $1{:}10^4$. The networks to be used for the audiographic teleconference service are assumed to be so specified that such an error rate may only occur extremely rarely.

3.6.2. *Routing*

The choice of the routing should ideally not influence the quality of service. If satellite transmission is employed, the delay thus introduced affects the quality of service. More than two satellite hops should hence be avoided.

3.6.3. *Confidentiality*

Must be guaranteed.

3.6.4. *Audio Quality*

The basic service provides for an audio channel 7 kHz bandwidth. For the case of interworking with telephony, videoconference and videophone details are given in chapter 9.

3.6.5. *Data transmission*

The data channel provided offers a worst-case bit error rate of $1{:}10^4$.

3.7. Network requirements

Service offerings are based on digital access at 64 kbit/s or 2×64 kbit/s. Further details are given in section 5.1.

3.8. Operational aspects

Subject to study.

3.9. Geographical availability

The service will be available wherever 1 or 2×64 kbit/s transparent links are provided.

3.10. Tariffs and charging aspects

Each connection of the conference contributes to the charge. An additional charge must be paid for providing the Multipoint Control Unit, MCU(s) (and terminal if applicable). Normally the convenor has to pay for the whole conference. The option should be provided that each participant will be charged for his connection to the MCU(s) and the convenor additionally for the MCU(s).

3.11. The audiographic teleconferencing process

One can distinghish several phases in the overall process of an audiographic teleconference:

1. reservation
2. set-up
3. session
4. recovery and reconfiguration
5. disconnection

3.11.1. *Reservation*

In order to ensure the performance of an audiographic teleconference service for an appointed time reservation will be required at least for the MCU(s). Normally reservation will be handled by the customer as the service will be an automatic one. Administrations may provide also a manual version of a reservation system. This process is normally left to each Administration, however general guidelines for an harmonized operation of the International Audiographic Teleconference Service follow.

The reservation is performed through the operator service. The following information must be given by the convenor:
— list of participating terminals,
— starting time of the session,
— closing time of the session,
— symbolic name of the session,
— symbolic names of the terminals.

It will be possible for a convenor to get access to the Reservation Centre (RC) through an ordinary telephone connection to register a reservation for a conference. A data terminal can be used in an automatic system, alternatively it can be performed by voice in a manually operated system.

The RC determines which MCU(s) and which connections are required for the meeting and subsequently reserves these facilities. It modifies each terminal to what MCU it has to connect.

During a conference session the conductor may have the possibility to get access to the reservation diary by using his control and indicator's device. In this way it will be possible during a conference meeting to agree on a following conference.

3.11.2. *Set-up of the conference session*

The start of a conference is decided by the conductor. There are two possibilities in switched networks: the connections are established from the terminal or the connections are established from the MCU's. For the case that a terminal is to be added during a meeting initiation from the terminal should be possible.

In the case that more MCU's are connected the RC notifies the MCU's which MCU establishes the connection(s) to the other MCU(s).

In order to avoid ambiguity each conference session may have a symbolic name which is given in the reservation phase.

As soon as the connection between a terminal and its MCU is established, the terminal enters the set-up mode. The terminal remains in that mode until the set-up procedure has been completed. During the meeting the set-up mode can be re-entered, e.g. when participants change.

In the set-up phase the terminals may exchange information about:
— terminal names,
— names of participants,
— terminal characteristics (meeting aids, etc.).

This information is not stored in the MCU's but in the terminals.

3.11.3. *The conference session*

General

The following section applies to a conference where the Message Channel facility is available. Otherwise only a non-conducted mode can be used.

For a successful conduct of the conference the role of conductor will be defined. The conductor is one of the participants at a given terminal.

At the start of the conference session the default mode of conversation is "non-conducted".

The token of conductorship can be taken by a terminal during the set-up phase and can be handed over the another terminal during the conference session.

Each terminal must be equipped in such a way that it can potentially accept the token of conductorship.

The terminal will offer the conductor and other conferees certain functions to lead the conversation between all participants such as request for floor signal by a conferee and grant request for floor signal by the conductor. Use of these functions can be realized by manipulating the control and indication facilities of the terminal.

Speakers' identification. In all modes of conversation the principle of identification of the current speaker can be used. Therefore each audiographic teleconferencing room is equipped with a display visible for all participants on which the identification of the speaker (name, terminal of origin) will be displayed. Alternatively, this may be an individual display associated with each microphone set.

Speaker localization could be used.

Conversation modes

Three modes of conversation may be supported in the system:

1. non-conducted mode
2. half-conducted mode
3. conducted mode

Non-conducted mode. In the case of the non-conducted mode each participant can speak at any moment he wishes. Enabling and disabling of the microphone set is under participant's control.

Half-conducted mode. In the case of the half-conducted mode of conversation the participant will issue a request to speak and has to wait until this request is granted. The decision is taken by the equipment. An upper limit of n concurrently speaking participants applies, where n (e.g. 3) is a default value or determined by the conductor.

No queue mechanism is used in this mode. When a speaker's request is not granted because the upper limit is exceeded, the speaker will have to issue another request.

Note 1. The half-conducted mode realizes speaker's interruption. As queueing is not used, quick reaction on the current speaker is possible.

Note 2. Necessity of the half-conducted mode is to be investigated by human factor experiments.

Conducted mode. In the case of the conducted mode the participant will issue a request to speak and has to wait until this request is granted. The participant has the possibility to cancel his request before it is granted.

In the conducted mode the conductor receives all requests-to-speak and can decide when and until which moment each individual participant is allowed to speak. The conductor will then enable and disable each participant's microphone set, alternatively the outgoing sound signal from each location could be disabled and enabled.

In the conducted mode a queueing mechanism can be used which handles all requests-to-speak on a first come–first served basis. When queueing is used each participant will end his speech by issuing a dedicated notification (e.g. by pushing a button). The conductor, however, has still the privilege to interrupt a current speaker, to disable his microphone set and to pass the floor to another participant.

All modes. In all modes of conversation the conductor will have an indication of the speakers currently involved.

In all modes of conversation the conductor is able to enable or disable each speaker's microphone set.

The actual mode of conversation will be indicated permanently to all participants at all locations.

3.11.4. *Recovery and reconfiguration*

Provisions must be made in case of the loss of a transmission path between certain parts of the total audiographic teleconferencing system during a conference session. Detection of loss of a connection to a terminal should be performed by the applicable MCU, in multipoint cases, or other terminal in point-to-point cases. The another terminals should get a message saying which terminal has been disconnected. Preferably an automatic re-establishment procedure for the lost line should be performed otherwise a disconnected terminal will have to reset up the call. If the conductor's terminal is lost then the conference will return to non-conducted mode. After reconnection a procedure for re-assignment of the token of conductorship has to be performed and the other terminals should receive a message that the lost terminal has been reconnected.

3.12. List of service attributes

Information transfer attributes

1.	Transfer mode	circuit
2.	Transfer rate	64 kbit/s and 2×64 kbit/s (2×64 kbit/s only when high-speed facsimile or SPTV is to be sent)
3.	Transfer capacity	unrestricted digital
4.	Structure	8 kHz integrity
5.	Establishment of comm.	demand and reserved both should be possible
6.	Configuration of comm.	point-to-point and multipoint both should be possible
7.	Symmetry	bidirectional symmetric

Access attributes		
8.	Access type	basic access (in case of ISDN)
9.	Access channel and rate	64 kbit/s and 2 × 64 kbit/s (see under 2)
10.	Info access structure	multimedia (1 × 64 kbit/s), multimedia and multiservice (2 × 64 kbit/s)
11.	Signall. access protocols	
11.1.	Layer 1	depending on the network used, to be determined by network operator and customer
11.2	Layer 2	
11.3.	Layer 3	
12.	Inf access protocols	
12.1.	Layer 1	X.21 leased line
12.2.	Layer 2	—
12.3.	Layer 3	—
13.	*Type of user information*	Audio, SPTV, X-Y devices, Facsimile, Telewriter, Telematic data, user-to-user message
14.	*Transport attribute*	none
15.	*Session attribute*	none
16.	*Presentation attributes*	
16.1.	Audio	G.722, G.711 (for compatibility with telephony)
16.2.	Video	SPTV (to be defined)
16.3.	Auxiliary	T.30, T.6 (Facsimile), rest to be defined
16.4.	Dialogue	message channel, this is for further study
17.	*Application attributes*	
17.1.	Audio	microphone + loudspeaker, echo control device, headset (for further study)
17.2.	Video	object/document camera, person camera (for still pictures), monitor(s)
17.3.	Auxiliary	T.4, T.5 (facsimile), rest to be defined
17.4.	Dialogue	dedicated keyboard + display
General attributes		
18.	Supplementary services	in the switched service: as for telephony
19.	Quality of service	see section 3.6.
20.	Interworking	Telephone, Videophone, Videoconference (see section 9.)
21.	Operational, commercial	see sections 3.8. and 3.10.

