Recommendation T/TR 01-02 E (Nice 1985)
concerning the international digital videoconference service

Recommendation proposed by Working Group T/WG 12 "Transmission" (TR)

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

"The European Conference of Posts and Telecommunications Administrations,

Considering

that there is an increasing demand for videoconference services, and especially for international videoconferencing; such services basic audiovisual facilities and a variety of optional facilities according to customers' requirements, and

Recognising

the need for interconnectability between videoconference rooms in different countries and in different customers' premises, preserving not only the correct electrical operation but also satisfactory acoustic and vision conditions.

Recommends

to the members of the CEPT that they should implement an international videoconference service as described and defined in the attachment hereto."

Contents

1. SCOPE

1.1. Introduction

An increasing demand for videoconferencing service is now in evidence, resulting from technological progress which makes possible satisfactory performance in a commercially viable way. 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 videoconference service.

Reference is made in this Recommendation to compatibility with other related services, for example audioconferencing 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 bit rate options.

1.2. Definition of videoconferencing

"The videoconference service is a service for full-duplex bidirectional real-time audiovisual communication between groups of users in two or more separate locations, combining a good audio facility with moving pictures of those participants, and with optional auxiliary telematic facilities."

1.3. Scope of this document

This document presents the Recommendation, including definitions, description of the service 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.

Annex 4 to this document provides supporting information which can be used by bodies setting up videoconferencing services.

2. DEFINITIONS

2.1. Digital videoconference system

A complete set of equipment by means of which a videoconference service can be provided, comprising terminal equipment, coding equipment, and digital network equipment with interconnecting transmission media. For a multipoint service, multipoint service centres are located at network nodes and connected to terminals via several point-to-point connections.

2.2. Videoconference terminal or terminal

The users' equipment and accommodation for a videoconference system at each end of the connection including audio, video, and optional ancillary equipment. It must deliver signals to the coding equipment according to the interface specifications in Sections 5. and 6.

2.3. Minimum terminal

That part of the terminal which contains only audio and video equipment and conforms to the specifications in Sections 5.2., 5.3., 6.1., 6.2.

2.4. Terminal options

Those aspects of a terminal, not falling within the minimum agreed terminal, which a user wishes to have incorporated in the terminal in order to improve the service. The options when present, must conform to the specifications in Sections 5.4., 5.5., 6.3.

2.5. Video subsystem

The part of the terminal which contains video equipment including cameras, monitors, control and switching of video, video mixers, etc. In the following, the term monitor applies to both CRT monitors and video projectors.

2.6. Principal monitor

That (those) monitor(s) which normally display the incoming video signal.

2.7. **Local view monitor**

A monitor which normally displays the outgoing video signal.

2.8. **Conference camera**

That (those) camera(s) which form images of the participants at a videoconference.

2.9. **Document camera**

A fixed camera disposed to view documents or objects placed on a horizontal surface.

2.10. **Auxiliary camera**

Any other camera.

2.11. **Split-screen transmission**

Two video sources are combined into a single video signal according to CCITT Recommendation H.100; at the receiving terminal the images may be split and displayed on adjacent monitors.

2.12. **Audio subsystem**

The part of the terminal which contains the audio equipment including microphones, loudspeakers, amplifiers, audio mixers, echo suppressors/cancellers, etc.

2.13. **Voice-switched video system**

The selection between video signals from two or more conference cameras is determined by the level from corresponding microphones.

2.14. **Voice controlled audio system**

To prevent echo between remote locations either a voice switch (echo suppressor) switches attenuation into one or both directions of audio transmission, or an echo canceller is employed to cancel the coupling between loudspeaker and microphone.

2.15. **Ancillary subsystem**

Any equipment or facility for communication between participants in a videoconference, the signals for which are conveyed to or from the terminal together with the primary video and audio paths. This includes for instance facsimile, telewriting or electronic blackboard, still picture television (SPTV), etc.

2.16. **Codec information and control**

Digital information issued from the terminal to effect control of the video and/or transmission coding in the codec; information injected into the digital stream to ensure complementary operation of coders and decoders.

2.17. **User-to-network signalling channel**

A channel separate from the user information to convey the necessary signalling for dialogue with the network, as for call set-up or clear down.

2.18. **User-to-user signalling channel**

A channel to convey user-to-user information for service management.

2.19. **Maintenance and monitoring signals**

Necessary signals on a local or remote basis which provide monitoring of the terminal equipment.

2.20. **Alignment procedures**

A set of measurements which specify the calibration of audio and video levels in terminals connected to the videoconference network.

2.21. Codec or 2 Mbit/s codec or videoconference codec

A unit or a set of equipement interfacing between the terminal and the digital network. In particular it provides analogue to digital conversion, compression and coding of information and the reverse operation. It may comprise the video codec, the audio codec, the interfaces to digital equipment in the terminal, the transmission muldex and the network terminating equipment. In some network configurations, the videoconference codec may be located in a network node *or in the international gateway* and shared by several terminals: in such cases the connection between a terminal and the codec is made via a *wideband connection.*

2.22. Wideband connection

An analogue connection, or a digital connection using a high bit rate codec, which carries the signals between a terminal and the videoconference codec through a wideband network.

2.23. Video codec

A part of the videoconference codec having interfaces with both the analogue video signals of the terminal and digital streams to and from the transmission muldex. It carries out redundancy reduction, compression and coding of video information.

2.24. Audio codec

A part of the videoconference codec or a separate unit having interfaces with the analogue audio signals of the terminal and digital streams to and from the transmission muldex.

2.25. Transmission muldex

A part of the videoconference codec or a separate unit which composes an outgoing 2,048 kbit/s digital stream, according to the interconnection specifications (Section 8.) from digital video, audio and other signals, and splits an incoming 2,048 kbit/s stream into these same components.

2.26. Network interface unit

A part of the videoconference codec, or a separate unit, which ensures the electrical interface between the transmission muldex and the network access.

3. SERVICE AND FACILITIES

This section defines the videoconference service from the point of view of customers and marketing organisation. 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 moving pictures of participants, and optional auxiliary facilities such as still pictures, facsimile, telewriting, character or graphic format, etc. The service is applicable to companies' private videoconference rooms, as well as to public-access videoconference rooms for hire on an occasional basis; it is applicable to a variety of types of videoconference terminal, including multi-purpose committee rooms used only part-time for videoconferencing as well as dedicated studios; it is applicable also to "mobile" or "site-transferable" videoconference facilities.

The service is bidirectional via public telecommunication networks, and provides for interconnection of 2 or more terminals on an equal basis (the possibility of adding more terminals to a call, perhaps on an audio-only basis, is under study). Users may require the implementation of certain measures, including encryption, to ensure confidentiality. In the case of switched calls, the user must be assured of availability, for example by advance reservation, and may also require an assurance of confidentiality.

3.2. Basic facilities

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

3.2.1. *Audio system*

Provided that the videoconference room has adequate acoustic damping (e.g., by means of drapes and carpet) participants may listen and speak simultaneously; speech transmission is of 0.3-3.4 kHz bandwidth, though the perceived audio quality is considerably better than telephony because better transducers are used. In anticipation of standardisation of a wider bandwidth transmission system, the terminal audio equipment is specified to have a minimum bandwidth of 8 kHz (section 5.3.).

Microphones are fixed in position; loudspeakers are associated with the picture screens.

3.2.2. *Vision system*

There are a variety of "basic" configurations, according to user needs; as a minimum, one screen is provided for the display of incoming video signals (in colour), and one colour camera mounted closely above this screen televises a view of all or some of the participants; further cameras may be provided for a variety of purposes (e.g., more participants, display of objects, flip chart, documents, etc.)—in this case local control is provided for the selection of a single outgoing video signal. In general, participants will sit at a preferred distance of six times the picture height from the picture screen; the choice of picture size will be a matter of customer choice, a likely choice being approximately 600 mm diagonal (360 mm picture height, 2 metre viewing distance).

3.3. Optional facilities available

Here follows a list of the auxiliary facility options which are available in the videoconference system of this Recommendation.

3.3.1. *Data transmission*

Standardised data ports can be provided operating at 64 kbit/s. These enable data transmission to be carried out during the call. Data transmission service at 32 kbit/s is possible via the codec interface (see Section 8.2.) by bilateral agreement.

3.3.2. *Split-screen*

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

3.3.3. *Still picture TV system*

Whereas for pictures of participants it is necessary to preserve the motions accurately, for other objects (charts/diagrams, documents, solid objects) it is often more important to preserve the *clarity* of the image: the system provides for the transmission of such images, either very rapidly by temporary displacement of the participant-picture transmission (this is known as "graphics mode") or simultaneously with the participant picture by abstraction of a 64 kbit/s channel in either case, the system may provide for indefinite retention and display of the still picture at the remote end.

3.3.4. *Facsimile*

By use of a 32 or 64 kbit/s data transmission channel as in 3.2.3. above, facsimile documents may be transmitted at any time on a point to point call, using a digital facsimile machine; the problem of achieving this under multipoint conditions is far more complex, and still under study.

3.3.5. *Improved audio systems*

A customer may choose to implement an audio system having a bandwidth of 7 kHz: the system cannot provide for interconnection of 7 kHz and 3.1 kHz terminals, though it does transmit an indication as to which is in use. The question of stereophonic sound requires further study.

3.3.6. *Terminal equipment for multipoint operation*

For further study.

3.3.7. *Encryption*

End-to-end encryption of the digital signals can be provided by bilateral agreement; an indication is given when the encryption is activated.

3.4. Temporary use of facilities

It should be noted that some of the basic and optional facilities may not be required for the full duration of a call: this applies particularly to aids such as telewriting, facsimile, and still pictures used to display documents or objects. When required, these facilities are carried on a temporary subchannel, effectively reducing the video information rate: *the quality of the static video picture is not affected and the small reduction in movement rendition is not normally perceptible.*

3.5. Terminals

In general, it is not necessary to standardise completely all aspects of videoconference 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.

General information is given in section 5.1. as to conditions which will ensure the quality of the service offered as well as the comfort and convenience of participants.

The controls to be operated by users shall be kept to a minimum.

Normally one person 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 chairman.

