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1. Draft new ITU-T Recommendation F.MCVS – Multimedia Conversational Services

1. Introduction

The purpose of this recommendation is to define the Generic Multimedia Conversational Service and to describe their general features, regardless of the network environment in which these services are provided. These services allow conversational communication between two users in two different locations. The Generic Multimedia Conversational Services are one of the generic service types identified in Recommendation F.700, and their description follows the methodology described therein. The user requirements for the various applications supported by the service are translated into generic service specifications, independent of the constraints of specific implementations. Thus this Recommendation sets out the requirements that will satisfy the users' needs and allow proper intercommunication on an international basis of services offered by different providers and of equipments from different manufacturers.

Various instances (profiles) of this generic service are considered here with a network independent view. Specific service descriptions for each network will be issued in other recommendations. The detailed technical specifications of the terminal, network and protocol aspects for each of them are described in dedicated recommendations of the H.200, H.300, H.400 series.

Along with the F.700 methodology, the description relies on the media components and the communication tasks described respectively in Annexes A and B of Recommendation F.700. These are service independent modular communication capabilities.

2. Definition

The Multimedia Conversational Services provide real time transmission of voice together with motion video and/or various types of multimedia information between two users in two different locations. The information exchanged may include all information types. When moving pictures are present, their quality should be at least sufficient for the adequate representation of the fluid movements of a person displayed in head and shoulders view (see Note).

Note 1 – The smoothness of the movements in the reproduced picture is essentially dependent on the amount of motion with respect to the transfer rate and compression scheme of transmitted picture information. The above requirements are supposed to be met under such conditions where the amount of motion is limited or the throughput is high enough not to impair the received picture. Degradation may appear as increased blurring, jerkiness and/or various artefacts in the reproduced picture.

The media components used are described in the Annexes A of Recommendation F.700 on Audiovisual/Multimedia Services. Media component Audio (Annex A.1) is usually present, together with one or more of the media components Video (Annex A.2), Text (Annex A.3), Graphics (Annex A.4), Still Pictures (Annex A.5).

3. References and terminology

3.1 References

ITU-T Rec. F.700 - Framework Recommendation for audiovisual/multimedia services

ITU-T Rec. F.702 - Multimedia conference services

ITU-T Rec. G.114

ITU-T Rec. G.711

ITU-T Rec. G.722

ITU-T Rec. T.120

ITU-T Rec. T.140, Text Conversation Protocol for Multimedia Application. (1998)

ITU-T Supplement 1 to series H Application profile – Sign language and lip-reading use of low bit-rate video communication. (1999)

3.2 Terminology

Collaborative Document Handling Service (CDH): a service that provides bi-directional transfer of data between two or more locations so that users are able to work on a common document, for drafting or amending it collectively.

Communication modes : a communication mode is defined by the various channels supporting the media used for the service. Changes in communication mode may occur during the course of the call, in order to set up or eliminate one of the media, or to change its quality level and thus the bit-rate allocation. It may be used for instance to temporarily add a channel for transmitting still pictures.

Multipoint conference unit (MCU) : equipment that provides multipoint connections among 3 or more conference rooms.

Muting : preventing sound to be transmitted from a terminal equipment.

Videoconference Service : an audiovisual conference service providing bi-directional real-time transfer of voice and motion video between groups of users in two or more separate locations. Although the audio and motion video informations are the essential part of the service, other types of information, such as high resolution still pictures, text or graphics may also be exchanged.

Videophone Service : an audiovisual conversational service providing bi-directional symmetric real-time transfer of voice and motion video between two locations. The minimum requirement is that under normal conditions the picture information transmitted is sufficient for the adequate representation of fluid movements of a person displayed in head and shoulders view.

Total Conversation Service : an audiovisual conversation service providing bi-directional symmetric real-time transfer of motion video, text and voice between users in two or more locations.

Text Telephone Service : an audiovisual conversation service providing bi-directional real time transfer of text and optionally audio between users in two locations. Audio may be transmitted alternating with text or simultaneously with text.

4. Description

4.1 General description

A multimedia conversational service provides real time bi-directional communication via telecommunication networks between users in two different locations; it usually combines an audio facility with motion video of the users and/or transmission of multimedia informations; however the audio may not be present in some particular applications. The service is applicable to dedicated terminal equipments or to microcomputer based terminals.

Multimedia Conversational Services are essentially built around the communication task Conversing described in Recommendation F.700-Annex A.1. Other communication tasks (receiving, sending) may optionally also be used. The control functions are described in Annex C.2 – Middlewaren service element Conversation control – of Recommendation F.700.

4.2 Functional model

In a multimedia conversational service, two terminals exchange multimedia informations through a telecommunication network (Figure 1).

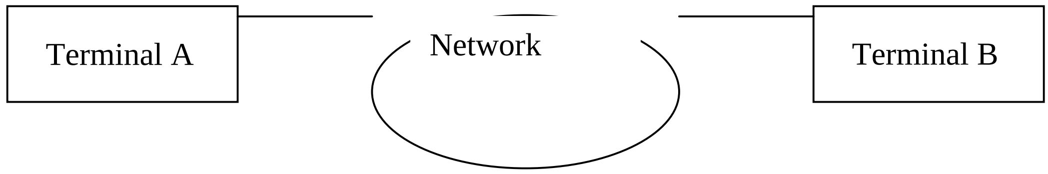

Figure 1 - Functional model.

4.3 Configuration

The basic configuration is point to point between two terminals communicating through a bi-directional connection. This connection is usually symmetrical, but in special cases the media components present in each direction may be different, or they may be the same but with different bit-rates and quality of service.

A conference call may be set up as a supplementary service in conformance with recommendation F.702, in which case the configuration is multipoint to multipoint; the terminals are then connected through a multipoint conference unit (MCU) that fulfils three functions :

a) managing the call, setting up and closing the connections;

b) managing the conference, through control and indication signals exchanged with the terminals;

c) handling the signals received and sent on each connection, switching, distributing, multiplexing and when necessary adapting and combining them as appropriate.

4.4 Terminal aspects

Audio is used for the service except in exceptional cases. Therefore in order to perform the basic functions necessary for multimedia conversational services, the terminal equipment should include the following units necessary for audio communication :

- a microphone,
- one (or more) loud-speaker(s),
- an audio codec,
- optionally some audio related controls,

The terminal must also include a network interface unit. The other types of information require specific equipments detailed below. The terminal should include the equipment(s) for at least one media component besides audio.

The equipment for handling multimedia documents includes one or more of the following functional units :

- a micro-computer with a screen and a modem,
- a still picture equipment with a camera or scanner, a screen and a modem,
- a telewriting equipment,
- optionally a printer.

The basic equipment for video includes :

- a camera,
- a screen,
- a video codec.

When video is present, means must be provided for displaying the outgoing picture, either permanently or by substituting it on the screen to the incoming picture.

Note - Testing of the outgoing picture : it should be possible for the user to put an off-line terminal into a self-test procedure, which includes the codec, in order to test and control the outgoing picture.

4.5 Applications

Some possible applications are indicated here as examples :

- various types of conversation between two distant parties, similar to a telephone conversation but enhanced with motion video, still pictures or text;
- conversation between two distant parties, where one of them at least has difficulty in hearing and uses another media, either because of a disability or because of a noisy environment; examples are conversation with sign language using a videophone, lip reading to help hearing impaired people, text conversation;
- elaboration of a document between two parties, with or without cooperative document handling,.

4.6 Supplementary services

For further study.

5. Static aspects

5.1 Service level

5.1.1 General aspects of the service

Throughout the communication phase, the communication task Conversing is usually active with the media component audio and one or more other media components.

Exchange of various types of documents may optionally be available, using the communication task Sending with one or more media components.

The relationship between the three levels of the multimedia service reference model (service, communication tasks and media components) is shown on Figure 2.

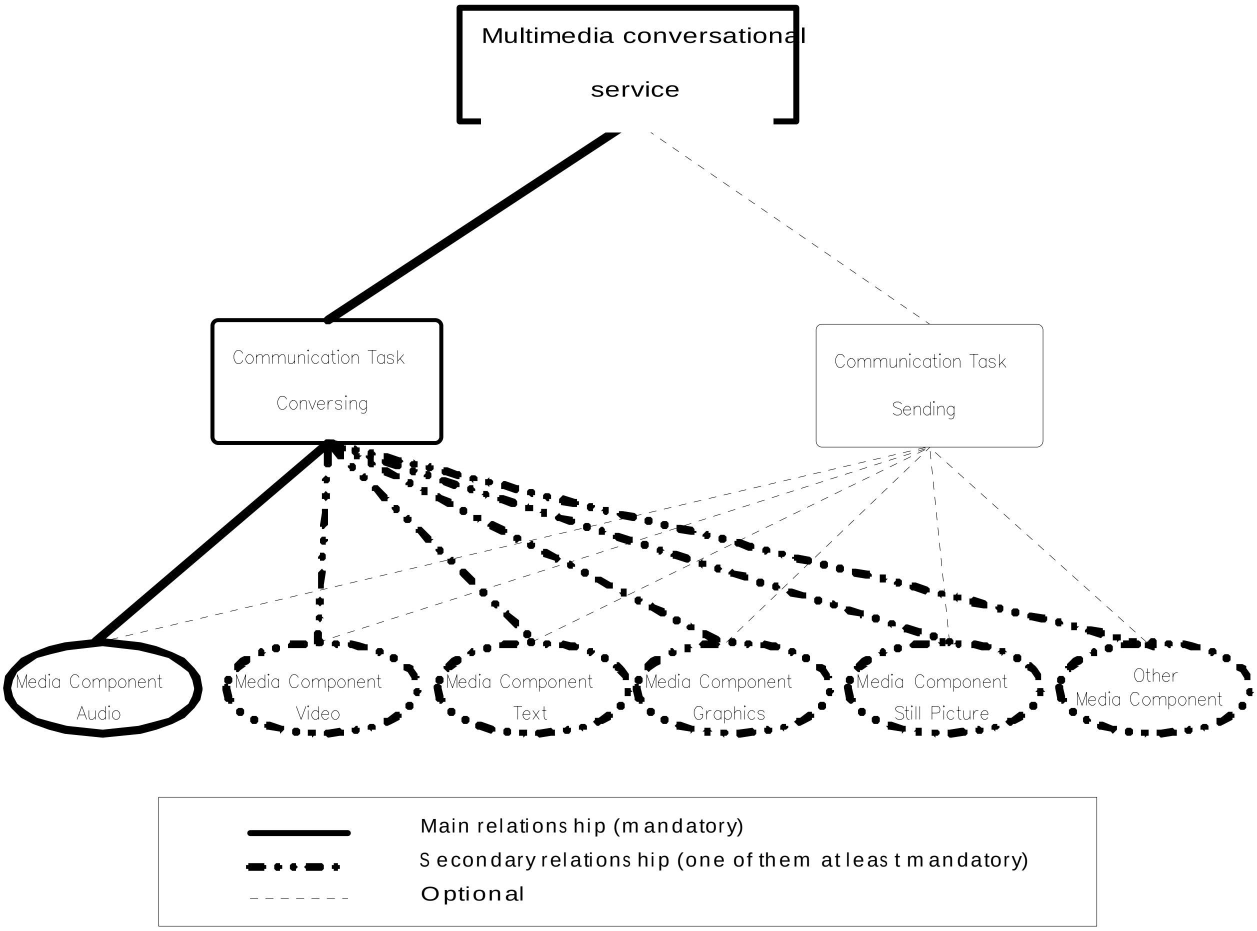

Figure 2 - Relationships between service level, communication tasks and media components.

5.1.2. Quality of service

5.1.2.1 General considerations

The quality of service depends on the quality of each media component (see section 5.3.1), of their association (e.g. synchronisation of sound and image), of the control and procedures. It also depends on the combined quality of the various parts of the system and its environment.

5.1.2.2 Overall delay

The nominal overall delay is defined as the sum of the transmission delay and the characteristic delay of the terminals (note2). The subjective effect of this delay on the quality of service has to be taken into account because an excessive value may impair user acceptability.

Under normal conditions the overall delay should not exceed 400 ms, in conformance with Recommendation G.114. Another recommendation is expected for audiovisual services. However, when no means are available to fulfil this condition, the call should not be rejected.

Note 2 : the characteristic delay of a terminal is the delay introduced by the terminal when the transmitted information contents for the various media is minimal; for the video, this means that there is no movement or only very small movements in the transmitted image, e.g. only lips and eyes of the users are moving.

5.1.3 Security aspects

No specific means for security are usually required. They may however be included for particular applications.

5.1.4 General charging principles

The same charging options should be offered as for telephony.

5.2 Communication task level

5.2.1 Communication tasks used for the service

Multimedia conversational services are built around the communication task Conversing (see Annex B.1 to Recommendation F.700). Other tasks such as Receiving and Sending may also be used for specific purposes.

5.2.2 Quality of service related to the communication tasks

Differential delay between sound and image

When video is present, the differential delay between sound and image should be kept low enough for lip synchronism to be subjectively insured. The differential delay should not be allowed to become more than 120 ms, and should preferably remain below 100 ms. ITU-T H-series supplement [ITU-T Supplement 1 to series H Application profile – Sign language and lip-reading use of low bit-rate video communication. (1999)] reports that asynchronism under 100 ms is required.

5.3 Media component level

5.3.1 Media components used for the service

The Media component Audio (see Annex A.1 to Recommendation F.700) is normally set up at the beginning of the call and is usually present throughout, but it may be temporarily interrupted to increase the bit-rate available for other media components.

One other media component at least should also be present, permanently or not, i.e. one of the following components:

- Video (Annex A.2 to Recommendation F.700)
- Text (Annex A.3 to Recommendation F.700)
- Graphics (Annex A.4 to Recommendation F.700)
- Still Pictures (Annex A.5 to Recommendation F.700)
- others (for further study).