4. NETWORK

4.1. Access

4.1.1. *Transmission*

The following networks are likely to be used for audiographic teleconferencing:
ISDN
IDN
leased lines
data networks
satellite networks, etc.

4.1.2. *Interface*

The interface of the terminal and MCU to the network (interface surface B in Figure 1 (T/N 33-01) and B′ in Figure 4 (T/N 33-01)) is X.21, leased line version with byte timing. The adaptation between terminal or MCU and the network is up to the national Administrations and manufacturers.

When ISDN will be available it is likely that a network interface adapting to the S interface will be incorporated in the audiographic teleconference terminal.

4.2. **Transfer rate**

One or two transparent 64 kbit/s transmission paths are necessary. It is presumed that bit and byte timing are provided by the network.

If the network does not provide byte timing, then the recovery time following loss of frame alignment may be extended.

4.3. **Call**

Call set-up, clearing and booking (reservation) procedures are necessary entities of the network to be used. This subject is to be determined by the national Administration.

5. BITRATE ALLOCATION, MESSAGE CHANNEL, FRAME STRUCTURE

5.1. Bitrate allocation

For the basic service information is to be multiplexed within one 64 kbit/s stream. For special cases (fast still picture, fast facsimile, full 64 kbit/s speech) a second 64 kbit/s stream can be used.

Thus either one or two 64 kbit/s channels can be used. In ISDN networks it will be possible to use the 2nd 64 kbit/s channel temporarily, on a on-demand basis, only on moments in the meeting when it is required.

	One 64 kbit/s channel	Two 64 kbit/s channels
Speech	56 kbit/s normally 48 kbit/s during fax or still picture transmission	56 kbit/s in 1st channel
Facsimile or still picture	8 kbit/s with simultaneous speech	64 kbit/s in 2nd channel
Service channel	8 kbit/s	8 kbit/s in 1st channel

Table 1 (T/N 33-01). Bitrate allocation.

The service channel contains the frame-alignment signal, bitrate allocation signal, time slot allocation (for use in videoconferencing) and user-to-user or terminal-to-terminal (or MCU) information.

The way in which a second B-channel will be used is for further study.

5.2. Message channel

The specification for a message channel is still under study, therefore this specification will be added later on.

The message channel is to be specified in Annex 2 of this Recommendation and is to be accommodated in the application channel as specified in Annex 3.

The message channel will be used for the following purposes.

(a) At the start of an audiographic teleconference to send initialisation information concerning configuration of terminals (for example names of participants) and room parameters between terminals or between terminals and the MCU.

(b) During the audiographic teleconference the message channel will be used to communicate requests and acknowledgments between terminals for the use of meeting aids.

(c) Using the Keyboard and Display, message may be sent selectively between participants over the message channel.

(d) The message channel could be used for low bitrate aids, e.g. Telewriter.

5.3. **Frame structure**

The frame structure is described in Annex 3 of this Recommendation.

The data channel (see 3.2.2. and 3.6.5.) occupies bit 7 in the frame thus forming an 8 kbit/s channel. Bits No. 8 of octets numbers 41 to 80, termed "additional data channel" in Figure 1 (T/N 33-01) and section 3.2.3. are reserved for the message channel. Bits number 8 of octets numbers 17 to 40 may be used for other applications such as temporary C&I's for national applications as long as no message channel is specified yet, or for videoconference applications (as done in CCITT SG XV).

5.4. **Mode switching**

CCITT have defined various combinations of bitrate for wideband speech and data path capability within a 64 kbit/s channel. These combinations of bitrate are known as modes and are defined thus:
Mode 0: 64 kbit/s PCM speech (A-law)
Mode 1: 64 kbit/s wideband speech
Mode 2: 56 kbit/s wideband speech plus 8 kbit/s data capability
Mode 3: 48 kbit/s wideband speech plus 16 kbit/s data capability (8 + 8 kbit/s)

An unpartitioned 64 kbit/s channel therefore corresponds either to Mode 1 above or to a PCM speech channel (A-law G.711).

In order to provide general compatibility between audiographic teleconference terminals and terminals using an unpartitioned channel, the audiographic teleconference terminal can work in three configurations:

1. Mode 2 or Mode 3 with frame alignment and data capability.
2. Mode 1 with no data capability.
3. Mode 0–PCM speech to G.711 with no data capability.

At the beginning of a session it is necessary for terminals to know which mode they should assume in order to enable different types of terminals to be interconnected. For example an audiographic teleconference terminal would need to switch modes in order to interwork with a digital telephone.

For the time being the working modes of the audiographic teleconference terminal, and if applicable the MCU(s), will for the set-up situation (starting mode) be adjusted manually. Later on this could be performed automatically according to the method to be standardised in CCITT (Draft Recommendation G.72Y, Annex A). Mode 2 will be the default mode.

Therefore in terminal-to-terminal communication (point-to-point) the user should know for the time being (from a directory or previous communication) which terminal he can expect to connect to. In multipoint communication an audiographic teleconference terminal in the set-up phase always works in mode 2; the MCU's are informed by the RC which kind of terminal they have to connect to.

Mode switching during the conference will be signalled in the BAS word.

6. CODING

Coding of Audio is described in CCITT Recommendation G.722, coding of meeting aids in 7.5. and coding of messages in Annex 2.

7. TERMINAL SPECIFICATION

This chapter consists of a functional description of the terminal, specifications for the audio system divided in interconnection, far end and near end and a description of the meeting aids that could be used.

As concerns the terms Interconnection, far end and near end, the explanation follows:
"Interconnection specifications: these are the basic requirements to allow the communication between several locations, without paying attention to the quality of the communication."
"Far-end quality specifications: requirements, the local audio terminal must fulfill in order to ensure a minimum quality on the sent signal which is received by the far-end users."
"Near-end quality specifications: these are the requirements on the local audio terminal which allow a certain level of quality as it is perceived by the local users."