3.6. Performance

3.6.1. *Audio quality*

Under normal conditions the basic service provides an audio channel of 0.3-3.4 kHz bandwidth—this is equivalent to normal telephony, but is perceived as somewhat better due to the use of microphones and loudspeakers instead of the conventional telephone handset.

3.6.2. *Moving picture TV quality*

Under normal conditions the static picture quality of participants is comparable with that obtained from a good quality video cassette recorder, though falling slightly short of the resolution achieved from broadcast quality pictures under good reception conditions. Movement rendition is compromised to some extent: the relatively minor movements of participants at a videoconference are protrayed reasonably faithfully. Movements affecting a substantial fraction of the picture area (e.g., panning, zooming) may cause momentary picture break-up.

3.6.3. *Still picture TV quality*

For further study.

3.6.4. *Performance under conditions of degraded transmission*

Normal conditions should obtain for at least 92% of a 2-hour period. Under worse case transmission conditions, which should not exceed 0.2% of a 2-hour period, the picture will be badly broken up and sometimes frozen, while the sound will be heavily distorted. During the remaining time, less than 8%, the picture may suffer occasional horizontal streaks, taking up to 2 seconds to disappear, while the sound degradation may be perceived as several clicks per second of varying intensity.

According to tighter transmission constraints now under study, these degradations will not be at an annoying level (several simultaneous streaks and many clicks per second) for more than 1% of time.

3.7. Network access

The videoconference service can be provided from the following types of access:

(a) 2 Mbit/s access, all or part of the digital path being devoted to the service;

(b) analogue video access or high-speed digital access: in these cases the interface conditions are referred to those for the terminal/codec interface—see section 7.

3.8. Availability

The service is available at predetermined times by reservation only; reservations may be made up to 90 days in advance, but a minimum of hours notice (24) is normally required. No visual or audible indications corresponding to approaching call maturity are given and disconnection is made at the appropriate time without prior indication from the network.

4. FUNCTIONAL DESCRIPTION

4.1. Hierarchical model applied to videoconference

(For further study.)

4.2. Terminal structure

Figure 1 gives a functional block diagram of a complete videoconference system including terminal and transmission equipment. The purpose of this diagram is to explain the functional relationship between the various subsystems. It is not intended that the complete equipment should be mandatory, but rather than, where provided, the subsystems should be related in the given way and conform to the specifications provided in this recommendation. The explanation of Figure 1 follows:

The terminal equipment, located in the videoconference room, consists of the following subsystems:

i) Network subsystem: an equipment carrying out local surveillance and control functions, together with interchange of information with the network as required for signalling, multipoint control, etc.

ii) Dialogue subsystem: a terminal set (e.g. keyboard and visual display) allowing user-to-user exchange of information. This unit can also control the other units of the terminal if desired. It can also be used for user-to-network signalling.

iii) Audio subsystem: microphones, loudspeakers, audio switching and mixing; this subsystem will provide in both directions a sound signal and the information necessary to the proper operation of an audio codec.

iv) Ancillary subsystem: an optional subsystem including such facilities as facsimile, electronic blackboard, and so on, together with the means for presenting them at the transmission muldex.

v) Video subsystem: cameras, monitors, video switch, split-screen unit; this subsystem will provide in both directions a video signal and the information necessary for the proper operation of a video codec.

The terminal interface signals are those specified signals passing in both directions between the terminal and the videoconference codec.

The videoconference codec gives access to a digital network: the encoder takes in the various signals coming from the terminal, encodes the analogue signals into digital, and multiplexes all the digital signals into a single digital stream passing into the network; the decoder performs the reverse operation. The videoconference codec comprises:

i) Video codec.

ii) Audio codec.

iii) Intermediate multiplex, which provides for good formatting with the service and ancillary signals coming from the terminal for easy multiplexing into the PCM frame structure.

iv) The transmission muldex, which multiplexes the different encoded information into a digital stream in conformity with CCITT Recommendations.

The following notes apply to Figure 1:

(a) The five columns correspond to five kinds of information flows.

(b) The diagram reflects the functional separation into the hierarchical layers.

(c) In the present system the audio codec is a 64 kbit/s PCM A-Law codec and occupies a whole 64 kbit/s time slot. Consequently the user-to-user signalling (flow ii) is conveyed separately through another time slot.

(d) In future systems using an audio codec with wideband coding at 64 kbit/s or less, user-to-user signalling will be multiplex in the 64 kbit/s channel via an "intermediate multiplex unit" represented in the diagram at the transport layer. This will form a single 64 kbit/s channel compatible with the audioconference service.

(e) The videoconference codec or 2 Mbit/s codec is represented by the box in dotted lines and may combine several functions in the same piece of equipment. Other arrangements may involve separate elements.

(f) In cases where the codec is not colocated with the terminals, signals marked with an asterisk (★) should be transmitted from the terminal to the codec via a "local loop multiplex", which is outside the scope of this Recommendation.

5. TERMINAL EQUIPMENT AND ACCOMMODATION

5.1. General disposition of equipment and facilities

It is not considered necessary to standardise every aspect of a videoconference terminal: correct international interworking is assured by adherence to the electrical interface specifications and observance of the alignment procedures for audio and visual signals. Considerable scope is left to designers and equipment suppliers in the matters of the style and disposition of videoconference room and equipment and the details of the facilities in their local aspects. In this section only minimum requirements are stated for the achievement of good operating performance.

We may identify three classes of videoconference terminals:
— dedicated studio;
— general-purpose committee room incorporating videoconference facilities;
— table-top videoconference unit, similar to a work station or viewphone.

The remarks in the following subparagraphs are couched in terms of the intermediate type: interpretation in terms of the other two types should pose little difficulty.

(a) Videoconference room

Figure 3 gives an indication of the general disposition of a videoconference room.

(b) The general decoration of walls, floor and ceiling should avoid black, brilliant white or saturated colours. All finishes should be free from patterns, stripes, etc., which may cause difficult working conditions for cameras, codecs, etc. For the background behind the conferees' desk (the camera/s field of view) pastel colours are recommended (blue or similar). Alternatively a light brown, beige or cream may be considered but darker shades of brown should be avoided. The background colour should produce approximately 0.35 V video level (luminance) when the cameras have been adjusted as in section 6.1.1.1. Floor coverings, wall coverings/decoration and ceilings should have matt-type finishes and good reflecting properties to aid the general ambient level of illumination within the room.

(c) Lighting

The lighting level should be high enough for cameras to provide adequate performance (see sections 5.2.1., 5.2.5. and 6.1.1.) but should not cause discomfort to the conferees—diffused or reflected lighting arrangements are recommended. The general ambient lighting should be as uniform as possible and also give suitable modelling to conferees (e.g. faces); where necessary additional modelling lighting should be added. The lighting plan adopted should minimise the amount of illumination that falls on the monitor screens.

The colour spectrum of the lighting units must provide accurate rendition of colours. Fluorescent lamps (such as the Thorn EMI "Kolorite" or similar) can meet the requirements.

(d) Ambient noise and acoustic insulation

Assuming that the noise N in the premises around the conference rooms is typically around 40 dB (A), the speech level transmitted by the room walls (and doors) should be less than 25 dB (A), in order to assure a good privacy. It follows that an insulation of about 45 dB ($I_a \leqslant 50$ dB) is required, assuming that the average talking and listening levels into the room are around 70 dB (A). Any air conditioning or ancillary equipment within the room (e.g. facsimile) should be very quiet and conform with the audio specifications under 6.2.

(e) Cameras

Cameras should be positioned to minimise the eye-contact angle for conferees viewing the principal monitor(s). Split-screen arrangements should use centrally located, crossfired cameras to minimise eye contact angles. In split-screen and multi-camera systems where colour is employed care must be taken with matching of colour performance. Camera(s) horizontal axis should be level with the median eye height, approximately 1,290 mm above floor level.

(f) Monitors

Principal and auxiliary monitors reproduce the necessary pictures for a videoconference. Pictures representing the conferees should be near to life size images for the viewing distance (see (g)).

The height of the centre of the principal monitor(s) shall always be not higher than the median eye height (see (h)). Where more than one colour monitor is employed care must be taken with matching of colour performance. Where split-screen presentation is on two monitors close together image continuity must be maintained.

(g) Conference table and chairs

The viewing distance from the conferees seating position to the principal monitor(s) should be 4-6 (preferably 6) times pictures height (i.e. actual screen height).

The table should be symmetrical and preferably curved or segmented to provide visual contact between conferees. The height of the table should be approximately 720 mm. The table-top should be large enough to provide a comfortable working area, at least 600 mm × 600 mm per conferee. The table surface shall be matt to avoid glare, but of light colour to provide some lighting reflection to aid modelling of conferees' faces.

Chair heights should be adjustable, especially if split-screen equipment is used. Where chairs without arms are used a width of greater than 600 mm per conferee will be required.

(h) Viewing angle

The viewing angle to principal monitor(s) should not exceed 60° from the normal. The vertical viewing angle for the principal monitors with respect to the eye axis shall be between 0° and $-10°$ (see Figure 3b).

(i) Additional participants

Where space is available in the room then more conferees could be accommodated by use of a second row of chairs interleaved with the front row. Alternatively, other participants can be seated outside the conference camera view, with additional monitor(s) and audio equipment and an optional auxiliary conference camera.

(j) Microphones and loudspeakers

They should be as discret as possible to reduce the "studio atmosphere".

(k) Directions for use

They should be clearly explained for each facility of the room. Use of pictograms should be made.

(l) Help

A conventional telephone should be provided within the conference room and a number given to contact in case of failure or need of information.

5.2. Video subsystem

5.2.1. *Cameras*

All cameras within one terminal shall be synchronous and shall conform to the following.

625 lines monochrome or colour, 50 fields/s, 2:1 interlace; 5 or 5.5 MHz nominal bandwidth; synchronisation waveform to 625-line systems in CCIR Recommendation 470-2 and report 624-2; 1.0 volt nominal peak to peak at white of EIA test card (see 6.1.1.1.) in 75 ohms unbalanced; line frequency 15,625 Hz + 3 Hz.