5.3.2 Quality of service related to the media components

5.3.2.1 Sound quality

The following two levels of sound quality are preferred:

- the basic quality is equivalent to that of the PCM telephony conforming to recommendation G.711 (level 0 of media component Audio);

- the high quality is 7 kHz bandwidth (conforming to recommendation G.722) equivalent to present broadcast TV sound (level 1 of media component Audio).

For the quality levels of the media component Audio, see Annex A.1 of Recommendation F.700.

Means for efficient echo control must be provided. When video is present, these means should have the capability to accommodate the large transmission delay that may be induced by the synchronisation of sound and image (see Section 5.2.2).

5.3.2.2 Image quality for the video

When video is present, its quality should be sufficient for reproducing the fluid movements of a person in head and shoulders view (Note 5). Four grades of image quality may be offered:

- minimum videophone quality : one person with only very small movements can be viewed with a limited resolution; this is level 0 of the media component Video and is applicable to low bit-rate connections ;
- basic videophone quality : one person with only small movements can be viewed; this is level 1 of the media component Video ; it allows limited language perception ;
- enhanced videophone quality : at least 2 persons with only small movements can be viewed; this is level 2 of the media component Video ; it is equivalent to the basic videoconference quality ; it allows a good language perception level, a good usability for sign language and lip-reading, and a comfortable intercommunication with the videoconference service ;
- high quality videophone : the quality should be that of broadcast TV and possibly that of HD-TV; this is level 3 or 4 of the media component Video.

For the quality levels of the media component Video, see Annex A.2 of Recommendation F.700.

A factor of quality related to video is the recovery time for picture build-up when the video source is changed; it is important to keep it low in the case of conference calls, but may also apply when two or more cameras are used alternately, e.g. one showing the user and another one for documents.

Complete definition and methods of evaluation of the quality of motion pictures require further studies.

Note 5 - When motion video is present, the smoothness of the movement in the displayed image depends on the ability of the system to convey rapid changes; the amount of information which has to be transmitted will of course increase with the affected area in the picture and the speed of the movement. It is generally considered that in a normal conversation, only part of the picture will be moving and that any fast movement will be limited to a small portion of the image. Some degradation of the picture quality such as blurring or other artefacts may be allowed to occur when these conditions are not met.

5.3.2.3 Text Quality

Text conversation is supposed to be supported according to the principles of T.140 [ITU-T Recommendation T.140, Text Conversation Protocol for Multimedia Application. (1998)].

Preferred display areas and other requirements can be found in ETSI ETR333 [Human Factors: Text Telephony, User requirements and proposals].

Text quality is measured in corrupted characters, dropped characters and characters replaced by the missing text marker [see ITU-T Recommendation T.140].

Another factor of quality is the ability to use a complete set of characters for various languages.

For the quality levels of the media component Text, see Annex A.# of Recommendation F.700.

6. Dynamic aspects

6.1 Activation phase

6.1.1 Provision, withdrawal

The multimedia conversational services may be provided after prior arrangement with a service provider, or they may be generally available.

6.1.2 Call establishment

Call establishment is on demand. Incoming calls shall be announced with means that is perceivable by the users, e.g. audible, visual or tactile alerting signals preferably selected by the user.

Call progress shall be announced to the caller with visible and audible signals.

6.2 Communication phase

6.2.1 Call set-up

The call is made according to the general procedure applicable to the network involved. If several modes are allowed, negotiation occurs between the terminals for the selection of the mode to be used. This is usually an automatic process relying on the general options of each user. When required, additional channels are established and aggregated to the initial channel.

6.2.2 Change of communication mode

Changes in user requirements arising during the course of the communication may require a change of communication mode, i.e. changes in the network connections or changes in the in-band channel that are transparent to the network. If the network supports it, then in-call change of mode should be allowed when both terminals have the capabilities for the new mode. A change of mode that entails an increased cost for the communication may only be initiated from the charged end.

6.2.3 Conference calls

A conference call is a supplementary service. It is established in conformance with Recommendation F.702 and is made through a multipoint control unit (MCU). It uses the communication task Conferencing (Annex B.2 to Recommendation F.700), and the middleware service element Conference Control (Annex C.1 to Recommendation F.700).

6.2.4 Audio muting and video inhibition

Any participant may temporarily prevent his terminal from sending out audio or video signals. A suspended video should be replaced by an adequate notice. A terminal should provide a visual indication when its audio is muted or its video inhibited.

6.2.5 Access to other services

6.2.5.1 General

Access to other services may be made by one of the participants, who forwards the multimedia documents or informations received to the other participant in the call; this requires an additional access to the network, the appropriate rights of access to the service and possibly specific rights for forwarding documents to other users; the service may be public or private, e.g. retrieval of multimedia documents from a private server.

In the case of a conference call, access may also be made through the MCU, which then has to provide an access to the network, rights of access to the service (unless it can use the rights of one or several of the participants in the conference through some authentication procedure), and possibly conversion of protocols, of coding or of media.

6.2.5.2 Retrieval services

For further study.

6.3 Termination phase

Call termination may be initiated indifferently from the calling or from the called end.

7. Service profiles

7.1 Different types of Multimedia Conversational Services

Multimedia conversational services can be divided into different types of profiles according to the various kinds of information exchanged and to their quality levels. They usually have in common the capability for transmission of sound. The different types are:

- Videophone service, with audio and moving pictures and optionally various types of data;
- Voice and data services, with audio and various types of data;
- Text telephony, with real time text, optionally combined with audio
- Total Conversation service, with moving pictures, real time text and audio.
- Collaborative Document Handling Service (CDH), with real time text, data and possibly graphics; this service is often used in multipoint configurations and is therefore described in Recommendation F.702.

7.2 List of service profiles

The following profiles of the Generic Multimedia Conversational Services are defined. As stated in Recommendation F.700, these profiles are references for the offering of services, but do not preclude any enhancement or additional functions. Additional profiles may be included in the future to respond to evolving user needs.

1) Videophone service
   - Profile 1a, Low bit-rate videophony : low bit-rate audio, QCIF video or less with limited movements capability;
   - Profile 1b, Basic videophony : PCM telephony audio, QCIF video or CIF video with limited movements capability;
   - Profile 1c, Enhanced videophony : wideband audio, CIF video.
   - Profile 1d, High quality videophony : wideband audio, TV quality video or HD-TV quality video.

2) Voice and data services

These services may be offered with two levels of audio quality and with three types of exchanged data. The minimum audio quality is level A0, low bit-rate audio (profiles 2.a); the basic audio quality is level A1, equivalent to PCM telephony audio. The three types of exchanged data are still pictures from a camera or equivalent system (profiles 2a1, 2b1), text (profiles 2a2, 2b2), and microcomputer files (profiles 2a3, 2b3).

- Profile 2a1, low bit-rate audio with still pictures;
- Profile 2a2, low bit-rate audio with text;
- Profile 2a3, low bit-rate audio with file transfer;
- Profile 2a1, basic audio with still pictures;
- Profile 2a2, basic audio with text;
- Profile 2a3, basic audio with file transfer.

3) Text Telephone Service

- Profile 3a, Usable text conversation, with text only.
- Profile 3b, Usable text conversation alternating between text and audio (Introduced here for historical reasons, to describe existing services. There is no desire to limit new implementations to the alternating text/audio profile)
- Profile 3c, Good text conversation with simultaneous usable audio

4) Total Conversation Service

- Profile 4a, Minimum total conversation, with usable audio, usable text and perceivable video
- Profile 4b, Standard total conversation : with usable audio, good text and good motion optimised video.

The following table recapitulates the requirements for the various types of profiles in terms of media components and their minimal levels of quality.

Service	Profile	Audio	Video	Text	Graphics, and/or Still Pictures	File transfer
VPS	*P1a*	*A0*	*V0**	*O*	*O*	*O*
	P1b	*A1*	*V1**	*O*	*O*	*O*
	P1c	*A2*	*V2*	*O*	*O*	*O*
	P1d	*A3*	*V3 or V4*	*O*	*O*	*O*
VDS	*P2a*	*A0*	*O*	*O**	*O**	*O**
	P2b	*A1*	*O*	*O**	*O**	*O**
TTS	*P3a*	-	-	*T1*	*O*	*O*
	P3b	*A1**	-	*T1**	*O*	*O*
	P3c	*A1*	-	*T2*	*O*	*O*
TCS	*P4a*	*A1*	*V0*	*T2*	*O*	*O*
	P4b	*A1*	*V2*	*T2*	*O*	*O*

A0, A1, ... minimum mandatory audio quality level

A1* minimum mandatory audio quality level alternating with text

V0, V1, ... minimum mandatory video quality level

V0*, V1*, ... minimum mandatory video quality level, with a limited amount of movement

T0, T1 minimum mandatory text quality level

T0* minimum mandatory text alternating with audio

O optional

O* one at least of the media components mandatory

Note : telephony and text only service are monomedia services, but apart from that difference they share the same specification as the above services.

8. Interworking and intercommunication

8.1 General

Terminals with different characteristics and capabilities may be connected in a multimedia conversation. They use a common mode that both terminals can handle. Any media that both terminals supports will be present with the lower of the two quality levels if these differ.

8.2 Intercommunication with a telephone terminal

The communication is made with sound only.

8.3 Videophone and other multimedia conference terminals

8.3.1 Audiographic conference and videophone terminals

These terminals can intercommunicate with sound only. Basic videophones have a lower quality level than audiographic conference terminals which will have to fall back to this lower level to establish communication. However with broadband videophones, the situation may be the reverse. The ability to select one video picture in the video terminal (or in the MCU in the case of a multipoint call) for sending it as a still picture to the AGC terminal would be a desirable feature.

8.3.2 Videoconference and videophone terminals

If the video codecs have a common mode, then this can be used for intercommunication. However, with a basic videophone terminal, this may not be desirable in a multipoint call because of the limited image quality, insufficient to show several participants simultaneously. The other possibility is then for the videoconference rooms to communicate between themselves in their usual mode, while the videophone terminal participates with sound only.

8.3.3 Communication with a facsimile terminal

The communication is not possible unless the multimedia conversational terminal supports facsimile on its data channel.

8.3.4 Text telephone interworking

It may be of interest for users of multimedia conversation terminals to be able to conduct text conversations with text telephones. This interworking may be possible either if the terminal has text conversation capabilities and a gateway for interworking with the text telephones is available, or the terminal has capabilities for direct interworking with the actual type of text telephone.

8.3.5 Total Conversation interworking

Total Conversation terminals should be capable of interworking with Total Conversation terminals in other networks through gateways. They should also interwork with other multimedia terminals with a subset of their functionalities.

2 Draft new Recommendation F.user – Guideline Recommendation for identifying multimedia service requirements

1 Scope

This Recommendation provides guidelines for describing user requirements that are to be used as the basis for constructing new multimedia services. These guidelines are primarily intended to support the Multimedia service development methodology described in ITU-T Recommendation F.700. However, they can also be used as the basis for a structured dialogue between End-Users and Service Providers in order to arrive at a responsive service solution when applicable service Recommendations are not yet available.

2 Definitions

For the purpose of this Recommendation the terms defined in Recommendation F.700 will apply. The definitions of some important terms are reproduced in Appendix I for user convenience.

3 Multimedia service development methodology

A detailed methodology for developing Multimedia services is described in Recommendation F.700. Figure 1 provides an overview of this methodology and shows how end-user requirements are inserted into the service development process through the use of Application Scripts. The construction of these Scripts from End-User requirements is described in the remaining sections of this Recommendation.

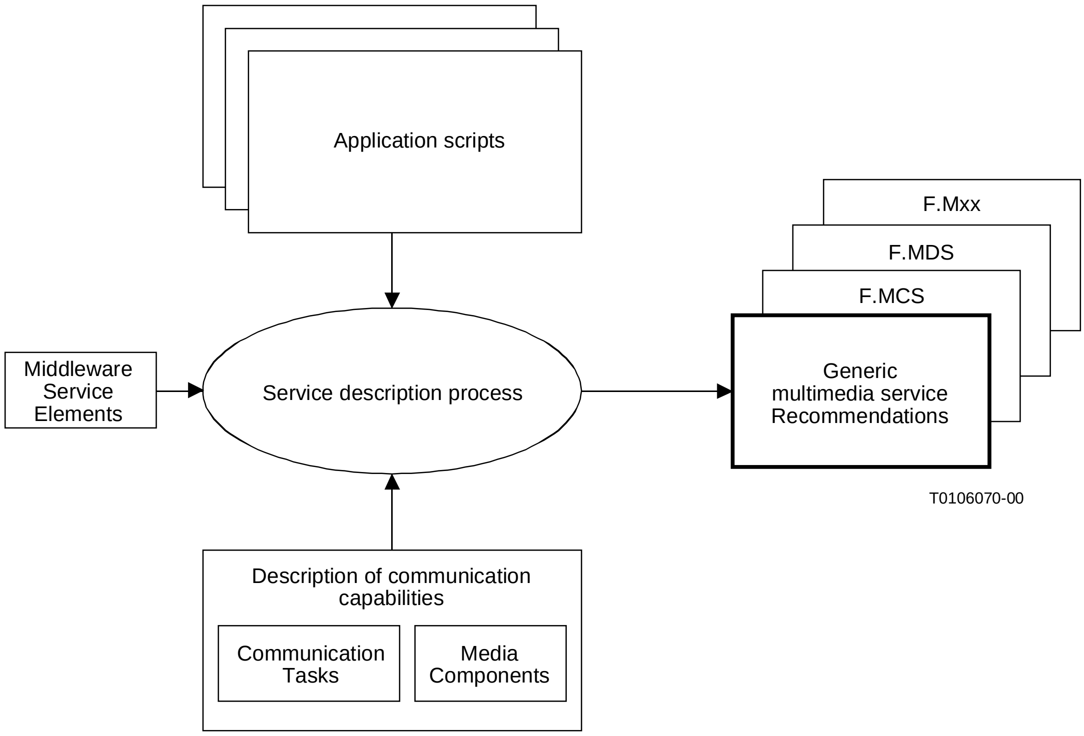

FIGURE 1/F.700

Multimedia service development methodology

3.1 Application scripts

An application script is a document that describes the essential characteristics of an End-User application so as to facilitate identification and evaluation of the multimedia communication capabilities required to support it. The script, when properly validated, provides the baseline requirements for new multimedia services. The procedure for developing and validating application scripts is described in clause 2.