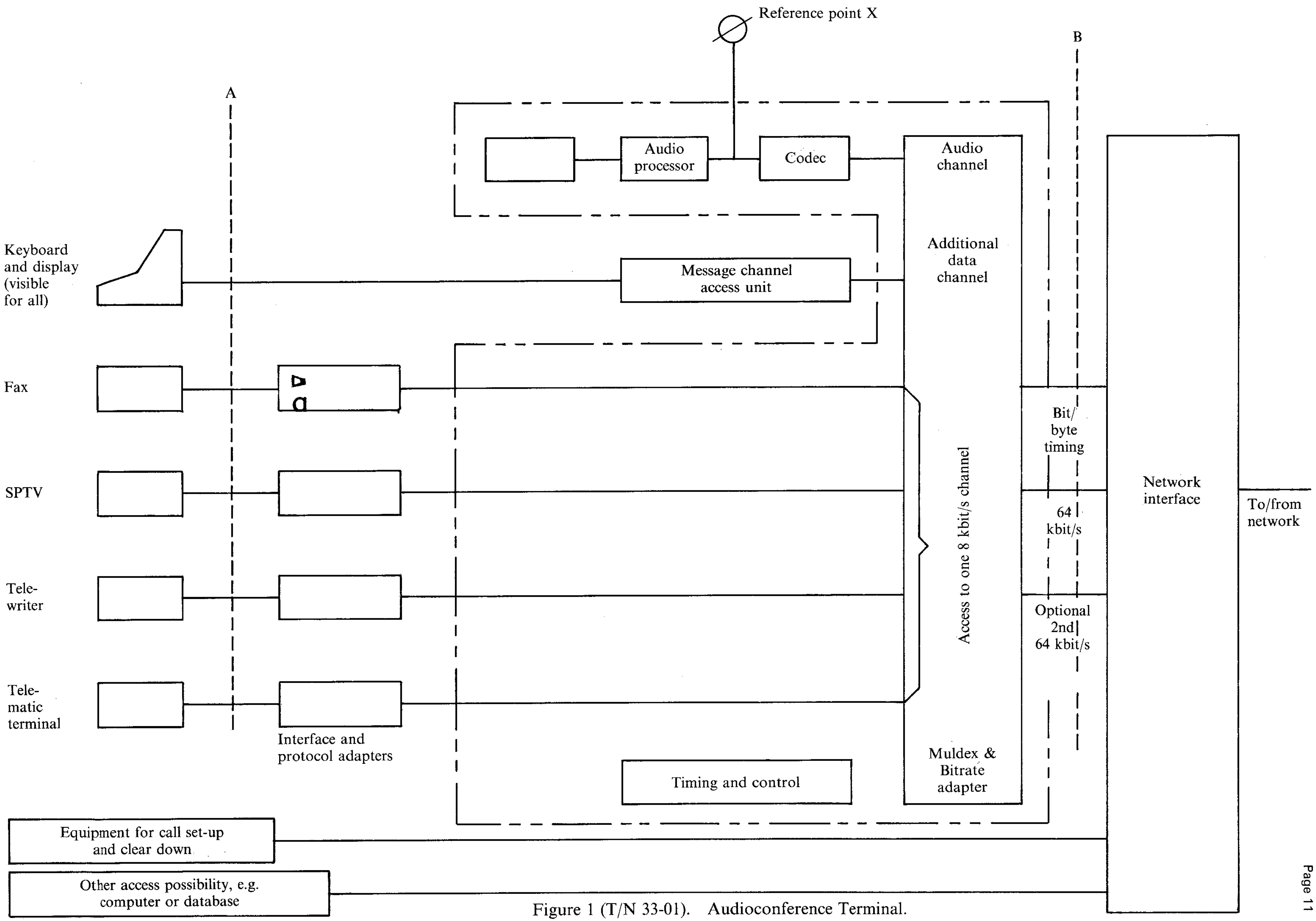


Figure 1 (T/N 33-01). Audioconference Terminal.

7.1. Functional description of an audiographic teleconference terminal

The functions of the equipments are (see also Figure 1 (T/N 33-01)):

*Loudspeaker/Microphone Unit**

This unit contains a (set of) microphone(s) and (a) loudspeaker(s).

* In principle it is possible to employ headset/microphone combinations. The use of such equipment is for further study.

Loudspeaker microphones and amplifiers

The Audio Amplifying Equipment amplifies the microphone and loudspeaker signals. This unit may contain test generators and audio-level measurement equipment for performing acoustic alignment adjustments.

Audio Processor Unit

This Unit performs the function of equalizing conference room parameters and setting send and receive side audio levels as well as echo suppressing or cancelling (although this could alternatively be done in the digital path). The unit produces a speaker identification signal and microphones can be switched individually on and off by this unit.

Audio Codec

The processed audio is coded and decoded in the Codec. The coding scheme used is as described in CCITT Recommendation G.722 with a bitrate of 48, 56 or 64 kbit/s. The bitrate in use at any time depends on the activity of other meeting aids (e.g. Still Picture). The bitrate switching is initiated by signalling from the Timing and Control Unit.

Keyboard/Display Unit

The Keyboard/Display Unit enables the conferees to exchange messages such as floor request (for this purpose separate pushbuttons for each participant are recommended) and input of names of participants. In addition messages such as speaker identification would be shown on a Display visible by all. Also this Keyboard/Display could be used to access databases and other computers (such as management support systems or a computer conference system) via the telephone network. Alternatively a separate Keyboard/ Display unit could be used for this purpose as shown in Figure 1 (T/N 33-01).

Message Channel Access Unit

This Unit accesses the message channel for conveying messages between Audiographic teleconference Units and ensures that such information signals as terminal characteristics are transmitted to the other terminal or MCU.

Muldex and bitrate adapter

The signals from the Audio Codec, Message Channel Access Unit and from the Interface and Protocol Adapters are multiplexed in the Multiplexer/Demultiplexer (Muldex). The framing structure according to which the multiplexing is performed is described in Annex 3.

Interface and Protocol Adapters

The Units intermediate between the standard meeting aids (e.g. SPTV, Telewriting) and the Muldex. Thus they handle the procedures to interface the CCITT standardised (or still to be standardised) equipments.

Timing and Control Unit

This Unit arranges all timing matters in the Terminal, deriving the timing from the Network or from the frame. This Unit also controls all functions of the Terminal: it controls mode switching. It effects interchange of all system control and set-up information with the other connected Terminal or MCU such as Terminal characteristics. It adapts the cursor signal to the particular Telewriter and SPTV equipments which are being used.

Network Interface

The Network Interface provides bit and byte-timing derived from the network clock. If required locally this unit can be responsible for call set up through the D-channel of ISDN.

The Network Interface may differ from country to country and will therefore not be described in this specification.

Meeting Aids

Meeting Aids (such as Fax, SPTV, Telewriter and Telematic terminal) will be CCITT standardised equipments (unless a standard is not available and not emerging). The Meeting Aids are described in more detail in section 7.5. These equipments will be connected to the Interface and Protocol Adapters which signal the use of a Meeting Aid to the Timing and Control Unit. When a request to use a meeting aid has been sent, and acknowledged via the message channel the Timing and Control Unit ensures that the Muldex makes available the appropriate time slots and takes care of signalling the accompanying frame partitioning to the MCU (in multipoint connections) or to the other terminal in point-to-point connections.

A keyboard and display-unit serves to exchange messages between terminals or between a terminal and the related MCU. In addition terminal equipment can be connected directly to the network interface, e.g. for database access.

Cursor

Some Meeting Aids like the telewriter and the SPTV system require a Cursor. The Cursor position may be transmitted through the Message Channel, in a separate channel or as part of the telewriter signal (under study).

Equipment for Call Set-up and Clear-down

In Figure 1 (T/N 33-01) a connection point is indicated which is used to set up and clear down the call through the Network Interface.

Basic Audiographic teleconference Terminal

The part of the terminal enclosed in by the dashes and dots in Figure 1 (T/N 33-01) is termed a Basic Audiographic teleconference Terminal or Basic Terminal. Every Audioconference Terminal should contain this part of the system.

Reference Interface Surfaces

Two Reference Interface Surfaces have been defined, termed Interface Surface A and B in Figure 1 (T/N 33-01), each indicated by a dotted line: — — —.

The Interface Surface A represents the interface-points of (standardized) inputs and outputs from Meetings Aids.

The interface on surface B is X.21, leased line.

7.2. Interconnection specifications for the audio system

7.2.1. *Audio channel*

In the normal working mode wideband coding is used while the terminal is equipped with an A-law codec for interworking with digital telephone sets.

Audio coding

An audio channel applying wideband coding shall meet CCITT Recommendation G.722. An audio channel applying A-law PCM coding shall meet CCITT Recommendation G.711.

Transmission characteristics

The transmission characteristics of the audio channel shall be in accordance with CCITT Recommendation G.722 for a wideband application and Recommendation G.711 for an A-law PCM application.

The audio ports (reference point X in Figure 1 (T/N 33-01)) shall in addition to the requirements given in Recommendations G.722/G.711 fulfill the following requirements.

— Impedance
  The nominal impedance at the input and output audio ports shall be 600 ohms, balanced. The return loss, measured against the nominal impedance, shall not be less than 20 dB over the frequency range.
  — 50 Hz to 7 kHz (wideband)
  — 300 Hz to 3.4 kHz (PCM)

— Relative level
  The nominal relative level at the input and output audio ports shall be 0 dBr at the reference frequency 1,000 Hz for the wideband as well as the PCM application.

— Overload point
  — + 9 dBmO for wideband coding (Recommendation G.722)
  — + 3.14 dBmO for PCM coding (Recommendation G.711)
  Care shall be taken to guarantee that any amplitude limiting, where provided, occurs before the input antialiasing filter.