Quality requirements for cameras:

— Signal to noise ratio:

The signal to noise ratio (SNR) of the cameras shall be not less than 45 dB (weighted). This number depends on the general quality of the camera, the aperture of the iris, the lighting conditions and should be measured at setting quoted in the camera manufacturer's specification. Weighting should be to the unified weighting curve of CCIR Recommendation 567.

— Resolution of cameras:

The horizontal resolution of conference cameras measured in the centre should be at least 30% amplitude at 300 lines (150 line pairs) per picture height and for display cameras 60% amplitude at 400 lines (200 line pairs) per picture height.

5.2.2. *Monitors*

The monitors shall conform to the following:

625 lines monochrome or colour, 50 fields/s, 2:1 interlace; 5.5 MHz nominal bandwidth; synchronisation waveform to 625-line systems in CCIR Recommendation 470-2 and report 624-2; 1.0 volt nominal peak to peak in 75 ohms unbalanced; line frequency 15,625 Hz + 3 Hz.

Where monitors are used for the display of graphical material (diagrams, documents, etc.) a minimum screen resolution of 400 lines per picture height is required.

5.2.3. *Split screen unit*

When present, this should be to Recommendation H.100. A bidirectional control should be available to and from the videoconference codec. It should enable the display of the split-screen image on two monitors.

5.2.4. *Auxiliary video sources*

Output signals as in section 5.2.1.; commercial video cassette recorders are unlikely to meet this specification without the use of a time base corrector.

5.2.5. *Decor/Lighting*

The minimum lighting level shall be approximately 500 Lux on the desk in front of the conferees.

5.2.6. *Other configurations*

Alternative implementations may involve the use of two or more conference cameras with switching (switching characteristics to be specified), the display of different video signals in a multi-screen configuration, and the use of colour equipments to handle monochrome signals.

5.3. **Audio subsystem**

5.3.1. *Microphones and sound system*

Microphones can be provided on an individual basis or shared between 2 or 3 conferees. They should be of the directive type if a voice-switched video system is in use.

The positioning and relationships between microphones and loudspeakers should be based on Supplement No. 25 of CCITT Recommendation G.172 (guidelines for placement of microphones and loudspeakers in conference rooms). In particular, the maximum distance between users and microphones should be computed taking into account the acoustic noise level (see 5.3.4.) and the number of microphones simultaneously opened. For example with an acoustic noise of 40 dBA and 6 microphones, the maximum distance is 60 cm.

The 3 dB bandwidth points of the sound system, excluding the echo attenuation system, should be 100 Hz-8,000 Hz.

5.3.2. *Loudspeakers*

A single loudspeakers should be placed centrally but if two loudspeakers are used they should be placed symmetrically and connected in phase minimum; they should have a bandwidth of 100 Hz-8,000 Hz ($\pm$ 5 dB). The diffusion cone of the loudspeaker(s) should be oriented towards the conferees for good intelligibility (see section 6.2.).

5.3.3. *Echo*

The echo return loss between loudspeakers and the microphones should be higher than 6 dB over a wide frequency range including 100 Hz-8,000 Hz (without any electronic equipment for echo attenuation).

The return signal is a function of the acoustic conditions of the room (direct and indirect acoustic path from loudspeakers to microphones). The 6 dB margin can be obtained by a proper acoustic treatment of the room combined with the respective positions of microphones and loudspeakers.

The reverberation time (when the measured level has fallen by 60 dB) should be between 0.2 and 0.4 second for a frequency range of 100 Hz-8,000 Hz.

5.4. **Other facilities**

5.4.1. *Facsimile equipment*

This shall be to CCITT Recommendation for group III (T.4) or IV (T.5).

The interfacing with the codec is under study. (If the facsimile machine is to be in the videoconference room itself, it must be very quiet (see 5.1. (d).)

5.4.2. *Telewriting and electronic blackboard*

Under study.

5.4.3. *Still picture TV*

Under study.

5.5. **Customer information and control**

The details of information and control facilities provided at a terminal are left to each Administration (see also Annex 4).

6. TERMINAL ALIGNMENT PROCEDURES

6.1. **Video alignment**

6.1.1. *Cameras*

Cameras must conform to CCIR Recommendation 470-2 and report 624-2. Line period, field period, line and field blanking intervals must be checked. In case of a colour camera, the colour filter of the camera must be adjusted for the colour temperature of the lighting of the room.

6.1.1.1. Camera setting

For each camera, measurements should be made with a grey scale pattern (e.g. EIA logarithmic reflectance chart ≡ BBC test chart No. 57) with 9 levels from black 3% to white 60%, used in normal lighting conditions. The test chart should be placed at each conferees position and the following should be checked:

— Visually, all grey levels should be visible.

— Electronically, i.e., on a waveform monitor or oscilloscope, one should measure:

-300 mV ± 9 mV for sync level	Tolerances with respect to blanking level.
0 mV for blanking level	
50 mV ± 2 mV for reference black	
700 mV ± 20 mV for reference white	

— Weighted noise measured on blanking level and on white level should be less than -45 dB with respect to 700 mV.

6.1.1.2. Iris aperture

Working aperture of the iris in normal lighting conditions should be not less than F 2.8 to provide some depth of field. Gain correction (maximum 6 dB) can be used. Manual and not automatic iris aperture is recommended, because of the effect of continual small variations on the video coding efficiency.

6.1.1.3. Colorimetry

— White balance is made on the white of the EIA test chart.

— Black balance (if present on the camera) is made on the black 3% of the chart.

A test card with flesh tones (such as BBC test chart No. 61) and a colour chart (such as the Macbeth colour checker) should be used at each conferee's position and subjective assessment made of the accuracy of flesh tones and the range of colours reproduced on the monitors (adjusted as in 6.1.2.2.).

6.1.2. *Monitors*

6.1.2.1. Monochrome

The brightness and contrast controls should be adjusted using a picture line-up generator. Linearity and amplitude of scanning should be checked using electronically generated test cards and cross-hatch generators. (*Note:* picture line-up, test card and cross-hatch generators are standard equipments as used in the broadcast television field.)

6.1.2.2. Colour

As with monochrome monitors, picture line-up, test card and cross-hatch generators should be used for adjustment, the cross-hatch generator also being suitable for convergence adjustment where provided.

An electronically generated staircase (grey scale) should be used to adjust the "grey scale" tracking of the monitor to ensure good monochrome performance. The balance of the three RGB channels should be controlled by displaying a uniform grey picture electronically generated (e.g. 350 mV). No colour dominant should be visible.

Electronically generated colour bars should be used for colour performance checking in conjunction with a colour analysis where required.

6.1.2.3. A final overall system performance check should be made by placing a colour bar test card at each conferees position and observing the resultant picture on the local principal monitors.

6.2. **Audio alignment**

6.2.1. *Use of a calibrated sound source*

The sound source to be used in the measurements for the audio alignment of the rooms consists of two parts:

(i) Noise source

Although it is acknowledged from CCITT studies that an artificial mouth is the best way to simulate human speech, the following arrangements are taken for reasons of practicality and economy:

The noise source consists of a white noise generator on the minimum bandwidth of 50 Hz-10 kHz followed by a suitable filter to limit the bandwidth of the transmitted noise. The filter is specified as either:

— a filter corresponding to CCITT G.227 which simulates the vocal electric currents to be transmitted; or:

— a filter which produces a flat (within 3 dB) noise response within the band 250-3,000 Hz rolling off at 48 dB/octave outside these limits.

In the case of a wider transmission bandwidth (e.g. 0-7 kHz), other filters could be specified.

(ii) An audio amplifier and associated loudspeaker able to deliver:
— Sound pressure level of at least 100 dBSPL at 150 mm of the loudspeaker in the axis of the loudspeaker.
— The acoustic properties must be near the average human mouth (as to the law of decreasing acoustic pressure in the axis emission and the law of directivity).
— The source must be stable and reproducible.

Note the loudspeaker must be a single small loudspeaker (diameter lower than 15 cm) conforming to DIN 45,500.

6.2.2. *Methods of measurement*

The units to be used in the measurements must be:

(i) Electrical measurements:
— Line noise measurements: dBm using a simple first order bandpass filter with 3 dB points at 150 and 4,000 Hz.
— Line level measurements: unweighted dBm.

The electrical measurements will be made on the analogue line-in and line-out ports of the studio before the anti-aliasing filter in the audio codec.

(ii) Acoustic measurements:
— Calibration of the sound source: linear scale i.e. dBSPL.
— Acoustic signals produced by the loudspeakers of the room: "A" weighting scale.

In all the acoustical measurements, the position of the sound source and the sound pressure level meter will be defined with respect to an optical reference point (ORP), defined as a point located 1.20 m from floor level and 15 cm to the rear of the working edge of the conference desk and on the centre line of each conferee's position.

6.2.3. *Acoustic noise level in the room*

Measured at the ORP at the central position of the table, the acoustic noise level should conform to supplement 25 to CCITT Recommendation G.172 for ambient noise. For example a videoconference room for six people would correspond to a medium size conference room for 20 people and should have a noise level lower than 40 dBA.

6.2.4. *Send side alignment*

The sound source is calibrated (CSS) to provide 90 dBSPL linear at 15 cm from the loudspeaker (see Figure 2a).

The CSS is placed with respect to the ORP as shown in Figure 2b on the centre line of each conferee's position. All microphones being opened, the microphone gain control must be adjusted to achieve a level to the send line of −13 dBm (+ 2 dB) by averaging with respect to all positions.

6.2.5. *Receive side alignment*

The band limited electrical noise source is adjusted for an output of −13 dBm into 600 ohms and connected to the line in port of the studio. The sound pressure level meter is positioned at the centre ORP as shown in Figure 2c. The studio power amplifier must be adjusted for the required listening level which will be typically in the range of 64 to 75 dBA, see CCITT Recommendation G.172, Annex 1.

6.2.6. *Electrical noise (studio borne)*

The measurement of the electrical noise emitted by the room is first made with microphones off. The level to the send line must be lower than −53 dBm (with first order bandpass filter specified in 6.2.2. (i)) referred to a −13 dBm point (i.e. a 40 dB SNR).