3.2 Communication capabilities

Communication capabilities are the fundamental sets of communication tasks, media components and integration mechanisms required to develop the complex spectrum of multimedia services. The procedure for translating the application script into the required communications capabilities is described in ITU-T Recommendation F.700. Procedures are also identified for initiating the development of new communications capabilities when required to more fully support emerging user needs.

3.3 Middleware service elements

The middleware service elements contain all the control features and the processing functions associated with the service.They interact with the various communication capabilities in order to control them or to process the user information.

3.4 Multimedia service Recommendations

The translation of a particular application script into a description of the required multimedia service can be accomplished directly from the basic communication capabilities by utilizing the procedures specified in Recommendation F.700. However, this process can be simplified in many cases by recognizing that a significant number of end-user applications utilize just a few combinations of multimedia communication means. The methodology for describing these generic service architectures in a series of general ITU-T service Recommendations is also described in Recommendation F.700.

4 Application Scripts

4.1 An application script describes the essential characteristics of an end-user application in a manner designed to facilitate identification and evaluation of the required multimedia communications support capabilities. This is accomplished by first describing the application from the end user’s point of view and then translating this description into a form more useful for technical evaluation. The procedures for constructing an application script are described in 2.2 through 2.4.

Ideally, an application selected for the scripting process should represent a broad grouping of individual end-user applications which have the same essential functional characteristics and for which there appears to be a need for the development of a new multimedia service, service arrangement or enhanced service capability.

Differences between specific applications within this broad grouping can be represented by the specific values assigned to a particular requirement attribute. Examples are shown in 2.4. The procedures for validating the results of the scripting process are described in Clause 3.

4.2 Prose description

The prose description of an application provides a comprehensive statement of its scope and functional characteristics, together with the user’s expectations for the quality of service. This description is written in a language understandable to the end user, who need not be aware of the technical aspects of the underlying service or supporting communications networks.

The prose description may be augmented by an application scenario and a set of implementation notes which further describe the application, highlighting those aspects which might otherwise remain unclear. A sample prose description, with associated application scenario and implementation notes, is provided in Appendix I.

4.3 Functional model of an application

The functional model provides a pictorial representation of the essential functional elements identified in the prose description. This representation is presented from the perspective of the application, rather than from the supporting service or network, and contains only those elements visible to the end user. Figure 2 provides the functional model for the prose description contained in Appendix II.

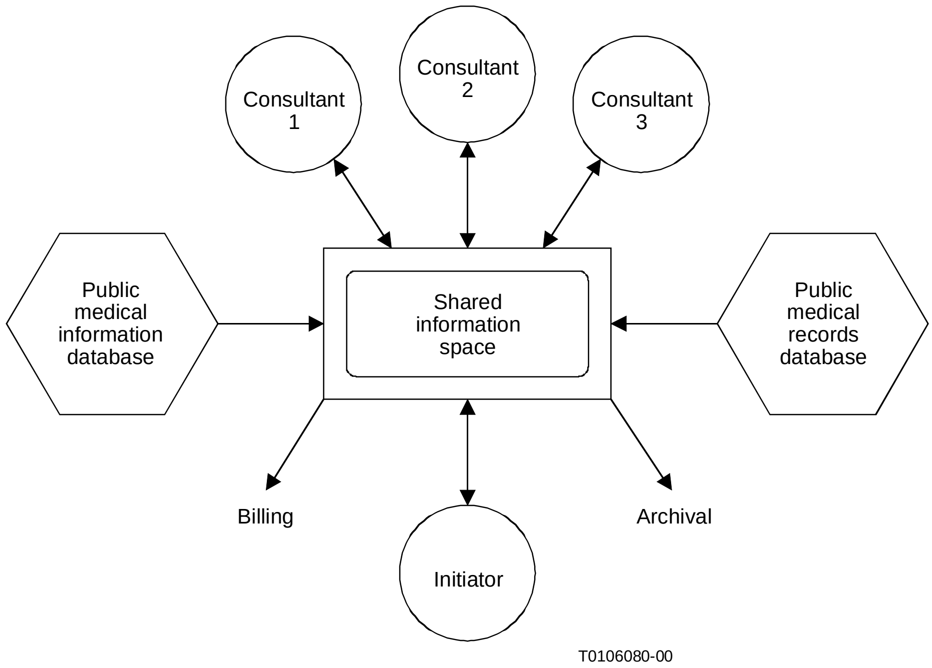

FIGURE 2/F.700

A sample functional model of an application (medical consultation)

The principle characteristics to be depicted in the model are:

- the shared information space in which the interaction takes place;
- the functional role of the major participants;
- the required supporting information resources;
- the type and configuration of the various interactions; and
- the need to interface associated application processes.

While there is no standard symbology for constructing the functional model, care should be taken to select a form of presentation that reflects the essential functional elements of the application in a clear and concise way.

4.4 Application matrix

An application matrix maps user requirements onto technical functionalities. The principles for developing attribute tables are the following:

1) Application matrices are intended to facilitate the mapping of user needs with technical functionalities in an easily understandable form.

2) Application matrices enable the evaluation of service functionalities in a systematic and compact form.

3) Application matrices facilitate assessing the importance of functionalities in regard to user needs.

Table 1 shows as an example a part of an application matrix:

TABLE 1/F.USER

Application matrix template

User needs	**Technical functionalities**		
	Differential delay between audio and video	**Image repetion rate**	**Image resolution**
Lip reading (with head view)	< 100 ms	> 20 pictures / s	QCIF (178 x 144 pixels)
Sign language	-	> 20 pictures / s	CIF (358 x 288 pixels)

The following are examples of user needs:

– discussion of a jointly viewed document;

– the need to move around;

– the need to scrutinize fine details of a presented object.

The following are examples of functionalities the applications may require:

– shared viewing space for images;

– cordless communication access;

– high resolution image transfer.

The development of the matrix requires further study.

4.5 Summary

A script may include a prose description, an application scenario, implementation notes and an application matrix. (or several matrices for different environments or different times in the communication) . Some scripts may contain only part of those elememts.

5. Harmonization of Application Scripts with other bodies: Application scripts can be developed by the ITU or by other standards organizations, industry fora, consortia, user groups or individual end users. An application script, before being used as the basis for initiating a new service development or evaluation effort by the ITU-T, should be discussed with the end user community if possible or reasonable. This discussion should take place between the relevant study groups and those organizations that have been identified as most representative of relevant end user interests, in accordance with ITU-T policies and procedures (see ITU-T Recommendation A.4).

Appendix I

Definitions

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

Definitions taken from Recommendation F.700.

I.1 **application**: An Application is a set of activities performed to respond to the needs of the users in a given situation for purposes such as business, education, personal communication or entertainment. It implies software and hardware utilization, could be performed in a fully or partially automatic way and could be accessed locally or remotely. In the last case, it requests use of telecommunication services.

I.2 **multimedia** {MHEG}: The term multimedia is an adjective which means relative to two or more media; it must be attached to a noun which provides the context. For example, multimedia service or application, multimedia terminal, multimedia network and multimedia presentation.

I.3 **multimedia application**: A Multimedia Application is an application that requests the handling of two or more representation media (information types) simultaneously, which constitute a common information space. Examples are cooperative document editing, long distance meetings, remote surveillance, medical document remote analysis and teletraining.

I.4 **multimedia service**: Multimedia services are telecommunication services that handle two or more types of media in a synchronized way from the user's point of view. A multimedia service may involve multiple parties, multiple connections, and the addition or deletion of resources and users within a single communication session.

Appendix II

Multimedia medical consultation

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

II.1 Prose Description

Medical consultation involves interactive multimedia communications between medical experts located at two or more separate locations. This communication is generally initiated by a doctor desiring to discuss a particular patient's case with subject matter experts and may occur between the doctor and one consultant only, or may require an interactive conference arrangement between the doctor and several consultants simultaneously.

In the course of the consultation, information may also be required from remote databases containing the patient's medical files; from one or more diagnostic test centres in the form of X-rays, sonograms, electrocardiographs or similar medical images; or from a reference library containing technical information, illustrative medical images, or other supporting material required to facilitate the consultation. This material may be textual, aural, graphical or imagery in nature and may be stored in a multimedia format.

Participants in the consultation may be located in an office or medical facility having access to the full range of broadband multimedia telecommunications capabilities; or located in a moving vehicle, on a golf course, or at some other remote location having limited communications access. In order to accommodate all eventualities, provisions for dynamic resource arbitration and allocation, both during "call" initiation and while the "call" is in progress, are required to ensure that the more important aspects of the interaction are fully satisfied.

II.2 Application Scenario

This scenario is provided in two parts to better represent the wide range of communication environments within which a multimedia medical consultation could take place.

II.2.1 Full multimedia support capability

Dr. "X" is a world recognized authority on bone structure and is widely consulted by other doctors on a frequent basis. Usually, this consultation takes place in Dr. "X"'s office where he has a state-of-the-art multimedia communications terminal with a large high definition video display. A typical consultation might proceed in the following manner:

Stage 1 – Dr. "X" is called by Dr. "Y" via videophone requesting consultation regarding a patient suffering from multiple fractures of the upper foot resulting from an automobile accident. After briefly covering the nature of the injury, Dr. "Y" transmits the patient's examination chart. The full screen video image on Dr. "X"'s screen immediately changes to a two-partition representation depicting the patient's chart in the left half and a reduced video image of Dr. "Y" in the right half.

Stage 2 – Dr. “Y” is on duty in the emergency room of a local hospital and, after discussing the general aspects of the case with Dr. “X” in a face-to-face videophone presentation, switches to his handheld remote videophone camera in order to provide Dr. “X” with a visual survey of the damaged foot.

Stage 3 – With the visual inspection completed, Dr. “X” requests transmission of the X-rays depicting the damaged area taken from different orientations. The two-partition screen presentation is quickly divided into four partitions, one for each of the X-rays to be transmitted.

Stage 4 – After careful scrutiny, Dr. “X” selects the partition which gives the best view of the upper ankle area where most of the serious damage appears to have occurred. The partitioned screen is quickly replaced with a full screen, high resolution depiction of the selected image, enabling Dr. “X” to make a more detailed inspection of the area of interest.

Stage 5 – Careful examination of the tarsal bone structure indicates considerable damage to the tibialis posterior tendon and associated muscle area, a complicating factor which requires the assistance of a third specialist. With the consent of Dr. “Y”, Dr. “X” initiates a videoconference call to Dr. “Z”, a specialist in tendon reconstruction.

Stage 6 – After advising Dr. “Z” of the nature of the emergency, the three doctors continue discussion of the case. As the videoconference progresses, the patient’s examination chart, medical files, X-rays and other reference information are brought into the conference as required, either through the transmission of additional data or recovered from local “memory” if previously transmitted.

Stage 7 – At the end of the conference, Dr. “Y” thanks Drs. “X” and “Z” for their assistance and terminates the consultation.

II.2.2 Restricted multimedia support capability

A week later, another emergency occurs, this time involving a patient whose foot has been crushed in a logging accident. Dr. “Y” again calls Dr. “X” for consultation. While Dr. “X” is available for consultation, it is his day off and all calls are automatically routed either to his home terminal or his portable terminal, depending upon the doctor’s location at any particular point in time. In this case, Dr. “X” happens to be on the golf course, accessible from the portable terminal in his golf cart.

In general, the consultation proceeds in a manner similar to that of the previous week. However, due to size limitations placed on the portable terminal and the reduced bandwidth available through the mobile network, service expectations are modified and focused on the more important aspects of the interaction. The less important features are relegated to a nice-to-have but non-essential category. With this in mind, the consultation proceeds in the following manner:

Stage 1 – Dr “Y” initiates a videophone call to Dr “X” to request consultation. Since Dr. “X” is now using his portable terminal, he has elected to receive calls in the “voice only” mode. The network, complying with this service request, establishes the initial connection for voice communication only.

Stage 2 – After advising Dr. “X” of the circumstances surrounding the emergency, Dr. “Y” asks Dr. “X” to switch his terminal to videophone operation in order to visually survey the area of injury. Dr. “Y”, recognizing that Dr. “X” is communicating from a portable terminal, bypasses the normal full field view camera on his videophone terminal and activates the handheld remote scanner, holding the camera steady in the vicinity of the injury to compensate for the reduced “motion” response characteristics of Dr. “X”’s portable terminal.

Stage 3 – With visual inspection completed, Dr. “X” requests transmission of an X-ray for the orientation he feels will best portray the extent of damage. To compensate for the size of the portable video display and the reduced transmission rate, Dr. “X” has purchased an enhanced storage feature for his basic portable multimedia terminal in order to capture the considerable amount of data required for high resolution X-rays. In addition, he is willing to accept longer transmission delay in order to obtain the necessary image resolution.

Stage 4 – After careful scrutiny of the damaged area, Dr. “X” requests transmission of an additional X-ray which he hopes will depict the damaged area to better advantage. He elects not to choose a split screen presentation due to the small size of the portable video display, but to take advantage of the local data storage and image manipulation features, which allow him to zoom in on areas of particular interest and to change from one locally stored image to the other at near “office” response times.

Stage 5 – Careful examination of the injury again indicates the need for additional consultation with Dr. “Z” regarding the extensive damage which has occurred to the tendons in the vicinity of the ankle. With the consent of Dr. “Y”, Dr. “X” places a voice only conference call to Dr. “Z”.

Stage 6 – After advising Dr. “Z” of the nature of the emergency and that he is calling from a mobile terminal, Dr. “X” asks Dr. “Y” to initiate a three-way videoconference to further discuss the case. In order to make maximum use of the bandwidth available for the more important imagery data, Dr. “X” elects to join the videoconference in the AUDIOGRAPHICS-only mode (audio plus still image and graphics). As the videoconference progresses, X-ray and other visual information is brought into the conference as required, either through the transmission of additional data or recovered from local memory if the information had been previously sent.