7.2.2. *Measuring instruments and methods of measurement*

Acoustic measurements

Acoustic levels shall be measured by a Sound level meter according to Publication IEC-651 (Class 1).

The sound source calibration should be performed with the Sound level meter in "Linear" mode, while the reception alignment and the measurement of the room noise shall be performed in the "A weighting" mode.

Electric measurements

Output line levels and noise level shall be expressed in dBm and measured across a 600 ohms resistive termination at reference point X.

The measuring instrument shall perform a true RMS measurement.

7.2.3. *Transmission sensitivity*

7.2.3.1. Acoustic test signal

The acoustic signal to be used in the measurements for the audio alignment should be generated by a sound source as shown in Figure 3a (T/N 33-01), which consists of two parts:

i) Noise source:
The noise source consists of a noise generator, simulating an average speech spectrum, followed by a 7 kHz low pass filter rolling off at a minimum rate of 48 dB/oct.

ii) Artificial mouth
The artificial mouth shall comply with Recommendation CCITT P.51.

The third octave spectrum of the acoustic signal generated at the MRP* is given in Table 2 (T/N 33-01), together with tolerances.

* Located on the axis, at a distance of 25 mm in front of the lip ring (see Recommendation CCITT P.64, Annex A).

⅓ oct. center freq.	Sound pressure level	Tolerance
Hz	dB SPL	dB
100	70.9	—
125	74.8	+ 3/ − 6
160	77.6	+ 3/ − 6
200	79.6	+ 3/ − 6
250	80.6	± 3
315	80.6	± 3
400	80.7	± 3
500	79.9	± 3
630	78.6	± 3
800	77.0	± 3
1,000	75.1	± 3
1,250	73.0	± 3
1,600	71.0	± 3
2,000	68.9	± 3
2,500	67.1	± 3
3,150	65.4	± 3
4,000	64.2	± 6
5,000	63.4	± 6
6,300	63.4	± 6

Table 2 (T/N 33-01). Long term third octave spectrum of the acoustic signal (see CCITT P.51).

Electric signal

The electric signal used for aligning the receiving side of the equipment should be generated by a source with an internal resistive impedance of 600 ohms. It consists of a noise simulating an average speech spectrum, band limited to 7 kHz by a low pass filter with a roll off of at least 48 dB/oct. It should be applied to the receiving port of the equipment at − 20 dBm. The third octave spectrum of the signal is given in Table 3 (T/N 33-01), together with tolerances.

1/3 oct. center freq.	Signal level	Tolerance
Hz	dBm	dB
100	−38.4	±3
125	−34.5	±3
160	−31.7	±3
200	−29.7	±3
250	−28.7	±3
315	−28.3	±3
400	−28.6	±3
500	−29.4	±3
630	−30.7	±3
800	−32.3	±3
1,000	−34.2	±3
1,250	−36.3	±3
1,600	−38.3	±3
2,000	−40.4	±3
2,500	−42.2	±3
3,150	−43.9	±3
4,000	−45.1	±3
5,000	−45.9	±3
6,300	−46.2	±3

Table 3 (T/N 33-01). Long term third octave spectrum of the electric signal.

7.2.3.2. Send side alignment

The sound source, calibrated for providing 89.3 dB SPL at the MRP, is positioned over the edge of the conference table, as shown in Figure 3b (T/N 33-01) (see Recommendation CCITT P.34) on the center line of each conferee's position.

The microphone gain controls must be adjusted to achieve, for each position of the source, an output line level of −20 dBm (± 1 dB).

7.2.4. *Stability test*

The audiographic teleconference terminal shall have a stability margin of 3 dB when the microphone and loudspeaker paths are looped in reference point X in Figure 1 (T/N 33-01) and the sound source is activated as described in 7.2.3.1. and 7.2.3.2. During the measurement the volume control shall be in maximum position.

7.2.5. *Impedances*

The input and output impedances of the audioprocessing units shall be 600 ohms. The return loss shall not be less than 20 dB in the frequency range 50 Hz to 7 kHz.

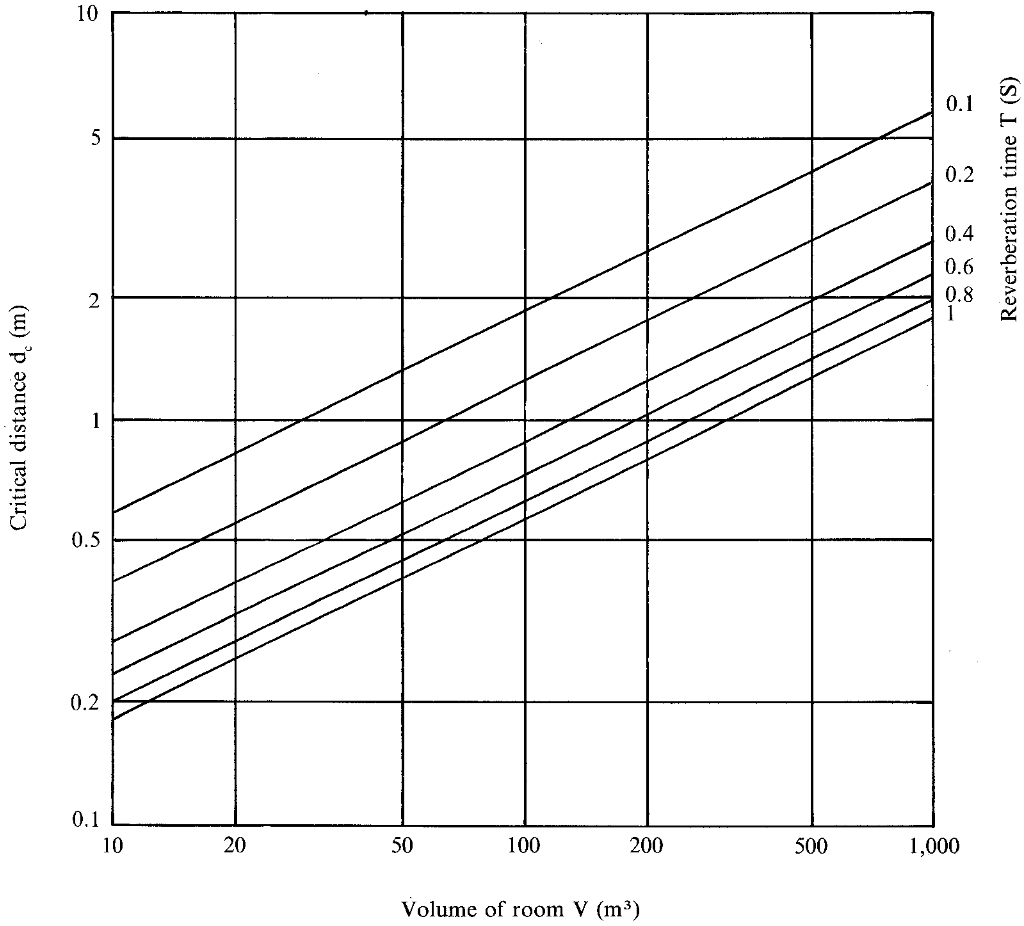


Figure 2 (T/N 33-01). Influence of the room volume to the quality of sound.

7.3. Far-end quality specifications for the audio system

7.3.1. *Echo performance*

7.3.1.1. Acoustic Echo Control

To get satisfactory suppression of acoustic echoes it is necessary to provide the audio processor with either an echo canceller or an echo suppressor. The echo cancellation technology is recommended if highest possible speech quality performance is aimed at.

The following requirements apply:

— the overall echo return loss of the audio system shall be as described in 7.3.1.2.;

— the echo canceller shall permit double-talk with negligible speech quality degradation.

If simple terminals with headsets are used (which is disuaded) then Acoustic Echo Control will probably not be necessary, this is for further study.

7.3.1.2. Overall echo return loss

The overall echo return loss of the audio system shall be measured in reference point X of Figure 1 (T/N 33-01) with the volume control in maximum position. When a noise source with a gaussian amplitude distribution, simulating a band limited average speech spectrum and adjusted for delivering −20 dBm into 600 ohms, is connected to the input port, the level measured over 600 ohms at the output port shall not be higher than −55 dBm.