The second measurement is made with open microphones. The level to the send line must be lower than −38 dBm (with first order bandpass filter specified in 6.2.2. (i)) referred to a −13 dBm point (i.e. a 25 dB SNR).

6.2.7. *Acoustic coupling*

The acoustic coupling requirements can be satisfied by the following means:

i) Acoustic treatment of the room.
ii) Relative disposition of microphones and loudspeakers.
iii) Echo cancellation.
iv) Echo suppression.

A minimum 6 dB margin should be ensured by the design of the studio (e.g. by action on (i) or (ii).

With the band-limited electrical noise source connected to the line-in port and adjusted for an output of −13 dBm into 600 ohms, the level to the send line must be lower than −45 dBm with the first order bandpass filter specified in 6.2.2. (i).

7. CODING EQUIPMENT AND INTERFACES

7.1. Video codec

The video codec equipment must conform to CCITT Recommendation H.120: "Videoconferencing using primary digital group transmission." Part 1: "A codec for 625 lines, 50 field/s. and 2,048 kbit/s, transmission for intra regional use and capable of interworking with the codec of Part 2a".

The video interface is defined as in sections 5.2. and 6.1. The multiplex structure is defined in section 8.

When physically present the interface between the video codec and the transmission muldex should be to CCITT Recommendation X.21.

7.2. Audio codec

The audio codec is to CCITT Recommendation G.711, A-law. The standardisation of codecs for higher audio bandwidths is for further study.

The audio interface with the terminal is defined in sections 5.3. and 6.2.

When present, the interface between the audio codec and the transmission muldex is to CCITT Recommendation G.703.1 or X.21 for 64 kbit/s transmission.

7.3. Other codecs

7.3.1. *Facsimile*

The facsimile codec is to CCITT Recommendation T.4 (group III), T.5 (group IV).

7.3.2. *Telewriting, electronic blackboard*

For further study.

7.3.3. *Still picture TV*

For further study.

7.3.4. *Message channel specification*

For further study.

7.4. Local network transmission

Referring again to Figure 1, it may arise under some circumstances that the 2 Mbit/s codec (outlined with a dashed line) may not be colocated with the videoconference terminal: it may be in another part of the customer's building, or possibly even at another site to enable it to be shared with other customers. In either case, a "local loop" is involved, capable of transmitting all the signals indicated with an asterisk in Figure 1. This Recommendation is not concerned with the method of implementing the local loop, which could be analogue or digital, with the various signals multiplexed or carried separately. However it is essential that the signals at both ends of the local loop should conform to the terminal/interface specifications of this Recommendation, sections 5. and 6.

8. MULTIPLEX STRUCTURE AND TRANSMISSION

8.1. General description of possible transmission and networking arrangements

For international connections, reference is made to CCITT Recommendation H.110 "Hypothetical reference connections for videoconferencing using primary digital group transmission".

(a) All calls are symmetrically bidirectional.

(b) All calls are "semi permanent", reserved hours to weeks in advance.

(c) Terrestrial intra-regional connections are made between a limited number of international "gateways", the audio visual transmission occupying n $\times$ 384 kbit/s within a 2 Mbit/s frame structure.

(d) Intra-regional connections by satellite may involve gateway earth stations and/or earth stations on customers' premises. Transmission may be within a primary 2 Mbit/s frame structure, or at a lower bit rate of 2 or 3 $\times$ 384 kbit/s by extraction of the videoconference data from the terrestrially-conveyed 2 Mbit/s frame.

(e) Inter-regional connections involve the interconnection of a 2 Mbit/s path to Recommendation G.732 to a 1.5 Mbit/s path to Recommendation G.733 by means of a remultiplex unit. The latter is a 2-port device, one port being to CCITT Recommendation G.732 at 2048 kbit/s, the other to Recommendation G.733 at 1544 kbit/s. Information digits entering the device in time-slots 1-15 and 17-25 of the 2 Mbit/s signal are transferred to the 1.5 Mbit/s frame structure. A signal from the remultiplexer to the 2 Mbit/s codec at the start of the call ensures that the latter transmits no information in the remaining time-slots 26-31. Similarly in the reverse direction, information digits entering the device in the 1.5 Mbit/s frame are transferred to TS1-15 and 17-25 of the 2 Mbit/s frame structure for onward transmission.

(f) It is the responsibility of telecommunication administrations to provide the national transport for the videoconference service between the customer's terminal and the international gateway.

8.2. Transmission at n × 384 kbit/s in the primary rate access

8.2.1. *Frame structure*

The frame structure is described in CCITT Recommendation H.130: "Frame structure for use in the international interconnection of digital codecs for videoconferencing or visual telephony." Part 1: "Characteristics of a 2,048 kbit/s frame structure for use in codecs described in part 1 of Recommendation H.120."

The time slot (TS) allocation is as follows:

TS0	contains frame alignment, network alarms;
TS1	contains the audio signal coded at 64 kbit/s;
TS2 odd	contains end-to-end service information;
TS16	contains network signalling or video;
TS17, 18	contains data at 64 kbit/s or video;
TS2 even	contains data at 32 kbit/s or video;
Other TSs	contain video.

The service time slot provides among other information the possibility to signal the use of lower bit rates than 1,920 kbit/s, within the primary rate channel. They correspond to the hierarchy defined by CCITT for videoconferencing transmission at n × 384 kbit/s. CCITT Recommendation H.130 provides a method for signalling and transmitting at 4 and 5 × 384 kbit/s. Transmission at 2 and 3 × 384 kbit/s is also possible.

Detailed below are the signalling arrangements for n = 2, 3, 4 and 5. The arrangements for n = 4 and 5 are the same as that in Recommendation H.130.

Bit rate signalling

Bit 4.9	Bit 4.15	
0	0	2 Mbit/s operation
1	0	4 × 384 kbit/s operation
1	1	3 × 384 kbit/s operation
0	1	2 × 384 kbit/s operation

At 5 × 384 kbit/s time slots 1-15 and 17-31 active,
At 4 × 384 kbit/s time slots 1-15 and 17-25 active,
At 3 × 384 kbit/s time slots 1-9 and 17-25 active,
At 2 × 384 kbit/s time slots 1-6 and 17-22 active,

(3 and 2 × 384 are consistent with Annex 1 of CCITT Recommendations I.431 and G.737).

Facility bits

Bit 3.1.2.	Bit 3.1.7.	
0	0	2 Mbit/s working only,
1	0	2 Mbit/s and 4 × 384 kbit/s working only,
0	1	2 Mbit/s and 2 × 384 kbit/s working only,
1	1	2 Mbit/s and 4, 3, 2 × 384 kbit/s working only.

In the event of two codecs signalling different bit rates in bit 4, the lowest bit rate prevails assuming the facility bit allows this operation. The meaning of the facility digits also needs to be modified in respect of the digital interface to the codec, e.g.,

with codecs having a 1.5 Mbit/s serial interface:

0 0 Never occurs
0 1 Means 2 × 384 kbit/s working only;

with a 768 kbit/s serial interface:

0 1 Never occurs
1 0 Never occurs
0 1 Means 2 × 384 kbit/s working only.

8.3. Transmission at other bit rates

(For further study.)

8.4. Transmission performance

The bit-error-rate performance of the interconnection shall as a minimum conform to CCITT Recommendation G.821. The value of T_L, the period over which the error performance is averaged, and which is not yet specified in G.821, shall be taken as 2 hours, a relevant period for a videoconference session. According to G.821, the following conditions apply:

(a) BER of zero (error-free second) at least 92% of T_L (namely, 110 minutes).
(b) BER of 10^{-3} to 10^{-6}: no more than 8% of T_L (namely, 110 minutes).
(c) See below.
(d) BER worse than 10^{-3} (severely errored seconds): no more than 0.2% of T_L (namely, 15 seconds).
Not included in G.821, but recommended here for study in connection with video services of long call-time as suggested in G.821, is the following additional condition:
(c) BER of 10^{-3} to 10^{-4}: no more than 1% of T_L (namely, 1 minute).

The observed performance under these conditions is as follows:

(a) No degradation.
(b) Some horizontal streaks appear on the visual display, taking up to 2 seconds (for 2 Mbit/s transmission) to disappear; the audio is distorted by clicks of varying intensity.
(c) Several visible streaks may be on the display simultaneously; audible clicks may reach some tens per second.
(d) Virtually unusable picture, broken up and/or frozen; bandly distorted sound.

8.5. Compatibility with other services

(For further study.)

9. MULTIPOINT VIDEOCONFERENCING

9.1. Network structure

The multipoint international videoconference service is provided by means of one or more multipoint conference units (MCU). Each MCU may serve one or more terminals, and be interconnected with other MCUs as shown in Figure 4. Terrestrial interconnections between MCUs will be at 2 Mbit/s; satellite interconnections may also be at 2 Mbit/s, but other configurations are possible, taking into account the specific satellite system.

9.2. General description of operating modes

The operating modes and consequently the switching decision criterion depend on the conception of the multiconference service from each administration. Any solution, automatic or manual, can be implemented without altering the basic philosophy of multipoint videoconferencing. In each solution, the MCU provides each output port with the mixed audio signals from all other ports.

The *minimum MCU working mode* is as follows: the MCU, by comparison of the incoming sound channels, selects the loudest speaker (called New speaker or NS). A second channel is selected by the MCU, being the previous loudest speaker (called Previous speaker or PS). The NS is sent the PS channel and the other rooms are sent the NS channel. This mode is normally used when the multiconference is established. The implementation of the minimum MCU is under study. For future improvements four override modes have been defined (see Annex 4).

9.3. Additional terminal facilities

The minimum MCU mode is automatic and does not require any extra facility.

Override modes imply some modifications to the MCU and extra control equipment (pushbuttons, LEDs, connections to the codec...) at the conference room. The manner of implementation is under study.