Stage 7 – At the end of the teleconference, Dr. “Y” again thanks both Drs. “X” and “Z” for their assistance and terminates the consultation.

II.3 Implementation notes

II.3.1 Related applications

This Application is closely related to REMOTE MEDICAL DIAGNOSTICS, but differs with respect to the time urgency of the interaction, the terminal facilities and transmission resources available, and the principal media of information interchange.

II.3.2 Associated applications

AUTOMATED ACCOUNTING AND BILLING for the consultants’ time, and a permanent record of the interaction (AUTOMATIC ARCHIVAL) are desirable adjuncts to this application.

II.3.3 Security/privacy

The communications associated with this application are privileged in nature and require access to databases containing confidential information protected by privacy laws in most locations.

II.3.4 Service flexibility

There is a need for automated service mechanisms which will allow for:

1) initial “call” establishment at the highest common denominator of service capabilities shared by all participants; and

2) the dynamic and selective modification of service parameters during “call” progress.

II.3.5 Performance trade-offs

The primary media components are VOICE and IMAGERY. Resolution requirements for the medical images take precedence over the associated increase in transmission delay. For portable terminal applications, resolution also takes precedence over the area of spatial coverage as long as mechanisms are provided for selecting the boundaries of the area to be viewed.

The consultation may be conducted in either a full-motion video or still-frame audiographic mode of operation, depending upon the terminal and transmission capabilities available to the participants.

3 Draft revised Annex A.3 to Recommendation F.700

FRAMEWORK RECOMMENDATION FOR AUDIOVISUAL/MULTIMEDIA SERVICES

Annex A.3- Media component Text

A.3.1 Definition

The media component Text allows for the capture and representation of information, its transfer from originating user(s) to destination user(s), its presentation to human user(s), processing, filing and retrieval.

A.3.2 Description

A.3.2.1 General description

Text is a representation medium consisting of formatted characters. It is stored and transmitted as a sequence of codes. Although it may be displayed on the same screen as video and still pictures, it requires decoding into specific fonts for presentation to the user, whether on the screen or on paper. The input is through a keyboard. The output may be a printer or a screen.

The following levels of quality are defined .

T0– minimum quality, basic alphabet and punctuation, no formatting or choice of font ;

T0 bis – videotex quality, basic alphabet and punctuation, basic graphic character set, no formatting or choice of font ;

T1– Usable text conversation quality characterised by:

Font support for ISO-10646-1 Language area Latin-1 plus the target language area for the implementation.

No more than 1 corrupted, dropped or marked missing character per 100.

Delay from character input in the transmitter to display in the receiver shorter than 2 s.

T2- Good text conversation quality characterised by

Font support for all characters in ISO-10646-1

No more than 1 corrupted, dropped or marked missing character per 500.

Delay from character input in the transmitter to display in the receiver shorter than 1 s.

A.3.2.2 Additional facilities

The user may be given control over text through editing and presentation functions. He may also be able to insert graphics, still pictures or animated pictures within the text.

A.3.2.3 Requirements for various audiovisual services

When text is the support for conversational services, the timing aspects of text entry and display are critical. Text may be transmitted and diplayed in near real time, as text is entered. It may also be transmitted only after specific end-of-sentence action or on a specific send request. In a conversation between two users, the near real time conversation is important for optimised benefit of the conversation. For multi-user conferences, a sentence based transmission may be more relevant in an open discussion, while for a subtitled speech, the real time text transmission is preferred.

For retrieval services, it may be accepted to transmit and display a whole page of text in one operation.

For conversation, editing may be reduced to “new line”, “erase last character”, while the editing for information retrieval should contain a possibility to replace text anywhere on the page and add various formatting effects to any part of text. Annotations that stand out distinctly are also desirable.

The levels of text quality required for various services are the following :

Service	T0	T0 bis	T1	T2
Telex	X			
Videotex		X		
Text telephony			X	X
Total Conversation				X
Messaging services			X	X
Retrieval services			X	X

A.3.3 Quality Aspects

The quality of text depends mainly upon the capabilities for formatting and using different types of fonts and special characters When no error correction is made, for instance in conversation, text quality is also measured in terms of corrupted characters, dropped characters and characters replaced by the missing text marker [see ITU-T Recommendation T.140].

4 Draft new Annex C.2 to Recommendation F.700

Annex C - Middleware service elements descriptions

C.2 Middleware service element *Conversation Control*

C.2.1 Definition

The generic middleware service element *Conversation Control* provides the control functions for point to point real-time exchange of information between two users

C.2.2 Description

The generic middleware service element *Conversation Control* provides the basic means for managing various types of conversations. It is related to the generic communication task conversing described in Annex B.1. It controls the operation of the various media components, allowing appropriate channels to be opened and used between the terminals.

C.2.3 Control functions

At the beginning of the call, the terminals exchange information on their capabilities and a common mode of communication is chosen. This is usually an automatic process, but the users may intervene when several modes are available or if they want to avoid the use of one or more media component (for instance a user may not want his image to be sent out, or he may want to limit the bandwidth in order to reduce the cost).

During the course of the communication, a change of mode may also be initiated, adding or suppressing a media component or changing the bandwidths allocated to the various media components.

C.2.4 Implementation

There are two possible levels of complexity for the control functions:

– Level 1 uses only the basic signals in the control channel of the multiplex, with limitations in the capacity of the channel and in the available commands;

– Level 2 uses a packet oriented data channel with a multilevel protocol defined in the T.120 series of Recommendations on which control data and user data are multiplexed; it is more flexible and offers enhanced control capabilities for optional functions.

NOTE - For ISDN, the level 1 channel is defined in Recommendations H.221 and H.230, supporting the procedures of H.242. On other networks equivalent Recommendations are H.222.0 and H.245.

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ITU-T RECOMMENDATION V.18
DRAFT Version 3

OPERATIONAL AND INTERWORKING REQUIREMENTS FOR DCEs OPERATING IN THE TEXT TELEPHONE MODE

(June 2000)

Summary

This Recommendation describes modem procedures to follow for automatic interworking with text telephones. Text telephones use various modem technologies. This Recommendation specifies the signal analysis, signal transmission and logic needed to determine what kind of text telephone there is on a connection. It also specifies the actions needed to communicate in the mode supported by each terminal type.

This Recommendation is intended for use in text telephones, in interworking units, in text relay services, in emergency centers, and in computers to be used for text telephony in the PSTN.

This Recommendation specifies transmission of identification signals to determine when the connection is between two V.18-equipped terminals. For that case, V.21 is the default modulation used. For interworking in text conversation between humans, not only the modulation must be specified. Therefore, this Recommendation specifies that when the connection in V.18 mode is established, the presentation protocol specified in Recommendation T.140 should be used, including an internationally useful character set.

The text telephone types supported by this Recommendation are: EDT, 5-bit (or Baudot), DTMF, V.21, V.23, Bell 103 and V.18-based devices.

In originate mode, V.18 identification signals and V.23 stimulation signals are transmitted until a recognized text telephone signal is received and connection can be established in that mode.

In answer mode, this Recommendation specifies stimulation to connection by transmission of probing signals for the different kinds of text telephones while monitoring for text telephone signals. Once determined, the mode of communication is entered.

For cases when it is not obvious if the connection should be made in originate or answer mode, procedures are provided to resolve that and reach communication.

An automode monitor mode is provided for cases when it is desired to have a text telephone device prepared on the same line as a voice telephone and indicate when there is an active text telephone on the connection.

For DTMF and 5-bit text telephone types using character coding not commonly used with modems, conversion is specified in this Recommendation between these codes and T.50.

For selection between multimedia protocols and this Recommendation, and also for negotiating procedures for simultaneous voice and text, modem connection procedures based on Recommendation V.8 *bis* are provided.

Source

ITU-T Recommendation V.18 was revised by ITU-T Study Group 16 (1997-2000) and was approved under the WTSC Resolution No. 1 procedure on the 6th of February 1998.

Work with version 3 of this Recommendation was initiated in February 2000 with the following goals:

1) Include all amendments collected in implementor guides.

2) Resolution of the open question from the implementors guide on the contents of the XCI signal. The contents is decided to be 0x"FF",0x"FF".

3) Specify full V.8 connection procedures, in order to allow smooth initiation by a receiving terminal and allow modulation negotiation (still maintaining interoperability with the "TXP" based connection procedures of earlier versions). This obsoletes large parts of Appendix III.

4) Remind about the risk for false detection of DTMF signals as textphone signals when operating in automode monitor mode.

5) Clarify that the probe list may contain 0-6 elements.

6) Move the V.8 *bis* procedures specified in Appendix III to the main body of the Recommendation.

7) The new Appendix III is dedicated to V.18 implementation text.

Recommendation V.18

OPERATIONAL AND INTERWORKING REQUIREMENTS FOR DCEs OPERATING IN THE TEXT TELEPHONE MODE

(*revised in 2000*)

Background

The ITU-T,

considering

a) that text telephones place special operational needs on the use of DCEs;

b) that for historical reasons, many existing text telephones do not use V-series modulation;

c) that there is a desire to have all future GSTN text telephones employ V-series modulation;

d) that to provide a migration path from the diverse installed base, it will be necessary to provide interworking with existing text telephones;

e) that to provide for interworking, the DCE will need to convert the 5-bit character code or Recommendation Q.23 (DTMF) character set used by some existing text telephones into the 7-bit code set as given in Recommendation T.50;

f) that such character conversion in the DCE will be undertaken solely to enable interworking with existing text telephones and to impose no constraints on character sets used in future text telephones;

g) that new technology is being developed that could be used to provide additional text telephone modes,

recommends the procedure below:

1 Scope

This Recommendation specifies features to be incorporated in DCEs intended for use in, or communicating with, text telephones primarily used by the deaf and hard of hearing. One of the goals of this Recommendation is to provide a platform on which a universal text telephone can be built. To accommodate this goal, procedures for interworking with identified existing text telephones are provided in clause 5. In addition, this Recommendation has the goal of identifying ways in which the multimedia Recommendations could be used to support new modes of operation or create new multi-mode text telephone devices. To accommodate this additional goal, clause 6 identifies some possible uses of this technology to support text telephony and additionally specifies requirements for multi-mode text telephone devices.

To provide for maximum flexibility, it is envisaged that any of the text telephone modes of operation specified in this Recommendation will be invoked on an as required basis using the commands specified in Recommendation V.250 or some equivalent mechanism.

It provides for:

- calling identification signals;
- no DCE-initiated disconnect;
- procedures for call establishment;
- procedures for interoperation with existing text telephones;

• specification of requirements for the use of text telephones in a multimedia environment.

2 References

The following ITU-T Recommendations and other references contain provisions which, through reference in this text, constitute provisions of this Recommendation. At the time of publication, the editions indicated were valid. All Recommendations and other references are subject to revision; all users of this Recommendation are therefore encouraged to investigate the possibility of applying the most recent editions of the Recommendations and other references listed below. A list of the currently valid ITU-T Recommendations is regularly published.

– ITU-T Recommendation Q.23 (1988), *Technical features of push-button telephone sets*.

– ITU-T Recommendation T.50 (1992), *International Reference Alphabet (IRA) (formerly International Alphabet No. 5 or IA5) – Information technology – 7-bit coded character set for information interchange*.

– ITU-T Recommendation T.140 (1998), *Protocol for multimedia application text conversation*.

– ITU-T Recommendation V.8 (1994), *Procedures for starting sessions of data transmission over the general switched telephone network*.

– ITU-T Recommendation V.8 bis (1996), *Procedures for the identification and selection of common modes of operation between Data Circuit-terminating Equipments (DCEs) and between Data Terminal Equipments (DTEs) over the general switched telephone network and on leased point-to-point telephone-type circuits*.

– ITU-T Recommendation V.21 (1984), *300 bits per second duplex modem standardized for use in the general switched telephone network*.

– ITU-T Recommendation V.23 (1988), *600/1200-baud modem standardized for use in the general switched telephone network*.

– ITU-T Recommendation V.25 (1996), *Automatic answering equipment and general procedures for automatic calling equipment on the general switched telephone network including procedures for disabling of echo control devices for both manually and automatically established calls*.

– ITU-T Recommendation V.61 (1996), *A simultaneous voice plus data modem, operating at a voice plus data signalling rate of 4800 bit/s, with optional automatic switching to data-only signalling rates of up to 14 400 bit/s, for use on the general switched telephone network and on leased point-to-point 2-wire telephone-type circuits*.

– ITU-T Recommendation V.250 (1998), *Serial asynchronous automatic dialling and control*.

– ITU-T Recommendation V.70 (08/96) - *Procedures for the simultaneous transmission of data and digitally encoded voice signals over the GSTN, or over a 2-wire leased point-to-point telephone type circuits*.

– ITU-T Recommendation H.324 (02/98) - *Terminal for low bit-rate multimedia communication*.

– ANSI TIA/EIA-825 (03/00) - *A Frequency Shift Keyed Modem for use of the Public Switched Telephone Network*.

3 Definitions

This Recommendation defines the following terms:

3.1 carrierless mode: A mode for communication, where signals are only present on the connection when data is being exchanged (e.g. in response to the pressing of a key on a keyboard).

3.2 carrier mode: A mode for communication, where continuous signals (i.e. carriers) are present on the connection irrespective of whether data is being exchanged or not.

3.3 CI: A signal transmitted from the originating DCE to indicate the general communications function, consisting of a repetitive sequence of bits at 300 bit/s, using modulation in accordance with the low-band channel defined in Recommendation V.21. The cadence of this shall be bursts of 4 CI sequences separated by 2 s of silence. The CI sequence for textphone is defined in Recommendation V.8.

3.4 CM: The Call Menue signal defined in Recommendation V.8.

3.5 JM: The Joint menue signal defined in Recommendation V.8.

3.6 multi-mode text telephone: A device which incorporates simultaneous voice and data in addition to conforming to clauses 4 and 5.