7.3.2. *Electric noise*

The electric noise emitted by the audiographic teleconference system at the reference point X shall be no more than −55 dBm.

The measurement must be done with no conferees in the conference room and without incoming signals on the reception of the equipment in order not to activate the microphone circuits.

The noise emitted by the audiographic teleconference system at the reference point X when the microphones are active shall be no more than − 50 dBm. It must be measured by forcing the system in the emission mode as if one speaker is active in the room.

Care should be taken in order to assure that no frequency components are generated below 100 Hz and above 4 kHz at a level higher than −40 dBm. Because of noise sensitivity of the A-law codec.

7.3.3. *Room noise*

The room noise should be measured at the conferees positions, in the absence of conferees as shown in Figure 3c (T/N 31-01).

The background noise level should be as low as possible. It is preferred that the noise level is lower than 40 dB(A).

Maximum talker to microphone distance arising from background noise level

The following maximum d_{max} should be maintained, in order to achieve a signal-to-noise ratio of at least 30 dB for speech level of average talkers:

$$d_{max} = 50.10^{-(L + 10 \log n)/20} \text{ m} \quad \dots (1)$$

where L is the long-term average background noise level in dB(A), and n is the number of simultaneously open microphones.

In the case where cardioide microphones are used, the calculated distance d_{max} may be increased by 50%.

Preferred maximum talker to microphone distance

The preferred maximum talker to microphone distance is given by the least of the values obtained from formulas (1) and (2), see section 7.4.4.2. In the case of directional microphones this distance may be increased by 50%.

Relation of room volume, critical distance and reverberation time

Figure 2 (T/N 33-01) shows the influence of the three parameters:
— volume of room
— critical distance
— reverberation time

From this it is clear that a room of a proper size is required to ensure a good sound quality.

7.3.4. *Reverberated field picked up by the microphones*

The sound source, calibrated for providing 89.3 dB SPL at the MRP, is positioned in order that the distances between the sound source and all the microphones are greater than three times the distance between the microphone and the position defined in Figure 3 (T/N 33-01) for the send side alignment.

It is also recommended that the source is, at least, one meter from the walls. Then the signal measured at point X shall not be more than −27 dBm.

It must be measured by forcing the system in the emission mode as if one speaker is active in the room.

The test must be performed for each microphone in the room.

7.3.5. *Electroacoustical specifications for the send side*

7.3.5.1. Microphones

The electroacoustical characteristics of the microphones should conform to IEC Publication 581-5.

7.3.5.2. Frequency responses

The frequency responses of both the sending and receiving electrical channel should be flat (within plus or minus 1 dB) in the frequency range 100 Hz-7,000 Hz.

The send side frequency characteristics shall be measured for an output signal of 0 dBm. The receive side frequency characteristics shall be measured for an input signal of −20 dBm.

7.3.5.3. Octave band measurements

Further studies required.

7.3.5.4. Distortion

The harmonic distortion of the sending and receiving electrical channel should not exceed 1% in the frequency range 200 Hz-3,500 Hz.

The send side distortion shall be measured for an output signal of 0 dBm. The receive side distortion shall be measured for an input signal of −20 dBm.

7.3.6. *Double-talk performances*

Further studies required.

7.4. Near-end quality specifications for the audio system

7.4.1. *Electroacoustical specifications for the receive side*

7.4.1.1. Loudspeakers

The electroacoustical characteristics of the loudspeakers should conform to IEC Publication 581-7.

7.4.1.2. Octave band measurements

Further studies required.

7.4.2. *Reception sensitivity*

7.4.2.1. Volume Control

The audiographic conference terminal shall be provided with a volume control. The gain at maximum position should conform 7.4.2.2.

7.4.2.2. Receive side alignment

The noise source simulating a band limited average speech spectrum, adjusted for delivering −20 dBm into 600 ohms, is connected to the input port of the system during the alignment procedure. The receiving gain shall be adjusted in order to read a sound pressure level of at least 54 dB(A) at the conferees positions, as shown in Figure 3 (T/N 33-01). Experiences have shown that increased listening levels are desirable. Further study is required. The maximum level is determined by the terminal stability (see 7.3.1.1. and 7.2.4.).

The alignment procedure should be performed with the volume control in the maximum position.

7.4.3. *Background noise*

7.4.3.1. Noise emitted by loudspeakers

The noise emitted by the loudspeaker(s) at each participant position should not exceed the background noise at the same point for each octave band with centre frequencies from 63 Hz to 16 kHz.

The measurement must be done when the input of the system at the reference point X is terminated with a 600 ohms resistive load.

7.4.4. *Room acoustics*

The acoustic treatment of audiographic teleconference rooms should be carefully designed in order to achieve best overall electroacoustic performances.

Whilst audiographic teleconferencing may be possible in rooms not meeting the following Recommendations the overall system performance is likely to be degraded.

Particularly, barrel effects and background noises should be prevented and a suitable sound proof insulation of the room must be guaranteed.

The main parameters to be taken into account when installing audiographic teleconference systems are:
— room reverberation,
— background noise,
— sound insulation (privacy).

While the former two parameters independently affect the maximum talker-to-microphone allowable distance, the latter ensures privacy between the conference participants.

All given measurement values are for rooms not containing participants.

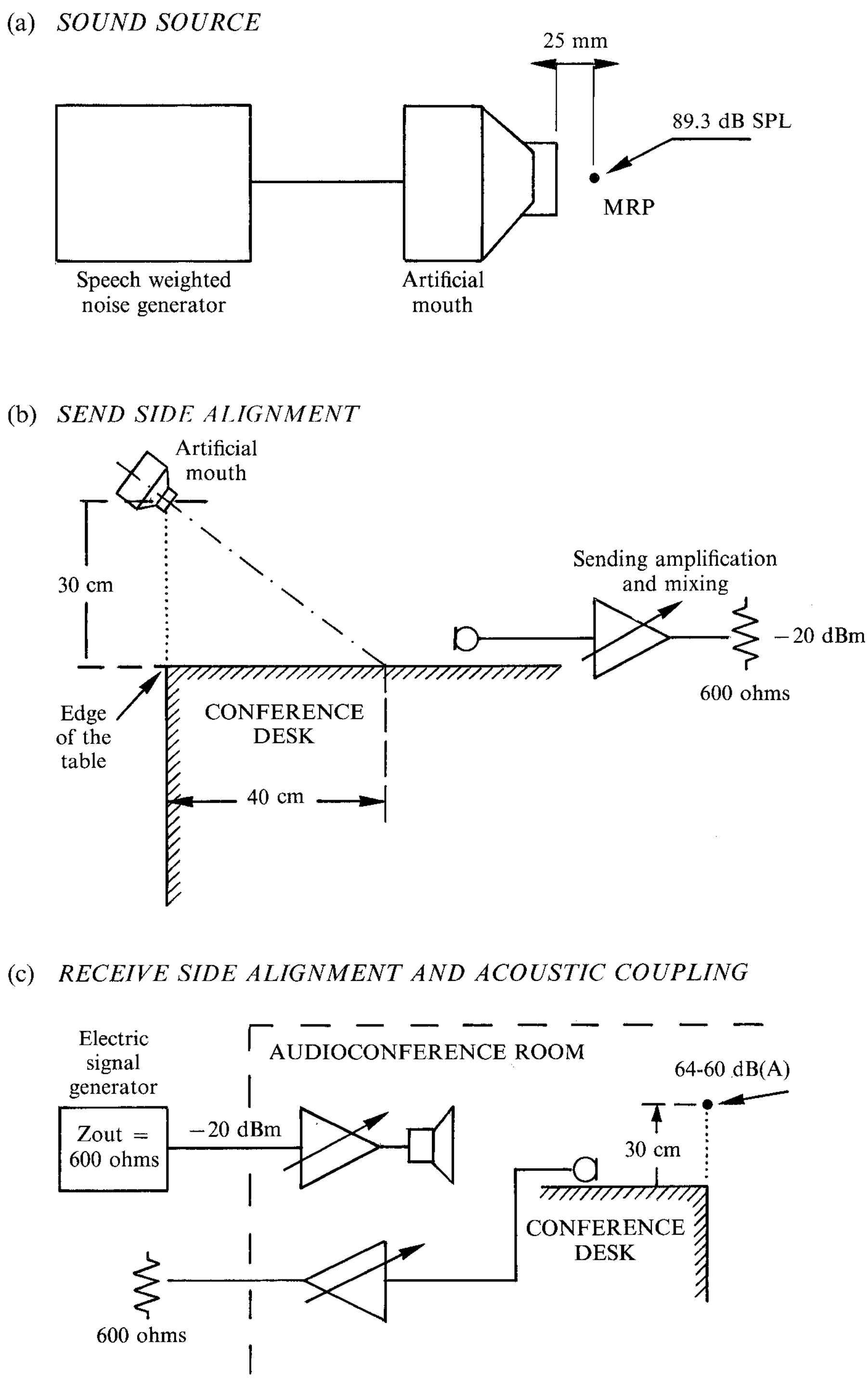


Figure 3 (T/N 33-01). Audio alignment.