9.4. Multipoint conference unit

The basic functions of the MCU for a terrestrial or a satellite network are identical. The MCU shall have the capability:

- To synchronise the incoming streams to a single 2,048 kHz pilot clock.
- To extract frame alignment from TS0 in order to synchronise the different streams to the frame clock, to extract frame parity, multiframe and supermultiframe alignment from TS2 in order to access in each incoming stream the codec-to-codec signalling channel.
- To process this signalling channel.
- To process the sound channels in order to create an open sound system, in the case of an unencrypted service.
- To decide image switching and dispatching according to a selection criterion (automatic or on request).
- To signal the decision of switching to the codecs in order to prepare them and to avoid any degradation during and after the switching.
- To multiplex the selected video channels with the open sound channel and the effective data channels.
- To distribute the reconstructed streams to the corresponding access ports.

10. NETWORK ARCHITECTURE

(For further study.)

11. OPERATING PROCEDURES

11.1. Reservation

The reservation procedure is defined in Annex 1.

11.2. Call set-up and pre-test procedure

These procedures are defined in Annex 2.

11.3. Fault procedure

This procedure is defined in Annex 3.

12. TARIFFS AND CHARGING ASPECTS

13. REFERENCES

CCIR Recommendation 470-2.
CCIR Recommendation 567.
CCIR Report 624-2.

CCITT Recommendations	G.172	H.100
	G.227	H.110
	G.703	H.120
	G.711	H.130
	G.732	I.431
	G.733	T.4
	G.737	T.5
	G.821	T.6
		X.21

DIN 45,500.
EIA Logarithmic Reflectance Chart.
BBC Test Chart No. 57.
No. 61.
Macbeth Colour Checker.

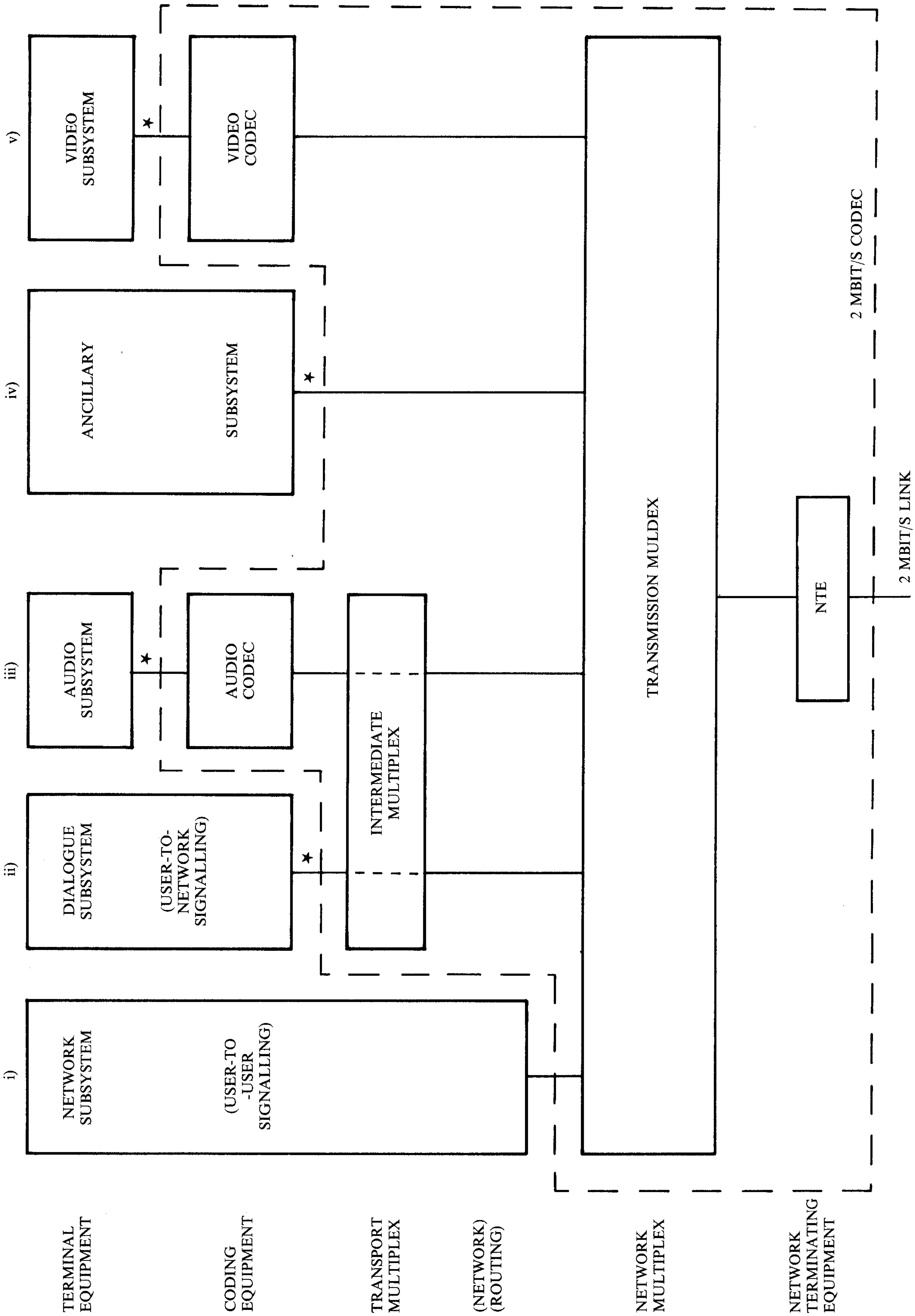


Figure 1. Functional diagram of a videoconference terminal.

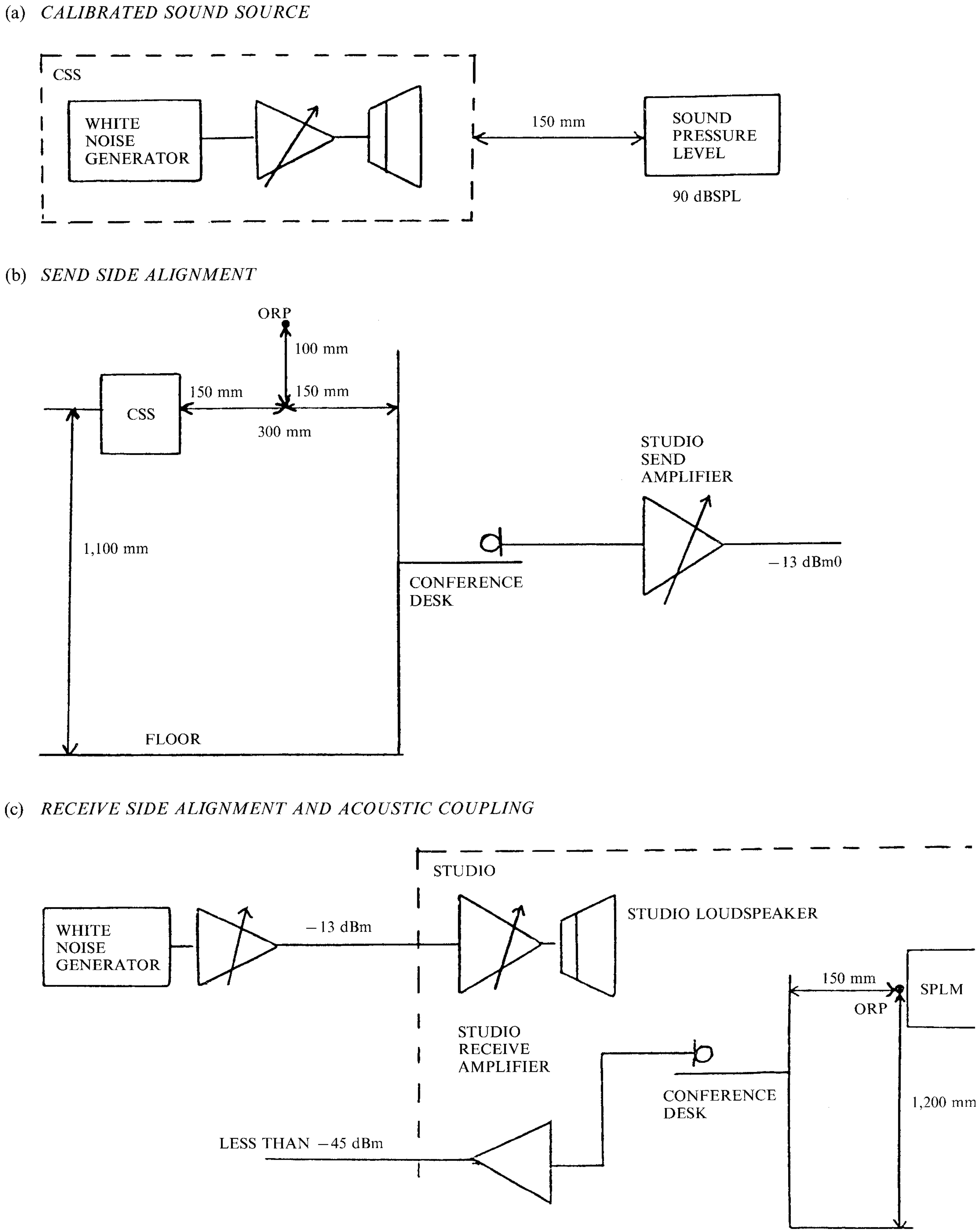
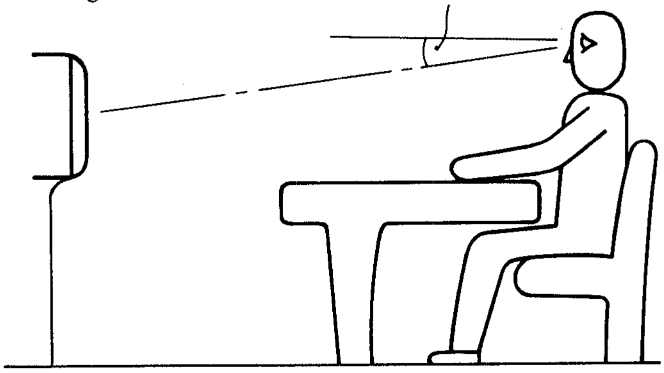


Figure 2. Audio alignment.

Figure 3b.