3.7 text telephone: A device incorporating text telephony functions.

3.8 text telephone mode: The operational mode when two devices are interconnected to provide text telephone communications.

3.9 text telephony: A telecommunications capability which supports real-time text conversation on communication networks.

3.10 TXP: A signal transmitted to allow early termination of answer tone, and also to confirm V.18 capability in the answering device. It consists of a repetitive sequence of bits at 300 bit/s modulating V.21(1) if transmitted from the originating DCE, or modulating V.21(2) if transmitted from the answering DCE. The 40-bit TXP sequence in left-to-right order of transmission is given by:
(1 1111 1111 1) (0) 0010 1011 (1) (0) 0001 1011 (1) (0) 0000 1010 (1) where brackets enclose start and stop bits. TXP is included for compatibility with earlier versions of V.18.

3.11 V.18 mode: The operational mode when two devices conforming to this Recommendation are interconnected to provide text telephone capability.

3.12 V.18 text telephone: A communications device conforming to the requirements of this Recommendation.

3.13 XCI: A signal transmitted in high-band V.23 modulation to stimulate V.23 terminals to respond and to allow detection of V.18 capabilities in a DCE.

The 3-s XCI signal uses the V.23 upper channel having periods of "mark" (i.e. 1300 Hz) followed by two bytes XCI marker containing the data pattern (0) 1111 1111(1)(0)1111 1111(1) sent at 1200 bit/s. The signal consists of:

– 400 ms mark;
– XCI marker;
– 800 ms mark;
– XCI marker;
– 800 ms mark;
– XCI marker;

– 800 ms mark;
– XCI marker;
– 100 ms mark.

4 Operational requirements

The DCE, when configured to support text telephone mode, shall:

1) not initiate a disconnect;
2) have the capability to be configured to automatically reassume the initial interworking state, (e.g. re-initiate the calling id signal and activate the appropriate detectors) whenever transmission has ceased for a period of 10 s (e.g. a call transfer). When this capability is not invoked, the DCE shall stay in the selected transmission mode awaiting resumption of the communication (e.g. in alternating between voice and text);
3) implement the CI signal coded as specified in this Recommendation. The use of CI is required by the calling DCE except where it is known *a priori* that the called terminal supports Recommendation V.8 *bis* (see clause 6);
4) provide call progress indications to the DTE. These signals shall include, but not be limited to: BUSY, RINGING, CARRIER, LOSS OF CARRIER and CONNECT(x) where x indicates the mode of connection (e.g. V.18, EDT, etc.);
5) implement circuit 135 – Received energy present (or its equivalent) (see Note).

NOTE – Because of the subjective nature of this indication, the operational thresholds of this circuit are left to the discretion of the implementors; however, means should be provided to prevent the presence of CI signal specified in this Recommendation from interfering with the indication of call progress signals.

5 Connecting in text telephone mode including procedures for interworking with the installed base of existing text telephones

This clause specifies procedures for connecting in text telephone mode. This includes procedures for establishing communications between two V.18 text telephones, and procedures for establishing communications between a V.18 text telephone and the existing text telephones specified in Annexes A to F. Although it is envisaged that for most connections the user will have *a priori* knowledge of the type of device being called and will preset the DCE to the correct mode, automatic procedures are provided for originating and answering calls and for connection in text mode in an established call. These procedures provided for automoding and, where required for interworking, modulation and protocol conversion.

When a connection between two V.18 text telephones is established, the DTEs shall apply the protocols and procedures specified in Annex G.

Although the obvious functionality of a DTE operating with a V.18 DCE, is to convert all forms of text telephone operation to and from the T.140 presentation protocol, for consistent application handling, this Recommendation does not specify any such conversion because it is out of scope of this DCE related Recommendation.

Section 6 specifies procedures for cases when V.8 *bis* is supported.

Recommended common procedures for user terminals using the V.18 DCE are specified in Appendix II.

Further background information on text telephony and the user requirements behind it can be found in the bibliography, listed in section 7.

5.1 Automoding originating

The following procedures assume that the DCE has been placed in the V.18 mode and that the called party is expected to be equipped with a text telephone. The procedure is defined below, and represented in Figure 1 as an aid to the reader.

5.1.1 After connecting to line, the DCE shall transmit no signal for 1 s, and then transmit V.18 identification signals starting with the CI signal as specified in this Recommendation with the ON/OFF cadence defined in clause 3. After three CI signals have been sent, the DCE shall transmit 2 s of silence followed by signal XCI. This cycle shall be repeated until terminated by one of the events described below. In summary, the transmission sequence is as follows:

Silence	1 s
CI	400 ms
Silence	2 s
CI	400 ms
Silence	2 s
CI	400 ms
Silence	2 s
XCI	3 s
Silence	1 s
CI	400 ms
Silence	2 s
etc.	

The DCE shall condition its receiver to detect the following signals:

– 2100 Hz modulated (ANSam) as defined in ITU-T Recommendation V.8;
– 2100 Hz (ANS) as defined in ITU-T Recommendation V.25;
– 2225 Hz;
– 1300 Hz;
– 1650 Hz;
– 1400 or 1800 Hz;
– DTMF tones;
– 980 or 1180 Hz (Note);
– 1270 Hz;
– 390 Hz (only when sending XCI).

NOTE – Care should be taken in the design of 980 and 1180 Hz detectors to prevent incorrect triggering by echoes of transmitted CI signals.

If any of the above signals are detected, the DCE shall stop transmitting. No disconnect timers shall be started.

During the transmission of the XCI signal, the DCE shall be conditioned to detect a 390 Hz signal. The detection of 390 Hz shall be disabled at other times during the above sequence.

5.1.2 If modulated 2100 Hz ANSam is detected, the V.8 connection procedures should be followed as explained in section 6.

5.1.3 If ANS is detected, the DCE shall stop transmitting, transmit no signal for 0.5 s, and then initiate the transmission of signal TXP in V.21(1) mode. The DCE shall then monitor for 1650 Hz, 1850 Hz, 1300 Hz and loss of ANS.

5.1.3.1 When the DCE detects the absence of ANS, it shall stop transmission of signal TXP after completion of the current TXP sequence and continue to monitor for 1650 Hz and 1300 Hz.

5.1.3.2 If the DCE detects TXP in 1650 Hz/1850 Hz, it shall connect as V.18, i.e. Recommendation V.21 with the operational characteristics given in clause 4. See Annex G.

5.1.3.3 If the DCE detects 1650 Hz for ≥ 0.5 s, it shall connect as per Annex F.

5.1.3.4 If the DCE detects 1300 Hz **only** for 1.7 s, it shall connect as per Annex E, transmitting on the 75 bit/s channel.

5.1.4 If the DCE detects 2225 Hz for 0.5 s, it shall connect as per Annex D.

5.1.5 If 1650 Hz is detected for 0.5 s, the DCE shall connect as per Annex F.

5.1.6 If 1300 Hz is detected for 1.7 s, the DCE shall connect as per Annex E, transmitting on the 75 bit/s channel.

5.1.7 If 390 Hz is received during transmission of XCI and is present during the last mark period of XCI, the mark transmission shall be extended until either 3 s of 390 Hz has been detected or the 390 Hz signal ceases. If 390 Hz was detected for 3 s, the DCE shall initiate a connection as per Annex E, transmitting on the 1200 bit/s channel.

5.1.8 If a sequence of 1400 Hz or 1800 Hz FSK signals (i.e. valid 5-bit characters) are detected, the DCE shall analyse the bit duration and connect in the appropriate signalling rate as per Annex A.

5.1.9 If Dual Tone Multi-Frequency (DTMF) signals are detected, the DCE shall connect in the DTMF mode using the character conversion and the operational characteristics specified in Annex B.

5.1.10 If 980 Hz or 1180 Hz signals are detected, the DCE shall start a 2-s timer (Tr) and attempt to determine the data signalling rate of the sequence.

5.1.10.1 If the data signalling rate is 110 bit/s, the DCE shall connect as per Annex C.

5.1.10.2 If 980 Hz only is detected for 1.5 s, the DCE shall connect as per Annex F in answer mode.

5.1.10.3 If the signal ceases for 0.4 s or timer Tr expires, the DCE shall return to monitoring, as specified in 5.1.1.

5.1.11 If 1270 Hz is detected for 0.7 s, the DCE shall connect as per Annex D in answer mode.

5.2 Automoding answering

5.2.1 When in the answer mode, the DCE shall connect to the line and condition its receiver to detect:

– V.23 high-band signals;

– 1300 Hz;

– 1400 Hz or 1800 Hz;

– DTMF tones;

– 980 Hz or 1180 Hz;
– signal CI;
– 2100 Hz;
– Modulated 2100 Hz according to ANSam specification in Recommendation V.8.
– 1270 Hz;
– 2225 Hz;
– 1650 Hz.

The 3-s timer Ta shall be started. No disconnect timers shall be started. The procedures are defined below, and are provided in Figures 2a and 2b as an aid to the reader.

5.2.2 If signal CI coded for text telephone is detected, or XCI marker in signal XCI (as described in 3.13) is detected, the DCE shall initiate transmission of answer tone, ANSam, as defined in Recommendation V.8, monitor for signals CM and TXP and start a 3-s timer (Tt).

5.2.2.1 If V.8 signal CM is detected, the V.8 procedures shall be entered to determine which call function and modulation to use. See section 6 and Annex G.

5.2.2.2 If signal TXP is detected, the DCE shall transmit no signal for 75 ± 5 ms, transmit 3 TXP sequences in V.21(2) mode, and then proceed as V.18 (i.e. Recommendation V.21 with the operational requirements specified in clause 4. See Annex G.

5.2.2.3 If Tt expires, the DCE shall return to monitoring, as specified in 5.2.1.

5.2.3 If 2100 Hz is detected for 0.7 s, the DCE shall continue to monitor for 980 Hz, 1300 Hz or 1650 Hz.

5.2.3.1 If 980 Hz is detected for 0.4 s, the DCE shall connect as per Annex F in answer mode.

5.2.3.2 If 1300 Hz is detected for 1.7 s, the DCE shall connect as per Annex E, transmitting on the 75 bit/s channel.

5.2.3.3 If 1650 Hz is detected for 0.4 s, the DCE shall connect as per Annex F in the calling mode.

5.2.4 If 980 Hz is detected, the DCE shall start a 2.7-s timer Te and monitor for 1650 Hz, 980 Hz and 1180 Hz.

5.2.4.1 If 1650 Hz is detected for 0.4 s, the DCE shall connect as per Annex F in the calling mode.

5.2.4.2 If a V.25 calling tone consisting of 980 Hz **only** for more than 470 ms but less than 730 ms is detected and followed by 1 s of silence, the DCE shall enter probing state as specified in 5.2.12.

5.2.4.3 If 980 Hz only is detected for 1.5 s, the DCE shall connect as per Annex F in answer mode.

5.2.4.4 If a low-channel V.21-modulated signal is detected, the DCE shall start a 2-s timer (Tr) and attempt to determine the data signalling rate of the data sequence.

5.2.4.4.1 If the data signalling rate is 110 bit/s, the DCE shall connect as per Annex C.

5.2.4.4.2 If the data signalling rate is 300 bit/s and it is neither CI nor TXP, the DCE shall connect as per Annex F.

5.2.4.4.3 If timer Tr expires, the DCE shall return to monitoring as specified in 5.2.1.

5.2.4.5 If timer Te expires, the DCE shall return to monitoring as specified in 5.2.1.

5.2.4.6 If CI is detected, the DCE shall continue the connection procedure according to the V.18 mode as detailed in 5.2.2, 5.2.2.1, 5.2.2.2 and 5.2.2.3 above.

5.2.5 If a sequence of 1400 Hz and 1800 Hz FSK signals (i.e. valid 5-bit characters) are detected, the DCE shall analyse the bit duration and connect in the appropriate signalling rate as per Annex A.

5.2.6 If Dual Tone Multi-Frequency (DTMF) signals are detected, the DCE shall connect in the DTMF mode using the character conversion and the operational characteristics specified in Annex B.

5.2.7 If 1270 Hz is detected for 0.7 s, the DCE shall connect as per Annex D in answer mode.

5.2.8 If 2225 Hz is detected for 1 s, the DCE shall connect as per Annex D in the calling mode.

5.2.9 If 1650 Hz is detected for 0.4 s, the DCE shall connect as per Annex F in the calling mode.

5.2.10 If 1300 Hz is detected for more than 470 ms but less than 730 ms followed by 1 s of silence, the DCE shall immediately enter the probing state specified in 5.2.12.

5.2.11 If 1300 Hz only (i.e. no XCI) is detected for 1.7 s, then the DCE shall connect as per Annex E, transmitting on the 75 bit/s channel. If XCI is detected, the DCE shall proceed as described in 5.2.2.

5.2.12 If timer Ta expires, the DCE shall enter the probing state, starting by sending ANSam, and then sending signals intended to stimulate the calling text telephone or its user to respond. The DCE shall select a mode to probe in and proceed as described in either 5.2.12.1 or 5.2.12.2 depending on the most likely scenario preset by the user (see Appendix I).

5.2.12.1 When probing in the modes specified in Annexes A or B or C, the DCE shall transmit a buffered message in the selected mode and start variable timer Tm (default 3 s) to allow for a response from the caller. The DCE shall monitor for all the signals specified in 5.2.1.

The DCE shall have a stored, user-changeable, default buffered message (e.g. V.18 pls type). Although the primary use of this stored message is to stimulate a response from a carrierless text telephone, it may also be optionally sent after a connection is established with a continuous carrier-based text telephone.

5.2.12.1.1 If any valid signal as defined in 5.2.1 is detected, the DCE shall act according to the specification in 5.2.2 to 5.2.11 and 5.2.13 with the exception that if no connection has been made within 20 s, the probing sequence shall be continued from where it was interrupted by the signal detection.

5.2.12.1.2 If timer Tm expires and no response is received, the DCE shall proceed to the next appropriate probe (e.g. next carrier mode, or next carrierless mode). If the probe list is exhausted, start again from the beginning of the appropriate list.