7.4.4.1. Sound insulation

In order to ensure a good privacy for the conferees the speech level transmitted to premises around the conference room shall be at least 15 dB below the noise level in these premises. Assuming that the maximum talking and listening level in the reverberation field in the conference room is 70 dB(A) and the noise level in the premises is around 40 dB(A) it follows that the sound insulation between the rooms must be at least 45 dB.

7.4.4.2. Reverberation time

For avoiding barrel effects the microphones shall be placed in such a way that the direct sound field of associated talkers is high enough compared to the reverberation field. This is achieved if the maximum talker-to-microphone distance d_{max} is restricted to half the critical distance d_c of the room:

d_{max} seq (⅛) sqr (0.161.V/(pi.T)) m ... (2)
V: room volume m^3
T: reverberation time s

Extremely directional microphones should not be used. When directional microphones (cardioide) are utilized, the distance may be increased by 50%.

The reverberation time of the room should be measured, alternatively the reverberation time of geometrically simple rooms can be calculated from room absorption characteristics of walls and furniture and from the room dimensions. Realistic talker-to-microphone distances for microphones placed at the conference table are normally attained if the reverberation time is less than 0.4 s in the frequency range 100-7,000 Hz (see Figure 2 (T/N 33-01)). With a longer reverberation time than this figure particular care should be taken in the placing of microphones and loudspeakers and regarding the echo control performance.

7.4.5. *Double-talk performance*

Further studies required.

7.5. Meeting aids

7.5.1. *Meeting aids*

Besides speech the following meeting aids can be used in audiographic teleconferencing.

SPTV

SPTV is a method of transferring still images derived from a video source between two or more locations by electrical means.

The equipment to be used with audiographic teleconference is of the transceiver type, that is each unit may both transmit and receive pictures although not necessarily simultaneously. The system may provide for indefinite retention and display of the still picture at the remote end.

Associated with the SPTV equipment will be suitable TV cameras and monitors operating to CCIR 625-line standards. For optimum performance these items should be in accordance with the requirements given in the CEPT Recommendation on International Digital Videoconference Service (T/L 01-02, section 6.1.). Colour or Monochrome equipment may be used.

The picture information obtained from the cameras will be coded for transmission in order to provide suitable update times.

The equipment will be provided with a control panel to give control of picture capture, transmitted resolution and transmission of pictures. Facilities will exist to control the movement of a cursor overlaying the displayed picture. The Recommendation CEPT T/A 06-01 will be used as the basis of the level 6 presentation protocol.

At the physical electrical level a single X.21 interface will be used to interconnect with the audiographic teleconference unit. Both picture information and SPTV control information will be sent through this interface. This subject is under study in CCITT SG VIII and ISO.

Facsimile

Group 3 or 4 facsimile will be used. Since group 3 facsimile has an analogue interface to the network, the IPA has amongst other things to take care of the A/D-conversion.

Telewriter

There is no agreed specification for telewriter equipment yet. Work on specifying telewriting is done in CCITT SG VIII.

Keyboard and Display Unit

The Keyboard and Display Unit is planned to be used to transfer messages via the message channel to an another other audiographic teleconference terminal or to a MCU.

Controls and Indications between terminals

Between terminals the following signals are planned to be conveyed via the message channel:
— floor request
— grant floor request
— speaker identification
— control of speakers' microphone (by conductor)
— room identification
— terminal parameters (available equipment, etc.)
— line breakdown signalling (if not provided by the network)
— signals for control of data channels and meeting aids

These may be transmitted on a temporary basis for national or experimental purposes using a simple bit protocol as mentioned in 5.3.

7.5.2. *Interfaces and protocols for meeting aids*

The Basic Terminal depicted in Figure 1 (T/N 33-01) has a number of input ports connected to IPA's. These IPA's are provided only as necessary.

The function of the IPA may involve to strip the existing protocol of the meeting aid and substitute a new protocol internal to the audiographic teleconference system.

8. MULTIPOINT AUDIOGRAPHIC TELECONFERENCE

A description of Multipoint Audiographic Teleconference is given here although it is stated that multipoint teleconferencing is only possible with the help of a message channel. As mentioned earlier the specification of this message channel will be added later on.

8.1. General requirements

A Multipoint Control Unit (MCU) is a piece of equipment located at a node of a digital network (terrestrial or satellite) which receives several 64 kbit/s channels from network access ports. Each access port is connected to corresponding audiographic teleconference terminals or to another MCU. The purpose of the MCU is to permit the transmission of coded audio and information from meeting aids between a number of separated audiographic teleconference terminals.

Although particular attention must be paid to network topology in the case of satellite transmission, see paragraph 8.2., the basic functions of the MCU for a terrestrial or a satellite network are similar.

In this description reference should be made to Figure 4 (T/N 33-01).

The tasks to be performed by a MCU are
i) Network access and interface
ii) Management of framing structure: multiplexing and demultiplexing
iii) Mixing of audio signals
iv) Processing of the subchannels
v) Analysis of control messages
vi) Routing of signals to audiographic teleconference terminals and other MCU's
vii) Handling of encrypted signals
viii) Terminal Interconnection
ix) Office Automation facility
x) Operator's console

A reservation could be part of the MCU or located elsewhere. It is presumed that reservation is performed through a so-called "Reservation Centre".