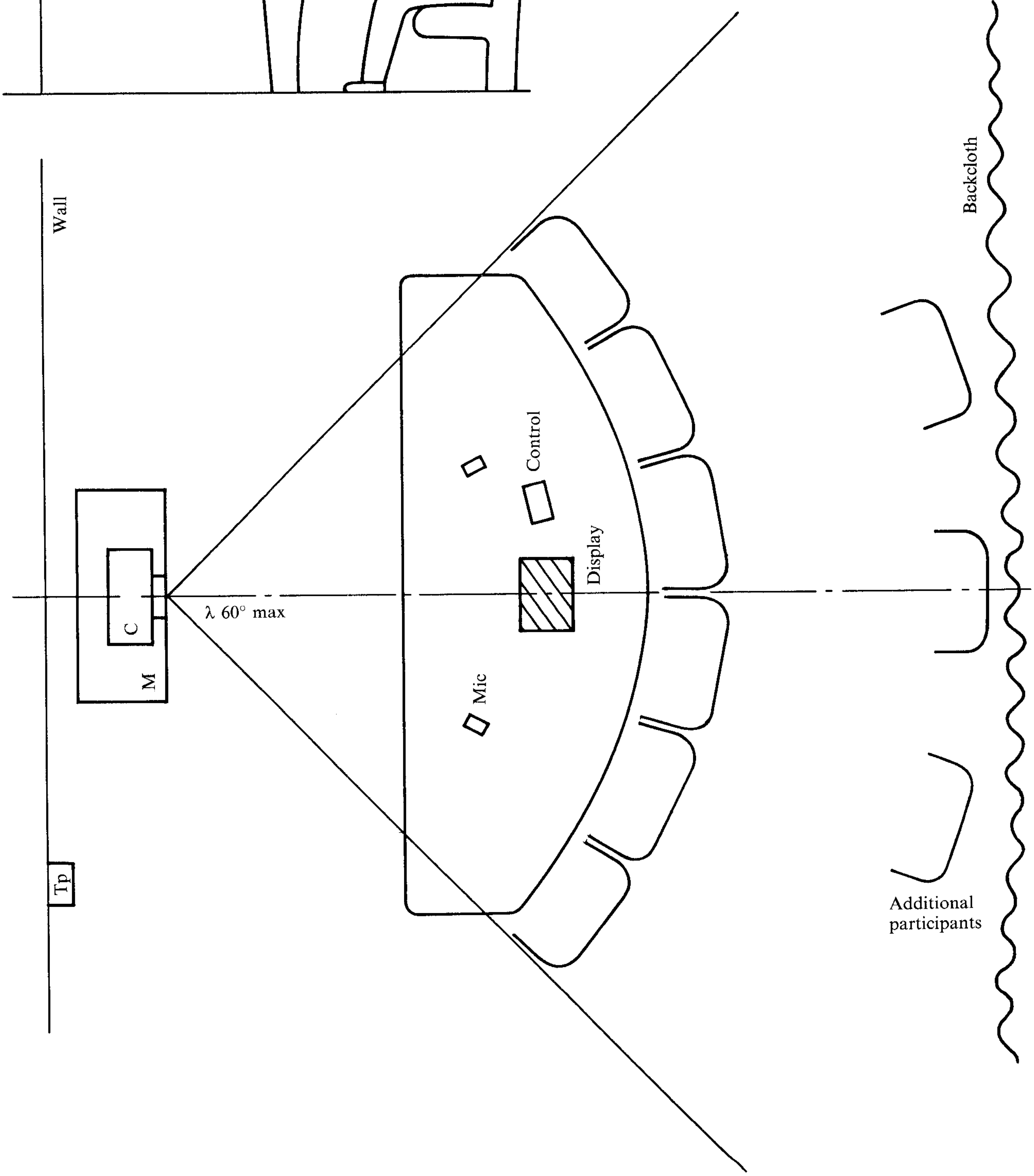


Figure 3.

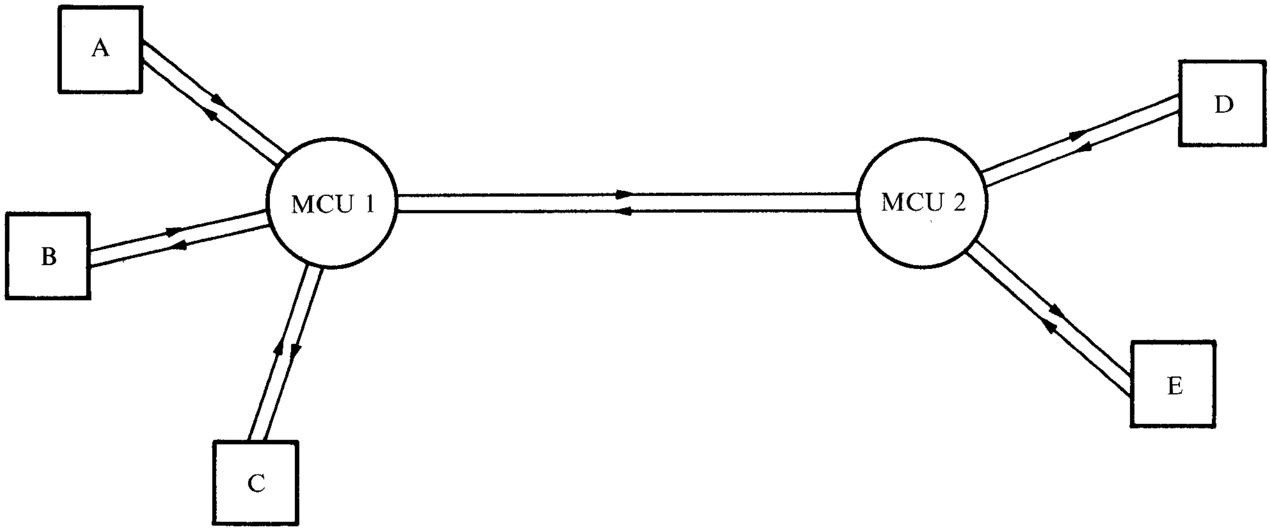


Figure 4. Use of MCU in a terrestrial network.

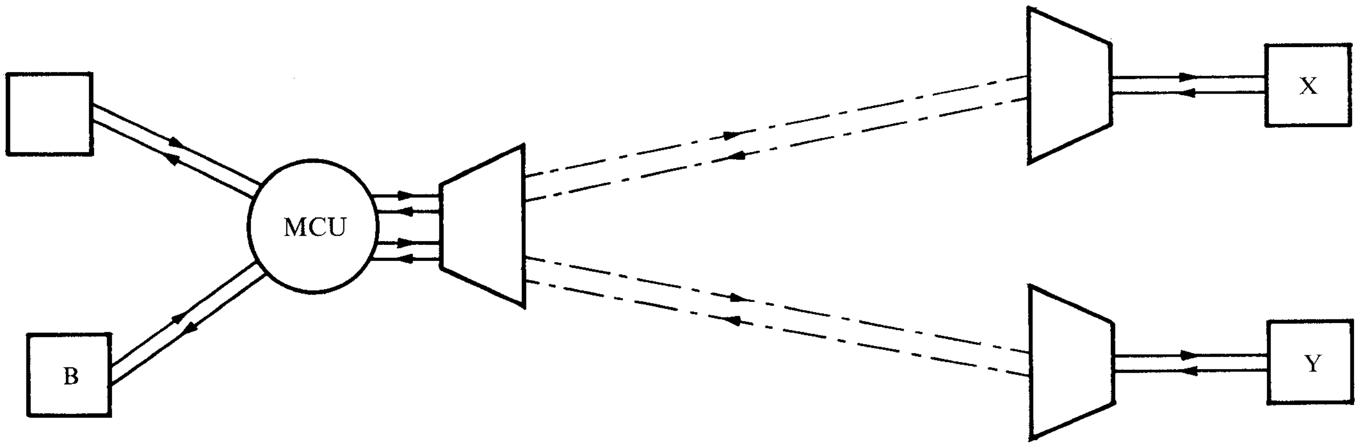
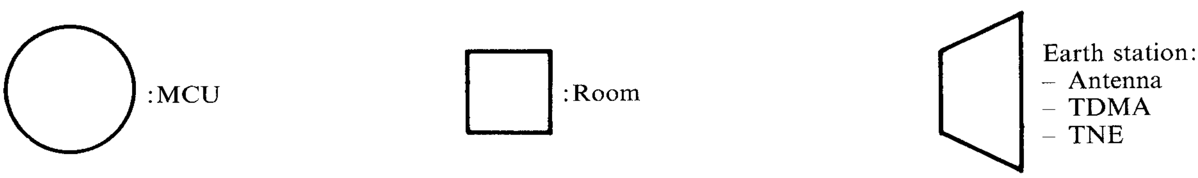


Figure 4. Use of a single MCU in a satellite configuration.



A, B, C, D, E: Studios belonging to the terrestrial network.
X, Y: Studios belonging to the satellite network.

CALL RESERVATION ARRANGEMENTS

A. Information for customers

A1. Reservations may be made up to 90 days in advance, but the minimum notice is 3 days (72 hours).

A2. Reservations will be accepted in increments of ½ hour, subject to a minimum of one hour; customers may wish to note that the Administrations may assign additional time before the call start, for any necessary engineering tests —such time will not of course be charged to the customer.

A3. Calls will be terminated promptly at the end of the reserved time.

A4. The customer will be informed by the Customer liaison officer (CLO) of the telephone and telex numbers to be used for reservations. Reservations will normally be negotiated initially by telephone, for greater speed of communication between the parties involved; following agreement of a satisfactory reservation the Administration will telex the details of the customer. The customer must confirm the reservation by letter or telex, to be received at the reservation office at least 72 hours prior to the start of the videoconference call.

A5. Times quoted by an Administration to a customer are always LOCAL times: no responsibility can be accepted for failure of a customer or his respondents to attend the videoconference terminal (VCT) at the correct time.

B. Information not intended for customers

B1. The procedure for establishing a reservation is inevitably complex, particularly if the date/time originally requested by the customer is not possible for some reason. To keep communication chains short, it is desirable that customers should negotiate directly with the National reservation office (NRO): see B6 below. However some participating Administrations may prefer such contacts to be through the Customer liaison officer.