5.2.12.2 When probing in the modes specified in Annexes D or E or F, the DCE shall transmit ANSam with preference for the type with phase reversals, for 1 s, then remain silent for 75 ± 5 ms and then transmit, for the duration of variable timer Tc (default 6 s) depending on the mode, 1300 Hz carrier, 1650 Hz or 2225 Hz. The DCE shall monitor for all appropriate signals while transmitting one of the above carriers. When 1300 Hz is transmitted, the DCE shall also monitor for 390 Hz.

5.2.12.2.1 If 390 Hz is detected for 3 s while 1300 Hz is being sent, the DCE shall connect as per Annex E, transmitting on the 1200 bit/s channel.

5.2.12.2.2 When any other valid signal as defined in 5.2.1 is detected, the DCE shall act according to the specification in 5.2.2 to 5.2.11 with the exception that if a connect attempt from 5.2.12.2.1 or from this subclause has not succeeded within 4 s, the probing sequence shall be continued where it was interrupted by the signal detection.

5.2.12.2.3 If timer Tc expires, the DCE shall proceed to the next appropriate probe (e.g. next carrier mode or next carrierless mode). If the probe list is exhausted, start again from the beginning of the appropriate list.

5.2.12.3 If V.8 signal CM is detected, the V.8 procedures are entered to determine which call function and modulation to use. See section 6 and Annex G.

5.2.13 If V.8 signal ANSam is detected, the V.8 procedures shall be entered, in calling mode, as explained in section 6 and Annex G.

5.3 Automoding monitor mode

An automoding monitor mode shall be implemented for the purpose of detection of text telephone connection attempts from voice mode and for use in automatic voice/text answering systems.

The function of this mode is identical to the automoding answering mode as specified in 5.2, except that the timer Ta is not set and 5.2.4.2 and 5.2.10 shall not result in entering the probing state. Instead, if either of the conditions in 5.2.4.2 or 5.2.10 is detected, this shall be reported to the DTE as a V.25 calling tone.

When operating in automode monitor mode it may be desirable to handle the line interface so that parallel use for voice is enabled.

When operating in automode monitor mode, precautions must be taken not to enter textphone mode on indications of DTMF signals on the line that may occur during voice mode from other applications than text telephony.

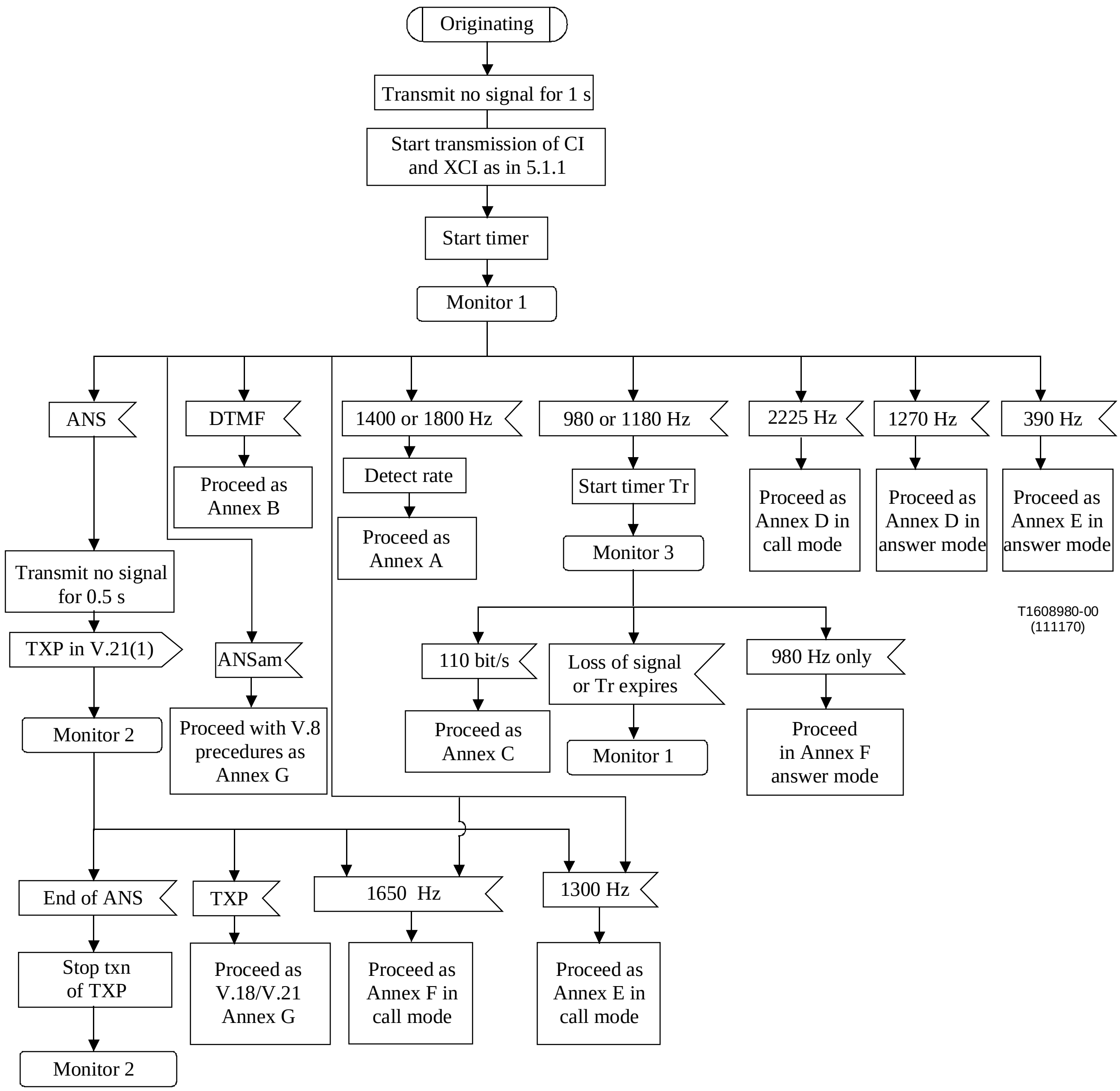
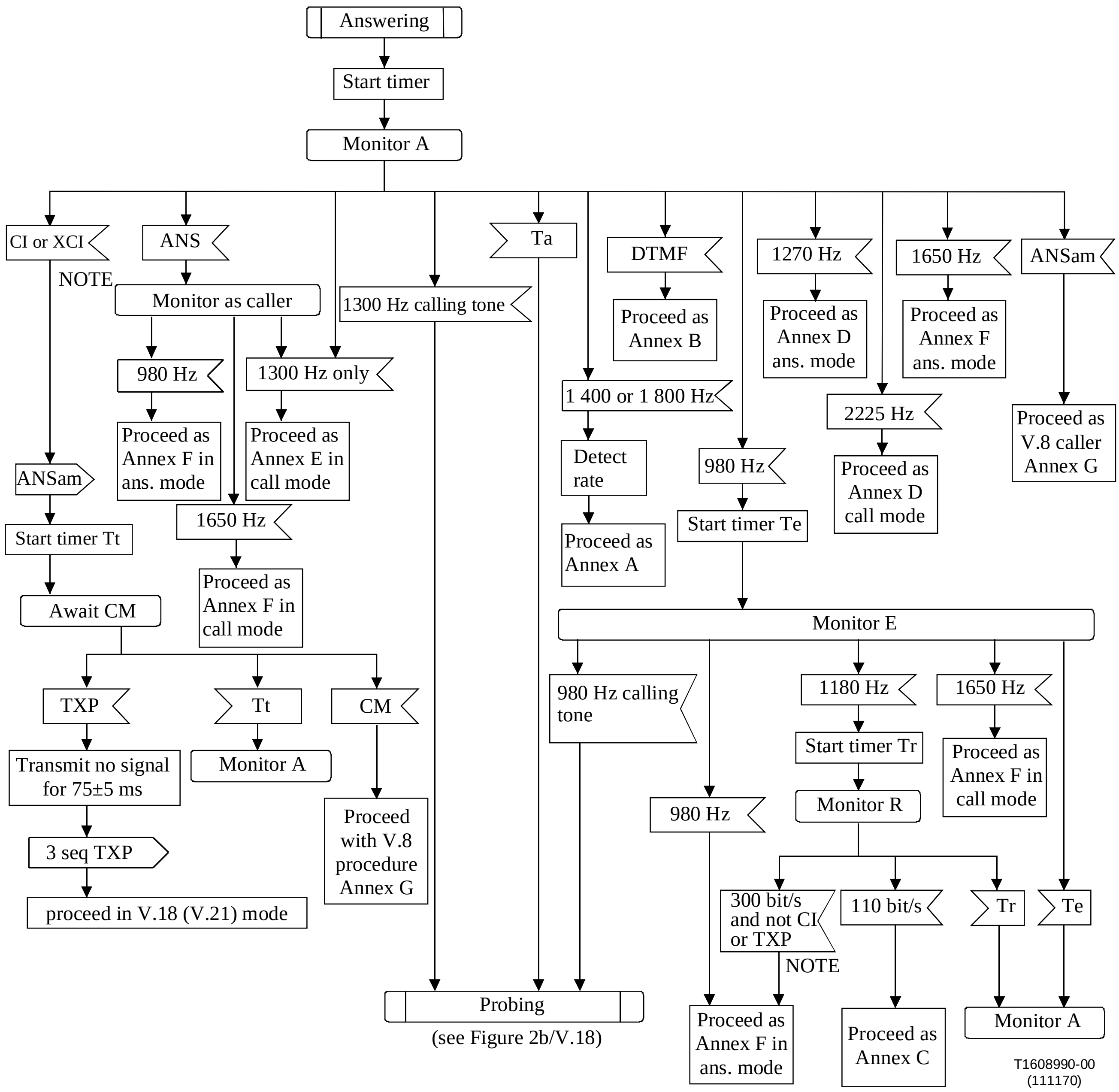


Figure 1/V.18 – Start-up procedure in the originating V.18 DCE with automoding to existing types of text telephone without use of V.8 *bis*



NOTE - The detection of CI may be integrated with the detection of other V.21(1) signals.
The detection of XCI may be integrated with the detection of other V.23(2) signals (i.e.1300 Hz).

Figure 2a/V.18 – Start-up procedure in the answering V.18 DCE showing automoding without use of V.8 *bis* procedures

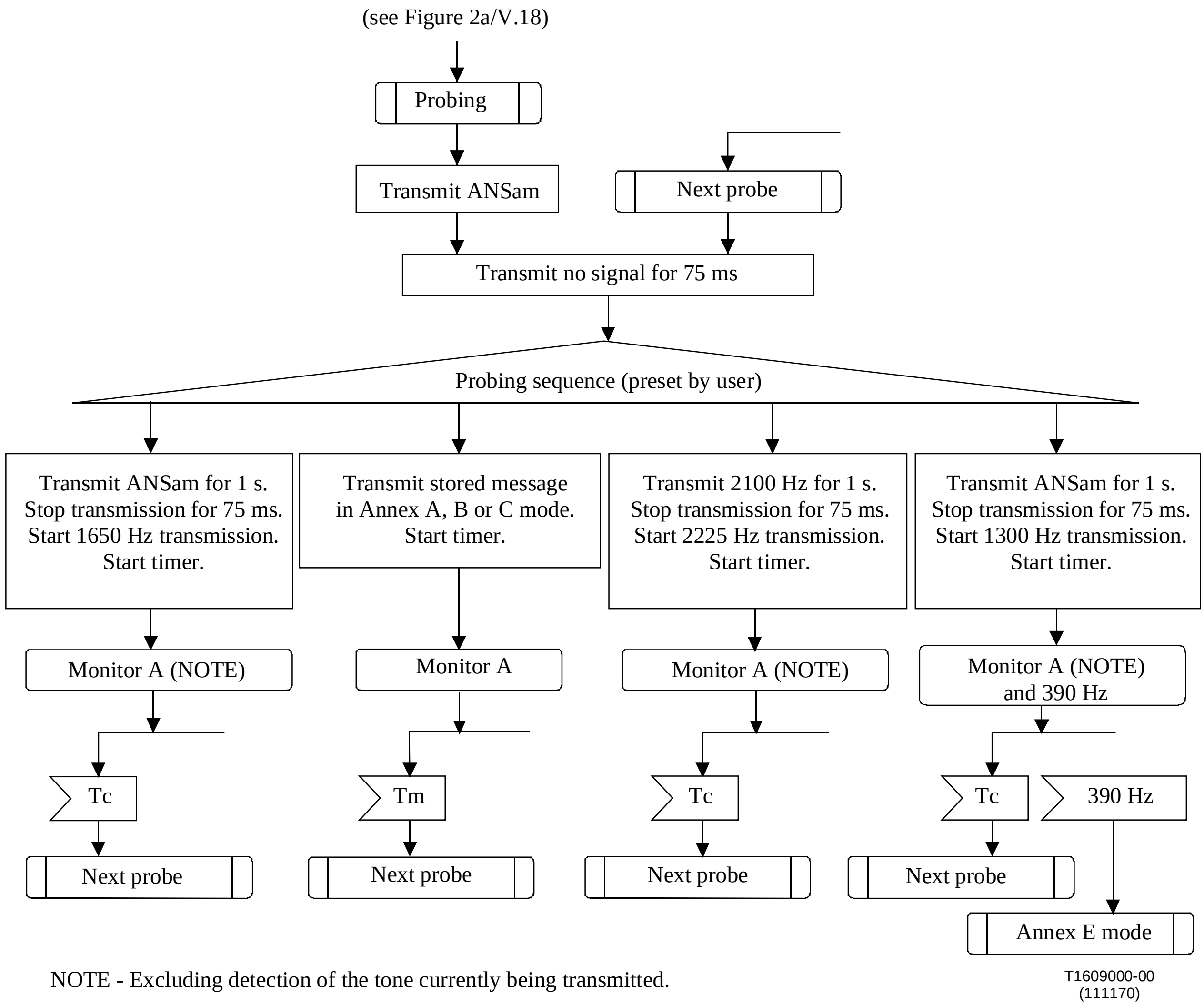


Figure 2b/V.18 – Automode probing

6 Connection in text telephone mode with V.8 and V.8 *bis* procedures

The procedures in this section are used when both DCEs have capability to perform a V.8 or V.8 *bis* negotiation. Negotiation on the reason for the call and modulation to use in the call can be performed. This enables possibilities to select optional modes of operation.