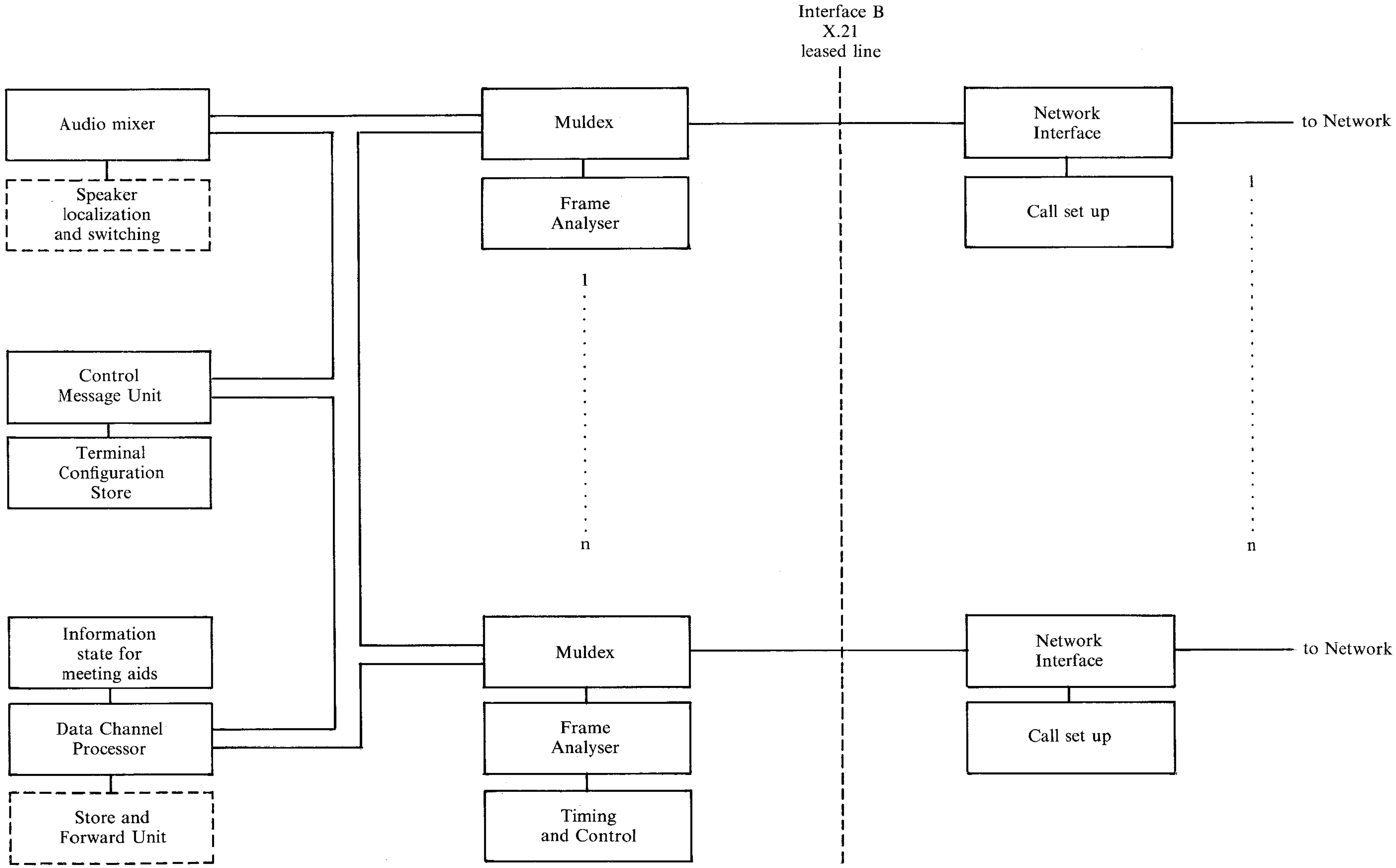


Figure 4 (T/N 33-01). MCU.

8.1.1. *Network Access*

When the links to the terminals are set up, the MCU receives and transmits 64 kbit/s streams through interface equipments.

Initially this interface will be X.21, leased line version.

The network interface units provide bit and byte timing derived from incoming bit streams and provide for each port a single duplex 64 kbit/s transmit and receive path to the network. When used with the ISDN these units will be responsible for call set up and answering.

8.1.2. *Management of framing structure*

Receive side

(a) For every port of the MCU Frame Alignment must be achieved for the incoming service channel in bit 8 of each octet sychronised to the single 64 kHz pilot clock. A loss of Frame Alignment should be detected and an indication about frame loss should be sent to the transmitting terminal using a spare bit within the Multiframe Alignment pattern.
(b) A mode switching procedure as described in section 5.4. will be performed.
(c) Signals from each port are demultiplexed into the audio components and the service channel. Depending upon the attribute value the service channel is subdivided to recover the message channel, telewriter channel and other information. Also depending upon the attribute value the appropriate audio coding mode is selected and the information at 8 kbit/s from meeting aids is recovered.

These tasks will be carried out by the Muldex and Frame Analyser.

Send side

The reverse operations are performed compared to the receive side.

8.1.3. *Mixing of the audio signals*

The audio mixer receives encoded speech from each muldex in the MCU. By means of an algorithm to be decided audio signals from various sources will be mixed and/or switched and routed as appropriate to other directions. The general principle of voice mixing is that each extremity should receive the audio from all other extremities except his own.

Particular points which require consideration in multiconferencing environments are:

Saturation of coders

Addition of room noises

Echo

This matter is for further study into different technologies of audiomixing/switching and may be resolved in the terminals and/or the MCU.

8.1.4. *Processing of the subchannels*

Data channel processor

The data channel processor mediates between requests for the data subchannel by the various transducers noted in paragraph 7.5. It is also responsible for the correct routing of information contained in the data subchannel to the various terminals and stores.

Telewriter

The telewriter signals from the terminals are broadcast to the other terminals through the MCU. A procedure to avoid collisions between messages is necessary. Details are given in Annexes 6 and 8.

Cursor

The cursor is used to point to parts of the screen in conjunction with Telewriter or SPTV pictures. The same comments apply for telewriting.

Facsimile

Transducers such as facsimile are designed to work on a "point-to-point" basis. To enable multipoint working of these devices a store and forward facility may be required. In operation, following a handshake procedure between each facsimile unit. Information is temporarily stored and then broadcast to other facsimile units connected as on a "point-to-point" basis. Alternatively, simpler protocols for facsimile may be devised. Details for group 3 multipoint procedures are given in Annexes 4 and 8.

If the handshake with a receiving equipment does not succeed after some period (to be specified) the transmission will start and the terminal with the disabled fax-equipment receives a message that a fax message was sent but that his terminal was not able to receive it.

SPTV

The exact protocol to be used for multipoint working is described in Annexes 5 and 8.

Speaker identification signal

For further study.

8.1.5. *Analysis and routing of messages*

For further study.

8.1.6. *Routing of signals to audiographic teleconference terminals and other MCU's*

For further study.

8.1.7. *Handling of encrypted signals*

For further study.

8.1.8. *Initialisation and terminal interconnection*

For further study.

8.1.9. *Operator's console*

For further study.

8.2. Configuration

The following two basic network configurations are likely for multipoint conferencing (see Figure 5 (T/N 33-01)).

The configuration of Figure 5a (T/N 33-01) is generally not suitable for a satellite connection as it requires a MCU in a node in the network. Satellite links could be employed instead of terrestrial links. Care should be taken not to involve more than one satellite link in unconducted meetings because of the introduced transmission delay. In conducted meetings and remote lecturing applications more satellite hops are allowed.

The configuration of Figure 5b (T/N 33-01) is suitable for satellite connections but less attractive for terrestrial connections.

Formation of meshes within this satellite network can lead to problems of addressing and stability; to avoid this problem the rule for MCU's within this configuration is that no MCU should retransmit signals back to the same satellite from which they were received.

As the employed error correction procedures of the message channel are not intended for broadcasting the satellite network has to be built up from point-to-point connections. This subject will be studied further.

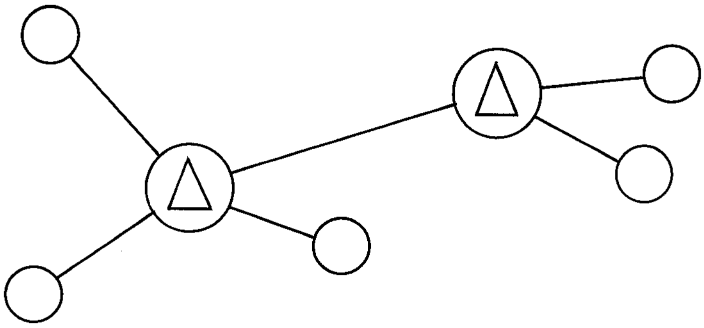
9. INTERWORKING

The following interworking possibilities with other services are envisaged:

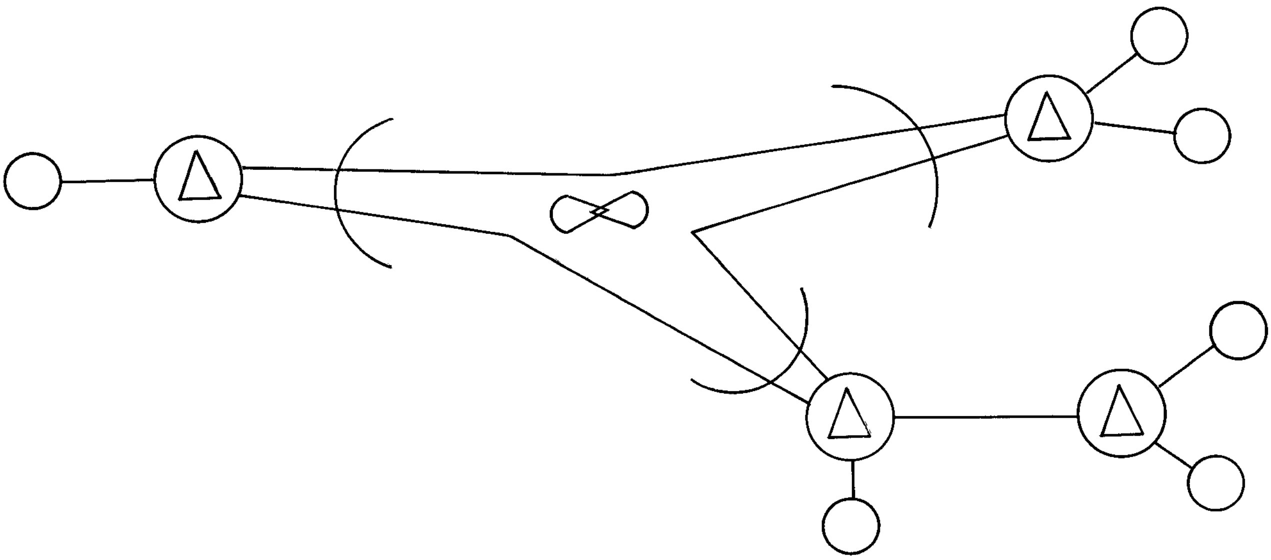
	Interworking between:	Level of interworking
(a) Direct interworking (terminal-terminal) possible *(Note 1)*	Audiographic teleconference and digital telephony A-law-G.711	A-law speech only
	Audiographic teleconference and digital telephony – wideband coded in 64 kbit/s - G.722	Wideband speech
	Audiographic teleconference and videophone – 2×64 kbit/s (to be defined in CEPT)	Speech and possible meeting aids
(b) Through time slot selection in a mux or other equipment *(Note 2)*	Audiographic teleconference and video – conference – 2 Mbit/s according to T/L 01-02	Speech only A-law or wide-band
	Audiographic teleconference and video – conference – 384 kbit/s 2nd generation to be defined	Wideband speech + meeting aids

Note 1. Via the initialization procedure described in section 5.4. the audioconference terminal adapts to the terminal on the other side of the line.

Note 2. The audio channel occupies one time slot in the videoconference frame.



(a) Terrestrial links.



(b) Mixed satellite and terrestrial links.

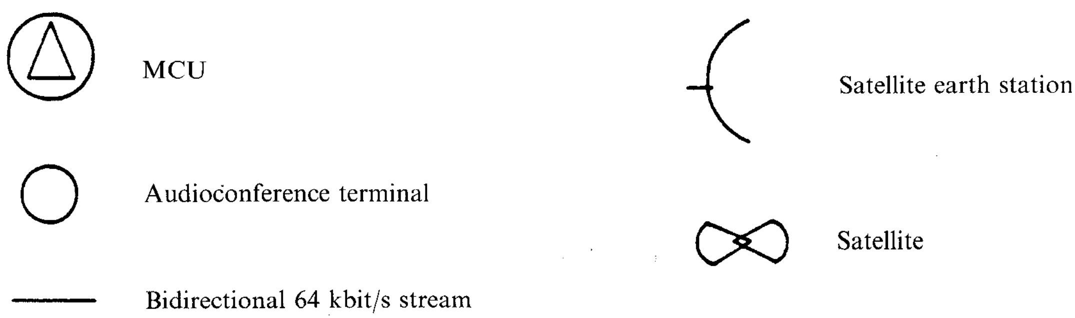


Figure 5 (T/N 33-01). Multipoint Network Configurations.

Annex 1

LIST OF DEFINITIONS AND ABBREVIATIONS

1. DEFINITIONS

1.1. Teleconferencing

The following definitions for a Teleconference service are given by CCITT.

1.1.1. The Teleconference service (TCS) provides the necessary arrangements for real time conferencing among single individuals or groups of individuals at two or more locations, by means of telecommunications networks.

The concept of conferencing implies that the exchange of speech signals is always provided for as a basic facility. The use of supplementary facilities, for the exchange of other signals than speech, is to be determined by the conference participants.

For the interconnection of terminal equipment at three or more locations, a specific interconnection facility is required namely the Multipoint Conference Control Unit MCU, to which all locations are connected individually.

The MCU provides proper distribution of the various signals among the connected locations and takes part in maintaining the proper procedures among the connected terminals.

1.1.2. TCS is a real-time service which can be divided according to the following categories:

(a) Audiographic Conference Service
A type of TCS in which audiosignals are exchanged together with non-voice information except moving video (data, text, graphic etc.). It may use existing protocols for non-voice informations or an integrated protocol.

(b) Videoconference Service
A type of TCS in which both audio and full motion video information can be exchanged together with optional non moving visual information, telematic information and signalling (speaker identification, floor request etc.).

N.B. A Telephone Multipoint conference may be considered as a Simple form of audiographic teleconference.

Other forms of Audiographic teleconference may imply loudspeaking terminals working full duplex of semi-duplex mode providing a considerably better sound quality than normal Telephone (they may even contain wideband speech coding). Supporting signalling like request for floor, grant request for floor and speaker identification may also be present.

1.2. Other definitions

Audiographic Teleconference Terminal: All equipment and accommodation to be connected to the network-interface (see Figure 1 (T/N 33-01)) which enables the user to conduct an audiographic teleconference.

Basic Audiographic Teleconference Terminal:
The minimum equipment-configuration which can be characterized as an "Audiographic Teleconference Terminal". The Basic Audiographic Teleconference Terminal (or Basic Terminal) configuration is indicated in chapter 7. and in particular in Figure 1 (T/N 33-01).

Chairman*
One who is chosen to preside over the audiographic teleconference in the management of the business of the meeting.

Conductor*
One who leads or guides the technical management of the audiographic teleconference.

Convenor*
One who summons participants to a meeting and makes all necessary prior arrangements.

Facsimile:
A means of transmitting and receiving fixed graphic material, photo, map, document etc. by means of a scanning process, the received image being reproduced on paper.

* *Note.* The conductor, convenor and chairman will generally be the same person.

Meeting aid:
A telematic equipment used during the audiographic teleconference.

Minimum Agreed Audiographic Teleconference Terminal:
To be decided.

Multipoint Control Unit (MCU):
A device in the network which enables more than two audiographic teleconference terminals to be interconnected.

Still Picture TV:
A means of transmitting and receiving non moving images derived from a television camera, the received image being displayed on a television screen.

Telewriter:
System used for hand written text in which writing movement at the transmitting end causes corresponding movement of a writing device at the receiving end.

2. TABLE OF ABBREVIATIONS

ACT	Audiographic Teleconference Terminal
BAS	Bitrate Allocation Signal
EVE	European Videoconference Experiment
FAX	Facsimile
ISDN	Integrated Services Digital Network
IDN	Integrated Digital Network
MCU	Multipoint Control Unit
MRP	Mouth Reference Point
MULDEX	Multiplexer/Demultiplexer
OSI	Open Systems Interconnection reference model
RC	Reservation Centre
SPTV	Still Picture Television

Annex 2

MESSAGE CHANNEL SPECIFICATION

This specification will be added later on.

Annex 3

FRAME STRUCTURE AT 64 KBIT/S FOR MULTIMEDIA APPLICATIONS

The framestructure to be used is composed of either
— 64 kbit/s (mode 0, mode 1) or
— 56 kbit/s + 8 kbit/s (mode 2) or
— 48 kbit/s + 16 kbit/s (mode 3)
based on CEPT Recommendation T/L 01-03. The initial mode (default) is assumed to be mode 2 i.e. 56 kbit/s + 8 kbit/s.

Annex 4

PROCEDURES FOR HANDLING OF FACSIMILE SIGNALS

This annex will contain a description from MIAC about an IPA- and MCU-procedure description of facsimile group 3.
Group 4 equipment is for further study.

Annex 5

PROCEDURES FOR HANDLING OF SPTV SIGNALS

For further study.

Annex 6

PROCEDURES FOR HANDLING OF TELEWRITER SIGNALS

For further study.

Annex 7

PROCEDURES FOR HANDLING OF TELEMATIC SIGNALS

For further study.

Annex 8

PROCEDURES AND PROTOCOLS TO BE PERFORMED BY THE MCU

For further study.