B2. All reservation arrangements between countries, and with Eutelsat, are to be made by the NRO only.* Negotiations are preferably made by telephone, but agreed reservations must be confirmed by telex at least 3 days before the call. The procedure for negotiating and confirming reservations is depicted in Figure 1. A telex message suitable for use to all parties is suggested below:

European videoconference service

Acceptance of reservation for: (date)
Originating customer:
Originating: Videoconference terminal
Start time (at originating videoconferencing terminal)
Duration:
Distant videoconferencing terminal

Note to originating customer: your confirmation of this reservation is requested, to reach this office at least 72 hours before the above start time.

This telex copied to: both customer parties
Eutelsat
NRO posts.

B3. Reservations of videoconference terminal and satellite are to be made with a start time of 30 minutes earlier than that for the customer's call, for technical checks.

B4. Last-minute extensions to the call are not to be encouraged. However if a customer makes such a request to the NRO, the latter may, at its own discretion, attempt to contact Eutelsat, the other NRO, and the earth-station personnel before the transmission is cut off.

B5. Regular booking: the Agreement made with the customer by the CLO sets out terms for any regular booking agreed specifically for that customer; the conditions and charges are established in co-ordination with the CLO of the other country concerned.
The NRO activates the regular booking for the periods specified in the Agreement: any change to the latter is dealt with by the CLO.

* *Note:* Reservation with Eutelsat must be made by the NRO through the national Eutelsat liaison office (see B6)

B6. **National reservation office (NRO):** terms of reference

(a) The NRO maintains the diary of all calls reserved, including those subsequently cancelled, and is responsible for ensuring that all calls are duly connected through at the appropriate time.

(b) The NRO and CLO will liaise to ensure that "regular" bookings (those made regularly on a weekly/monthly basis) are agreed with a customer in an acceptable manner and satisfactory to Eutelsat. Such bookings are then notified to all other NROs.

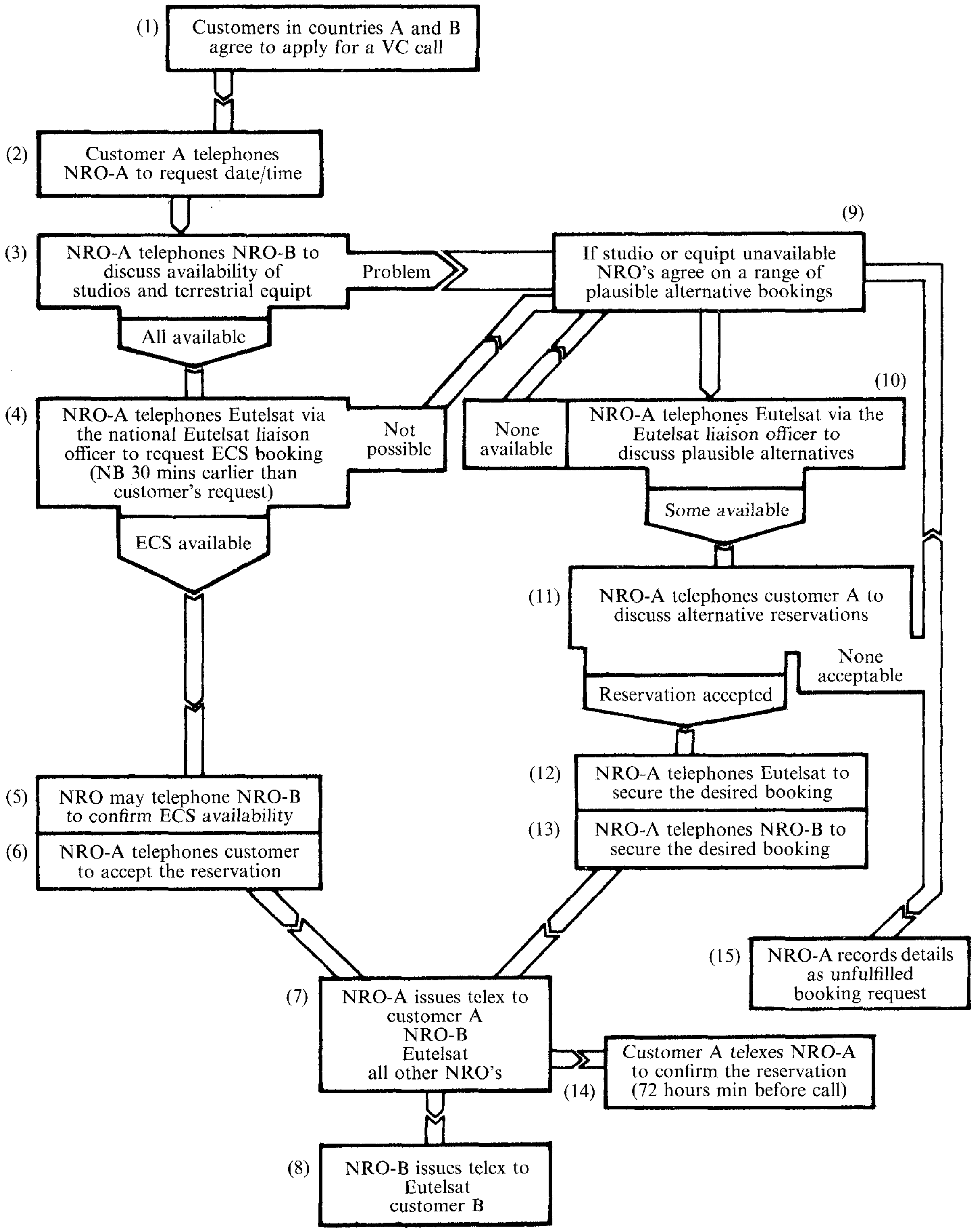
(c) All communications with Eutelsat concerning the details of reservations will be made by the NRO, through his national Eutelsat liaison office.

(d) Irregular bookings are established with the NRO directly by the customer in accordance with the procedure of para B2 above.

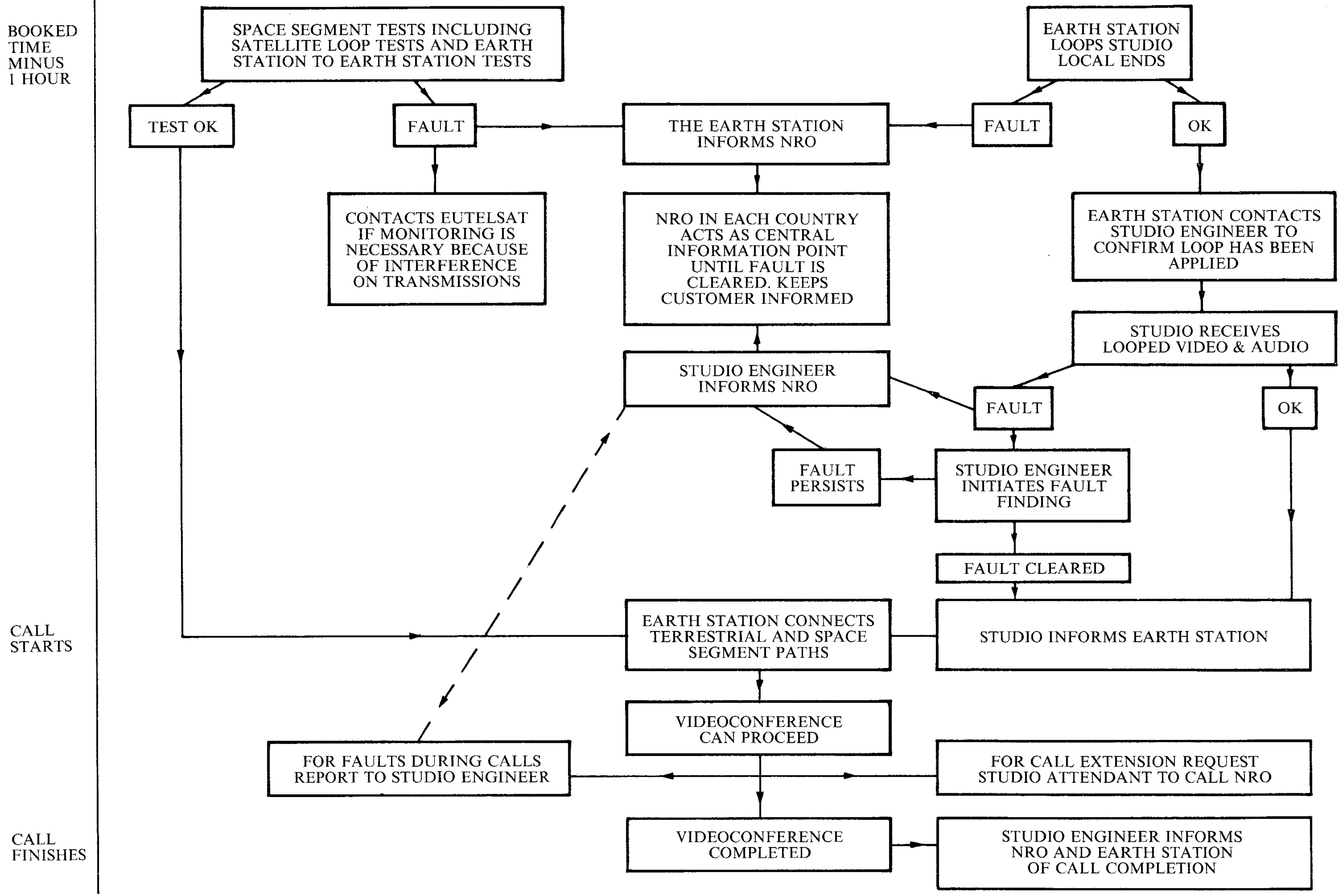
(e) The NRO will furnish the statistics of reservations, and also of requests not accepted ("congestion"), to CEPT TR1.

B7. Normally the call will be billed in both countries according to the reservation confirmed by telex (at least 72 hours in advance). However under some circumstances the outcome of the call may be different: late changes made by the customer to the NRO; failure of the call due to the Administrations; complaint by the customer (*during the call*) of poor quality. All such calls shall be accorded "asterisk" status. The NRO first applying the asterisk shall notify the other NRO immediately; following the call, it is the responsibility of the NRO-A (that is, the originating country) to establish the period for which the customer is to be charged, and to notify NRO-B by telex.