6.1 V.8 procedures

For connection in V.18 mode, using the V.8 procedures, the "textphone" call type and V.21 modulation shall always be offered. Other valid modulations may also be offered.

If the V.8 procedure results in an agreement to start a textphone session, then the connection shall proceed as V.18 with modulation selected in the V.8 procedure and the operational requirements specified in Annex G.

6.1.1 Originate mode

In originate mode, CI and XCI should be transmitted and detection of ANSam, ANS and text telephone signals should be enabled.

If ANSam is received, a CM signal should be transmitted according to the V.8 procedures and the connection procedure according to V.8 should be completed. If the text telephone function selection is completed, the selected modulation should be started and V.18 text telephone presentation protocol T.140 invoked as specified in Annex G.

If ANS is received, the TXP signal exchange is followed.

If other text telephone signals are detected, the V.18 procedures should be followed to enter a suitable mode.

6.1.2 Answer mode

In answer mode, detection of a CI with any call function or a XCI should cause ANSam to be sent.

If CM is received, the V.8 procedures should be followed to select a common call function and mode. If the selected call function is "textphone", the selected modulation should be started and V.18 text telephone presentation protocol T.140 invoked and the procedures of Annex G applied.

If "txp" is received, the original "txp" signal exchange is continued that normally ends in V.18 mode, V.21 modulation and T.140 presentation protocol as specified in Annex G.

If another text telephone signal is detected, the V.18 procedures should be followed to enter a suitable mode.

If no signal is detected within 3 seconds, the V.18 probing procedures should be initiated, still monitoring for V.8 signals.

6.1.3 V.8 procedure initiated by the answering terminal

If the V.8 sequence is started with ANSam by a DCE with the intention to start the textphone mode, the calling DCE has no indication on the purpose of the call, and may select to indicate another, unacceptable, call type in CM.

The answering DCE may then indicate the call type "textphone" in JM. The calling DCE can accept this mode by starting the DCE in the mode indicated in JM or deny it by not responding.

6.1.4 Enter text mode from voice

If the DCE is activated during a call without evident association to calling or answering, a 7-second timer should be started and the V.18 originating procedures described in 6.1.1 above should be initiated. If no text telephone signal and no V.8 signal is detected during this time, the V.18 modem should revert to answer mode as described in section 6.1.2.

NOTE - This clause is intended to address the transfer from voice mode to text. The procedure implies a small risk of connecting in one of the compatibility modes between two V.18 capable devices. The V.8 *bis* procedures should be preferred for entering text telephone mode during a voice call.

6.2 Simultaneous voice and text telephony (SVT)

The capability for Simultaneous Voice and Data (SVD) provided by Recommendations H.324, V.61 and V.70 can be used to support expanded modes of text telephony without the need for any special modifications. When this capability is added to a device that supports the provisions of clauses 4 and 5, the device shall be considered a V.18 multi-mode text telephone device. In this case, Recommendation V.8 *bis* procedures should be used, for the exchange and negotiation of capabilities as well as to provide the means for switching between supported text telephone modes and between text telephone mode and voice.

When Simultaneous Voice and Data (SVD) capability is included, in a V.18 text telephone, connections with functionality suitable for deaf, hearing-impaired, speech-impaired and hearing people are facilitated. In these cases, after the SVD capability is established, text and voice can be used simultaneously in any combination as required by the users.

NOTE – The audio channel provided by SVD DCEs (e.g. V.61, H.324) can, in many cases, support V.18 text telephony. In this case, the V.18 devices could be connected to the audio input of such devices and the text telephone connection would be established, in accordance with the provisions of clause 5, after the SVD connection is established. In this case, however, the SVD devices are not considered to be text telephone devices and therefore would not need to conform to the provisions of this Recommendation.

6.3 Connection procedures based on V.8 *bis*

ITU-T Recommendation V.8 *bis* offers possibilities to indicate more than one mode to use during the call. It also offers mechanisms for negotiating details about the selected mode. Only by completing a V.8 *bis* start-up sequence, the H.324 multimedia terminals can invoke the multilink protocol, the component selection, the encryption and the text conversation protocol T.140. V.18 has one mode for voice and text selectable through V.8 *bis*.

It is also possible to declare two or more available modes and agree on one. One example is that both V.18 and H.324 with T.140 can be declared, and any common mode for text conversation can be selected.

If Recommendation V.8 *bis* is implemented in the DCE, the following procedures should be followed.

In the V.8 *bis* procedure, a text telephone device should indicate "V.18 Text Telephone" in the V.8 *bis* parameters, and appropriate supported modulations, always including V.21. If other modes of interest for the current call are supported (e.g. H.324 with T.140), they should also be indicated, and the V.8 *bis* procedure used to select a common mode.

V.8 *bis* transactions 2 and 3 are preferred for use during a call and transactions 12 and 13 are preferred at the beginning of a call. When a V.8 *bis* sequence is completed, the procedures recommended in V.8 *bis* section 9.9 for assigning answer mode and originate mode when entering communication mode should be applied.

Examples of V.8 *bis* procedures are given in Figure 3/V.18.

6.3.1 As soon as the line put in off-hook state by any DTE on the connection, the DTE controlling the V.18 DCE should set the DCE to the automoding monitor state. The DCE should also be configured to monitor for V.8 *bis* signals.

6.3.2 If the DCE is activated in the calling mode, i.e. performs the dialling, then the V.18 originating procedures should be invoked with the following additions:

– The DTMF tones used in dialling should not cause detection as valid text telephone signals in the calling DCE.

– Configure the DCE to detect V.8 *bis* signals and text telephone signals.

If V.8 *bis* signals are detected, the DCE should perform the V.8 *bis* procedures to enter a common mode.

If text telephone signals are detected, the DCE should perform the V.18 procedures in section 5.1 to enter a common mode for text conversation.

6.3.3 If the DCE is activated within 10 seconds after a ring is detected, the DCE should monitor the line for network tones. If a ringing tone is detected, then the procedure in section 6.3.2 should be applied. This situation appears for example in the call from when the supplementary service: "Completion of call to busy subscriber" is invoked. If a ringing tone is not detected, then the V.18 answering procedures should be invoked applied as follows:

– Send V.8 *bis* signal Capability Request (CRe).

– Be configured to monitor for V.8 *bis* signals and text telephone signals.

– At detection of a CI signal or an XCI signal, start a 3-second timer, and then send CRe. If no V.8 *bis* response is detected during the 3-second timeout, or another CI or XCI is received, then continue according to the V.8 answering procedure. If a V.8 *bis* response is received, the V.8 *bis* procedure should be continued to select a common mode of operation.

6.3.4 If the DCE is activated during a call without evident association to calling or answering, a V.8 *bis* CRd signal should be sent, a timeout of 7 seconds should be set and the procedures according to paragraph 6.3.2 should be applied. If no V.8 *bis* or text telephone signals are detected during this time, the procedures according to paragraph 6.3.3 should be applied.

6.3.5 If the V.8 *bis* procedures are completed for V.18 text telephone parameters, the session procedures in Annex G shall be applied.

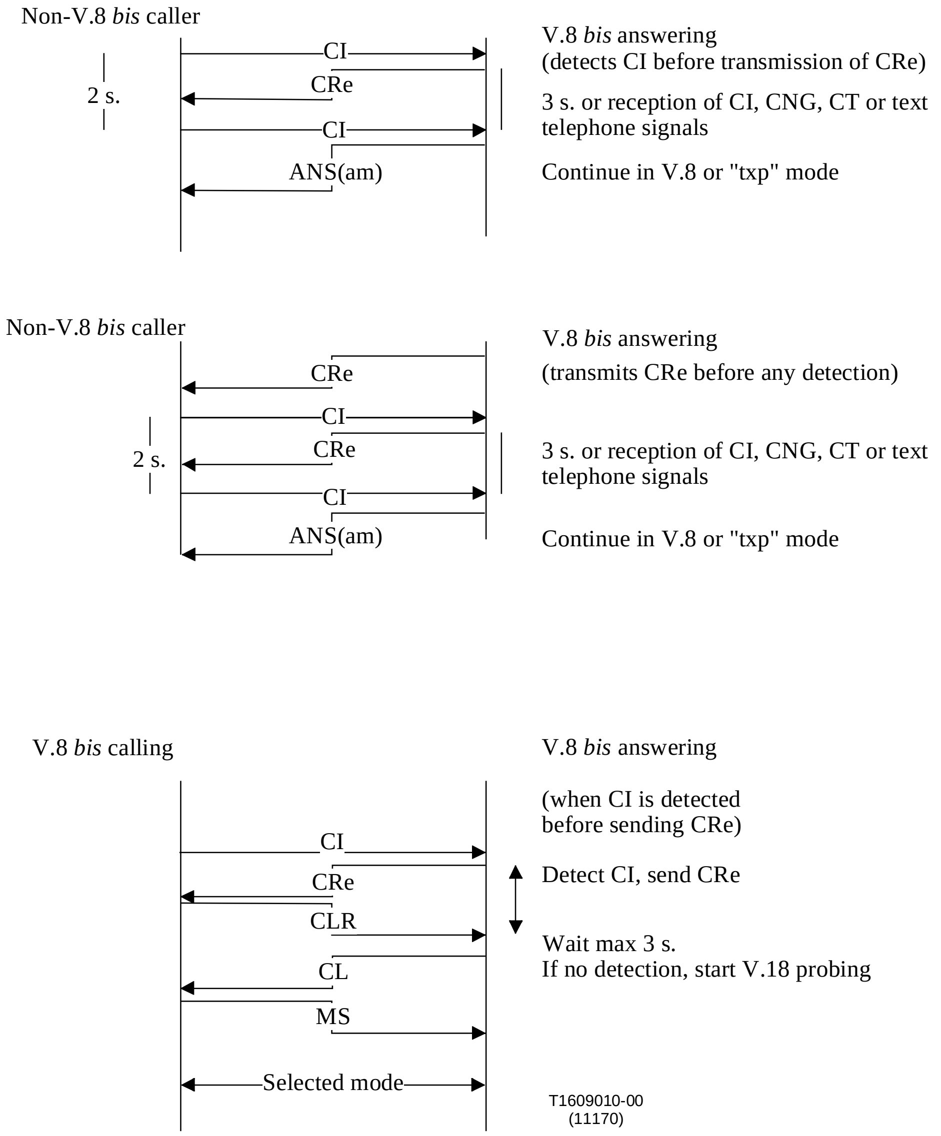


Figure 3/V.18 – Examples of sequences including V.8 *bis* capable of DCEs

7 Bibliography

Further informative reading about the background and needs of text telephony and text conversation can be found in the following document:

– ETSI ETR 333 (1997), *Text Telephony, user requirements and recommendations*.

ANNEX A

5-bit operational mode

A.1 Mode of operation

The 5-bit mode is defined in ANSI TIA/EIA-825 (03/00) A Frequency Shift Keyed Modem for use of the Public Switched Telephone Network.

The communication channel is half-duplex with no channel turnaround. Carrier is transmitted 150 ms before the first character is transmitted. The receiver shall be disabled for 300 ms when a character is transmitted to mitigate false detection of echoes (in non-V.18 devices, the carrier may remain for up to 1 s after the last character to provide this same function).

A.2 Modulation

The modulation is frequency shift-keyed modulation (i.e. no carrier is present when a character is not being transmitted) using 1400 Hz (± 5%) for a binary 1 and 1800 Hz (± 5%) for a binary 0. A bit duration of either 20 or 22.00 ± 0.40 ms is used providing either a nominal data signalling rate of 50 or 45.45 bits/s respectively.

A.3 Probe

The probe in answer mode shall be at a 47.6-bit/s data signalling rate.

A.4 Character conversion

The initial condition of the converter shall be the Letters (LTRS) mode; therefore, the DCE shall send the LTRS character (11111) to the line prior to transmitting the first translated character. The receiver decoding shall also start up in the LTRS mode. Additionally, the DCE shall send the appropriate mode character (i.e. LTRS or FIGS) every 72 characters.

The 5-bit codes supported are given in Tables A.1 and A.2. Each character shall consist of the 5-bit sequence given in the tables preceded by a one start bit and followed by a minimum of one and one-half stop bits.

The DCE shall convert the 5-bit coded characters received from the line to the appropriate 7-bit T.50 coded characters (see Table A.1/V.18) and transfer them to the DTE on circuit 104 (or its equivalent).

The DCE shall convert the 7-bit T.50 coded characters received from the DTE on circuit 103 (or its equivalent) to the appropriate 5-bit coded characters (see Table A.2/V.18) and transmit to the line.

A.5 Informative usage information

Informative comment: "GA" is the turntaking characters commonly used in English speaking environments. "GASK" is used for invitation to finish and "SKSK" as the finishing token.

"Baudot" is a term often used for the 5-bit mode. "TTY" and "TDD" are terms often used for the text telephones operating in 5-bit mode.