Figure 1. Call reservation procedure.



PRE-CALL TESTING PROCEDURE



EUROPEAN VIDEOCONFERENCE SERVICE FAULT PROCEDURE

The circuit designation is

The Fault reporting point (FRP) is

1. PROCEDURE FOR CALL SET-UP

1.1. For some time prior to the call set-up the circuit will be looped back at the international gateway, so that on the received picture monitor the local view should be seen; if this is not the case the FRP should be informed.

1.2. Approximately 10 minutes before the reserved videoconference the circuit will be extended through to the remote country. Picture and sound communication should be achieved simultaneously. If this is not the case the FRP should be informed.

2. FAULT FINDING MAINTENANCE PROCEDURES

2.1. If satisfactory local pictures were received while the loop at the international gateway was in place (see 1.1. above), but unsatisfactory after extension through to the remote country, no action is required other than to inform the FRP.

2.2. If unsatisfactory pictures are obtained while the circuit is looped at the international gateway, checks of the local equipment should be made.

2.2.1. Loop the 2 Mbit/s codec output back into the input and observe the receive monitor; if the picture is satisfactory the fault lies in the network beyond the codec and no further action need be taken other than to inform the FRP. If unsatisfactory, proceed to 2.2.2.

2.2.2. Take the analogue video connections from the codec and loop back; observe the receive monitor: if the picture is satisfactory the fault lies in the codec; otherwise the fault lies in the video subsystem or interconnecting leads. The FRP should be advised of the result of this test.

2.2.3. If the fault is in the video subsystem, and a display camera is included in this, operate the control which brings the display camera into operation; if the picture is now satisfactory, the conference camera(s) or associated equipment (e.g., split-screen) is at fault and should be replaced.

2.3. During fault localisation procedures for the transmission path the studio operator will be expected to co-operate with the by relaying back any changes in the received picture quality.

2.4. If intermittent faults occur during a videoconference the FRP should be informed.

2.5. Audio faults: if the audio performance is degraded but the picture is normal, then the network is not faulty and the codec unlikely to be so. The problem is most likely to lie in the audio subsystem or the room in which the terminal is located. Unlike the video, the audio equipment *cannot* be checked by looping as at 2.1. and 2.2. above.

2.5.1. Disconnect the audio in one direction at the input to the codec. Check correct operation in the other direction, giving visual confirmation; repeat this procedure for the other direction. If both directions work correctly independently, then the fault lies in the echo cancellation or level setting parts of the audio subsystems, and the alignment procedures in section 6. of this Recommendation should be repeated.

2.5.2. Likewise if one direction is not operating properly, checks should be made as in section 6. of this Recommendation.

2.5.3. The FRP should be kept informed of the progress of these checks.

SERVICE IMPROVEMENTS UNDER STUDY

(*NB:* Annex 4 has the same section headings and numbers as the Recommendation for ease of cross-reference.)

B1.1. Introduction

This document defines the equipment and procedural requirements for an international videoconference service within Europe. It covers: videoconferencing terminals, including mandatory requirements and options for the video, audio, and ancillary equipment; its disposition and accommodation; its alignment for point to point and multipoint connections; the procedural matters for call establishment. In a number of respects it has been inappropriate to make final detailed recommendations at this time (December 1984): technical and user-requirements studies are in progress or planned which will lead to recommendations in the near future; the relevant topic sections have been included for completeness, but at present contain only preliminary comments and the legend: "for further study".

B3.3. Optional facilities

Since practical experience of videoconferencing is still very limited, there is no clear consensus as to auxiliary facilities which must be provided; the videoconference 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. There follows a (non-exhaustive) list of the auxiliary facility options which a potential user may consider.

B3.3.4. *Improved audio systems*

It is possible to provide for stereophonic sound, using an auxiliary 64 kbit/s data channel for the second sound signal: however this channel will need an external delay to equalise with that of the signal passing through the normal sound channel. The problem of reduced stability under stereophonic conditions is a matter for considerable concern, particularly for a service in which a variety of different studios may be connected together on different occasions.

Other possible improvements include the provision of individual microphones, perhaps "tie-clip", pick-up with cordless connection.

B3.3.5. *Terminal equipment for multipoint operation*

If (as is probable) there is only one incoming video signal in a multipoint call, customers may wish to exercise choice as to which of the remote locations they may be viewing; a certain amount of control and messaging is associated with this situation. This matter is for further study.

B3.3.6. *Encryption*

End-to-end encryption of the digital signals can be provided; the system will indicate whether or not the signals are encrypted. As yet there is no encryption standard—it follows that encryption can only operate to other terminals having the same encryption equipment.

B3.3.7. *X-Y devices*

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

B3.5. Terminals

In general, the controls to be operated by the user should be kept to a minimum. Examples are: on/off, vision cut, audio mute, select display camera, zoom for display camera, etc.; further controls will depend on the optional facilities present.

In the multipoint configuration, provision has been made for certain functions to be assigned to a chairman if desired.

Some status indications could be made available to the participants; for example, warning of impending disconnection, operation of audio mute, operation of video cut, encryption active/deactivated, and so on. However it should be noted that discussions on standardisation of these aspects are at a very early stage. Further information is given in Section 5.5.

B3.7. Network access

Consideration is being given to further access options, taking into account the possibility of using as low as 768 kbit/s for the actual videoconference signal—see Section 8.

For the time being it is assumed that access to the integrated digital network will be at the primary access rate of 2 Mbit/s. Although within this Recommendation provision has been made for the audiovisual content of this 2 Mbit/s access to go as low as 768 kbit/s, for the time being there is no international terrestrial network infrastructure capable of supporting this lower bit rate, and therefore no means of realising the economies it ought to offer. 768 kbit/s by satellite is possible, extracting the videoconference data from the 2 Mbit/s frame at the earth station.

B3.8. Operational aspects

For a videoconference service it is essential that the connections should be available at predetermined times. An international dial-up 2 Mbit/s network of sufficient grade of service cannot be foreseen within a few years; it follows that a reliable reservation system must be instituted (see Annex 1).

It is probable that connections will be patched manually at the predetermined time, or semi-automatically by means of a manually programmed switch system. There might be no "ringing" indications, nor will the system have any knowledge as to whether participants have actually switched on their terminals or not.

Telecommunications administrations may wish to note that a centralised clearing house for international bookings of this kind will become highly desirable; it should have a fully automatic database and electronic mailing system.

B5. TERMINAL EQUIPMENT AND ACCOMMODATION

B5.1. General disposition of equipment and facilities

(m) Control units

It should be possible for participants to operate any necessary controls during a videoconference without leaving their seats.

(n) Document display

It should be possible for participants to display smaller documents or objects under the document camera during a videoconference without leaving their seats.

(o) Reception room

A reception room can be provided near to the videoconference room to allow space for welcome and introduction by receptionist, or for relaxation, private discussions, preparation for the meeting, etc.

(p) Warning light

(Outside the videoconference room, indicating that a conference is in progress.)

B5.5. Customer information and control

The details of information and control facilities at a terminal are left to each Administration. However some relevant guidance is given below.

B5.5.1. *Status indications*

The terminal should contain a monitor which at all times displays the outgoing video signal; if for any reason none of the terminal cameras are in operation a suitable character-generated identification could be transmitted instead.

Indications could be given to the customers at, for example, ten minutes, three minutes, and one minute prior to the clearing down of the transmission.

A control for sound muting is not recommended: however if provided a very clear status indication is necessary.

Further status indications may be necessary in connection with encryption and multipoint working.

B5.5.2. *Controls*

In general controls are for the operation of local equipment only, therefore not requiring international specification. However some remote controls may be found necessary:

(a) Multipoint working: selection of incoming video (manual override, or Chairman mode), request for data channel (e.g., for facsimile distribution).
(b) Mode selection: the user may wish to specify the graphics (high resolution still-picture) mode of transmission (if available) for an auxiliary video source.

B7.2. Work is carried on in CCITT SGXVIII and CEPT T/TR3 and T/TR1/HP6 for the study of wideband speech coding (0-7 kHz) at bit rates 32, 40, 48, 56 kbit/s to be mixed in the same 64 kbit/s channel with side information (frame sync, bit rate dynamic allocation, service messages, low bit rate data, ...).

B9. MULTIPOINT VIDEOCONFERENCING

The four override modes referred to in the main text are as follows:

(a) The system remains automatic but one location is considered as the chairman of the conference. Each room has the possibility to transmit a request for the floor (RF) to be displayed in one specific (chairman) room or all rooms. At appropriate time, the chairman gives *orally* the floor to the requesting conferee who, as he begins to speak is automatically selected as the new NS. This system is well suited to reproduce the normal way of performing a chaired conference in a real face-to-face meeting.

(b) One location (e.g., NS or chairman or other) has the possibility to choose the allocation of the second selected channel (normally the PS channel) by transmitting a request to the MCU.
(c) Each location has the choice between the channels, which can be made available by the first MCU connected to the location without affecting the displays of other locations. If the MCU can select more than 2 channels, the choice will be wider and easier to satisfy.
(d) Complete manual chairman control with no voice detection.

B9.3. **Additional terminal facilities**

The basic mode, automatic, does not require any extra facility, except maybe the visualisation of the name of the displayed location.
The four ouverrides require the use of a message channel currently being tested by the HP1 working group. At each location, an intelligent terminal is needed for the coding and decoding of the messages. The man-machine interface is under study.

B10. NETWORK ARCHITECTURE

It is intended that this section will set out in detail the structure of networks used for international videoconferencing service. For the time being, however, no such permanent networks are available: ad hoc arrangements are made on a per-call basis, and no automatic switching or signalling is involved.

B10.1. **General structure of the network**

B10.2. **Switching strategy**

B10.3. **Signalling**

B12. TARIFFS AND CHARGING ASPECTS

B12.1. **Point-to-point calls**

Factors to be taken into account may include:
(a) The distance between the international gateways.
(b) The bit rate used, if other than 2 Mbit/s.
(c) Time of day/week.
(d) Regularity, frequency.
(e) Cancellation charges.

B12.2. **Multipoint calls**