Table A.1/V.18 – Line-to-DTE code conversion (5-bit to 7-bit)

5-bit code	LTRS	7-bit T.50 code	5-bit code	FIGS	7-bit T.50 code
00000	(BACKSP)	000 1000	*00000*	(BACKSP)	000 1000
00001	E	100 0101	*00001*	3	011 0011
00010	LF	000 1010	*00010*	LF	000 1010
00011	A	100 0001	*00011*	–	010 1101
00100	SPACE	010 0000	*00100*	SPACE	010 0000
00101	S	101 0011	*00101*	–	000 0000
00110	I	100 1001	*00110*	8	011 1000
00111	U	101 0101	*00111*	7	011 0111
01000	CR	000 1101	*01000*	CR	000 1101
01001	D	100 0100	*01001*	$	010 0100
01010	R	101 0010	*01010*	4	011 0100
01011	J	100 1010	*01011*	'	010 0111
01100	N	100 1110	*01100*	,	010 1100
01101	F	100 0110	*01101*	!	010 0001
01110	C	100 0011	*01110*	:	011 1010
01111	K	100 1011	*01111*	(	010 1000
10000	T	101 0100	*10000*	5	011 0101
10001	Z	101 1010	*10001*	"	010 0010
10010	L	100 1100	*10010*	)	010 1001
10011	W	101 0111	*10011*	2	011 0010
10100	H	100 1000	*10100*	=	011 1101
10101	Y	101 1001	*10101*	6	011 0110
10110	P	101 0000	*10110*	0	011 0000
10111	Q	101 0001	*10111*	1	011 0001
11000	O	100 1111	*11000*	9	011 1001
11001	B	100 0010	*11001*	?	011 1111
11010	G	100 0111	*11010*	+	010 1011
11011	FIGS	(Note)	11011	FIGS	(Note)
11100	M	100 1101	*11100*	.	010 1110
11101	X	101 1000	*11101*	/	010 1111
11110	V	101 0110	*11110*	;	011 1011
11111	LTRS	(Note)	11111	LTRS	(Note)

NOTE – The translator must keep track of (e.g. toggle a memory location) the mode (i.e. LTRS, FIGS). The default should be the LTRS mode. The 7-bit T.50 character DEL (111 1111) sent from the keyboard shall force the receiving translator to the LTRS mode (see Table A.2).

Table A.2/V.18 – DTE-to-line code conversion (7-bit to 5-bit)

7-bit code	T.50 character	5-bit code	7-bit code	T.50 character	5-bit code
000 0000	NULL	NULL	100 0000	@ >> X	11101
000 0001	SOH	NULL	100 0001	A	00011
000 0010	STX	NULL	100 0010	B	11001
000 0011	ETX	NULL	100 0011	C	01110
000 0100	EOT	NULL	100 0100	D	01001
000 0101	ENQ	NULL	100 0101	E	00001
000 0110	ACK	NULL	100 0110	F	01101
000 0111	BEL	NULL	100 0111	G	11010
000 1000	BACKSPACE	*00000*	100 1000	H	10100
000 1001	HT >> SPACE	*00100*	100 1001	I	00110
000 1010	LF	*00010*	100 1010	J	01011
000 1011	VT >> LF	*00010*	100 1011	K	01111
000 1100	FF >> LF	*00010*	100 1100	L	10010
000 1101	CR	01000	100 1101	M	11100
000 1110	SO	NULL	100 1110	N	01100
000 1111	SI	NULL	100 1111	O	11000
001 0000	DLE	NULL	101 0000	P	10110
001 0001	DC1	NULL	101 0001	Q	10111
001 0010	DC2	NULL	101 0010	R	01010
001 0011	DC3	NULL	101 0011	S	00101
001 0100	DC4	NULL	101 0100	T	10000
001 0101	NAK	NULL	101 0101	U	00111
001 0110	SYN	NULL	101 0110	V	11110
001 0111	ETB	NULL	101 0111	W	10011
001 1000	CAN	NULL	101 1000	X	11101
001 1001	EM	NULL	101 1001	Y	10101
001 1010	SUB >> ?	11001	101 1010	Z	10001
001 1011	ESC	NULL	101 1011	[>> (	*01111*
001 1100	IS4 >> LF	*00010*	101 1100	\ >> /	*11101*
001 1101	IS3 >> LF	*00010*	101 1101	] >>)	*10010*
001 1110	IS2 >> LF	*00010*	101 1110	^ >> ’	*01011*
001 1111	IS1 >> SPACE	*00100*	101 1111	_ >> SPACE	*00100*
010 0000	SPACE	*00100*	110 0000	’	*00111*
010 0001	!	*01101*	110 0001	a	00011
010 0010	"	*10001*	110 0010	b	11001
010 0011	# >> $	*01001*	110 0011	c	01110
010 0100	$	*01001*	110 0100	d	01001
010 0101	% >> /	*11101*	110 0101	e	00001
010 0110	& >> +	*11010*	110 1110	f	01101
010 0111	’	*01011*	110 0111	g	11010
010 1000	(	*01111*	110 1000	h	10100
010 1001	)	*10010*	110 1001	i	00110
010 1010	*	*11100*	110 1010	j	01011
010 1011	+	*11010*	110 1011	k	01111

Table A.2/V.18 – DTE-to-line code conversion (7-bit to 5-bit) *(concluded)*

7-bit code	T.50 character	5-bit code	7-bit code	T.50 character	5-bit code
010 1100	,	*01100*	110 1100	l	10010
010 1101	-	*00011*	110 1101	m	11100
010 1110	.	*11100*	110 1110	n	01100
010 1111	/	*11101*	110 1111	o	11000
011 0000	0	*10110*	111 0000	p	10110
011 0001	1	*10111*	111 0001	q	10111
011 0010	2	*10011*	111 0010	r	01010
011 0011	3	*00001*	111 0011	s	00101
011 0100	4	*01010*	111 0100	t	10000
011 0101	5	*10000*	111 0101	u	00111
011 0110	6	*10101*	111 0110	v	11110
011 0111	7	*00111*	111 0111	w	10011
011 1000	8	*00110*	111 1000	x	11101
011 1001	9	*11000*	111 1001	y	10101
011 1010	:	*01110*	111 1010	z	10001
011 1011	;	*11110*	111 1011	{ >> (	*01111*
011 1100	< >> (	*01111*	111 1100	| >> !	*01101*
011 1101	=	*10100*	111 1101	} >>)	*10010*
011 1110	> >>)	*10010*	111 1110	~ >> SPACE	*00100*
011 1111	?	*11001*	111 1111	DEL	NULL (Note)
NOTE – Whenever the mode changes (e.g. an alphabet character is followed by a number), the translator must insert the appropriate mode code (i.e. 11011 or 11111) before transmitting the next 5-bit character code (see Table A.1). The 7-bit T.50 character DEL (111 1111) sent from the keyboard shall force the receiving translator to the LTRS mode.					

ANNEX B

DTMF operational mode

B.1 Mode of operation

The communications channel is half-duplex. The receiver is disabled for 300 ms when a character is transmitted to mitigate false detection of echoes.

B.2 Character conversion

The Q.23 (DTMF) characters supported are given in Tables B.1 and B.2. Each character shall consist of the appropriate code sequence given in the table.

The DCE shall convert the DTMF characters received from the line to their equivalent T.50-coded characters and transfer them to the DTE on circuit 104 (or its equivalent) as per Table B.1.

The DCE shall convert the T.50-coded characters received from the DTE on circuit 103 (or its equivalent) to the appropriate DTMF characters and transmit to the line as per Table B.2.

B.3 Timing

The DCE shall detect characters at least 40 ms in length with silent intervals of at least 40 ms. The DCE shall transmit DTMF characters at least 70 ms in length with silent intervals of at least 50 ms.

Table B.1/V.18 – Line-to-DTE code conversion (DTMF to 7-bit)

DTMF codes	T.50 character	7-bit code	DTMF codes	T.50 character	7-bit code
1	b	110 0010	**4	:	011 1010
2	e	110 0101	**5	%	010 0101
3	h	110 1000	**6	(	010 1000
4	k	110 1011	**7	)	011 1110
5	n	110 1110	**8	,	010 1100
6	q	111 0001	**9	LF	000 1010
7	t	111 0100	**0	NULL	NULL
8	w	111 0111	#*1	æ (Note 1)	111 1011
9	z	111 1010	#*2	ø (Note 1)	111 1100
0	SPACE	010 0000	#*3	å (Note 1)	111 1101
**1*	a	110 0001	#*4	Æ (Note 1)	101 1011
**2*	d	110 0100	#*5	Ø (Note 1)	101 1100
**3*	g	110 0111	#*6	Å (Note 1)	101 1101
**4*	j	110 1010	##*1	A	100 0001
**5*	m	110 1101	##*2	D	100 0100
**6*	p	111 0000	##*3	G	100 0111
**7*	s	111 0011	##*4	J	100 1010
**8*	v	111 0110	##*5	M	100 1101
**9*	y	111 1001	##*6	P	101 0000
**0*	BACK SPACE	000 1000	##*7	S	101 0011
#1	c	110 0011	##*8	V	101 0110
#2	f	110 1110	##*9	Y	101 1001
#3	i	110 1001	##*0	NULL	NULL
#4	l	110 1100	##1	B	100 0010
#5	o	110 1111	##2	E	100 0101
#6	r	111 0010	##3	H	100 1000
#7	u	111 0101	##4	K	100 1011
#8	x	111 1000	##5	N	100 1110
#9	.	010 1110	##6	Q	101 0001
#0	?	011 1111	##7	T	101 0100
**#1*	1	011 0001	##8	W	101 0111
**#2*	2	011 0010	##9	Z	101 1010
**#3*	3	011 0011	##0	SPACE	010 0000
**#4*	4	011 0100	###1	C	100 0011
**#5*	5	011 0101	###2	F	100 0110
**#6*	6	011 0110	###3	I	100 1001
**#7*	7	011 0111	###4	L	100 1100
**#8*	8	011 1000	###5	O	100 1111
**#9*	9	011 1001	###6	R	101 0010

Table B.1/V.18 – Line-to-DTE code conversion (DTMF to 7-bit) *(concluded)*

DTMF codes	T.50 character	7-bit code	DTMF codes	T.50 character	7-bit code
*#0	0	011 0000	###7	U	101 0101
**1	+	010 0110	###8	X	101 1000
**2	-	010 1101	###9	;	011 1011
**3	=	011 1101	###0	!	010 0001

NOTE 1 – National option.

NOTE 2 – Codes preceded by **# or *** are reserved for preprogrammed sentences and should be translated character by character to the corresponding T.50 codes.

Table B.2/V.18 – DTE-to-line code conversion (7-bit to DTMF)

7-bit code	T.50 character	DTMF	7-bit code	T.50 character	DTMF
000 0000	NULL	NULL	100 0000	@ >> X	###8
000 0001	SOH	NULL	100 0001	A	##*1
000 0010	STX	NULL	100 0010	B	##1
000 0011	ETX	NULL	100 0011	C	###1
000 0100	EOT	NULL	100 0100	D	##*2
000 0101	ENQ	NULL	100 0101	E	##2
000 0110	ACK	NULL	100 0110	F	###2
000 0111	BEL	NULL	100 0111	G	##*3
000 1000	BACKSPACE	*0	100 1000	H	##3
000 1001	HT >> SPACE	0	100 1001	I	###3
000 1010	LF	**9	100 1010	J	##*4
000 1011	VT >> LF	**9	100 1011	K	##4
000 1100	FF >> LF	**9	100 1100	L	###4
000 1101	CR	NULL	100 1101	M	##*5
000 1110	SO	NULL	100 1110	N	##5
000 1111	SI	NULL	100 1111	O	###5
001 0000	DLE	NULL	101 0000	P	##*6
001 0001	DC1	NULL	101 0001	Q	##6
001 0010	DC2	NULL	101 0010	R	###6
001 0011	DC3	NULL	101 0011	S	##*7
001 0100	DC4	NULL	101 0100	T	##7
001 0101	NAK	NULL	101 0101	U	### 7
001 0110	SYN	NULL	101 0110	V	##* 8
001 0111	ETB	NULL	101 0111	W	##8
001 1000	CAN	NULL	101 1000	X	### 8
001 1001	EM	NULL	101 1001	Y	##*9
001 1010	SUB >> ?	#0	101 1010	Z	## 9
001 1011	ESC	NULL	101 1011	Æ (Note)	#*4
001 1100	IS4 >> LF	**9	101 1100	Ø (Note)	#*5
001 1101	IS3 >> LF	**9	101 1101	Å (Note)	#*6
001 1110	IS2 >> LF	**9	101 1110	^ >> '	NULL
001 1111	IS1 >> SPACE	0	101 1111	_ >> SPACE	0

Table B.2/V.18 – DTE-to-line code conversion (7-bit to DTMF) *(concluded)*

7-bit code	T.50 character	DTMF	7-bit code	T.50 character	DTMF
010 0000	SPACE	0	110 0000	’	NULL
010 0001	!	###0	110 0001	a	*1
010 0010	"	NULL	110 0010	b	1
010 0011	# >> $	NULL	110 0011	c	#1
010 0100	$	NULL	110 0100	d	*2
010 0101	% >> /	**5	110 0101	e	2
010 0110	& >> +	**1	110 1110	f	#2
010 0111	’	NULL	110 0111	g	*3
010 1000	(	**6	110 1000	h	3
010 1001	)	**7	110 1001	i	#3
010 1010	_ >> .	#9	110 1010	j	*4
010 1011	+	**1	110 1011	k	4
010 1100	,	**8	110 1100	l	#4
010 1101	-	**2	110 1101	m	*5
010 1110	.	#9	110 1110	n	5
010 1111	/	NULL	110 1111	o	#5
011 0000	0	*#0	111 0000	p	*6
011 0001	1	*#1	111 0001	q	6
011 0010	2	*#2	111 0010	r	#6
011 0011	3	*#3	111 0011	s	*7
011 0100	4	*#4	111 0100	t	7
011 0101	5	*#5	111 0101	u	#7
011 0110	6	*#6	111 0110	v	*8
011 0111	7	*#7	111 0111	w	8
011 1000	8	*#8	111 1000	x	#8
011 1001	9	*#9	111 1001	y	*9
011 1010	:	**4	111 1010	z	9
011 1011	;	###9	111 1011	æ (Note)	#*1
011 1100	< >> (	**6	111 1100	ø (Note)	#*2
011 1101	=	**3	111 1101	å (Note)	#*3
011 1110	> >>)	**7	111 1110	~ >> SPACE	0
011 1111	?	#0	111 1111	DEL	*0
NOTE – National option.					

ANNEX C

EDT operational mode

C.1 Mode of operation

The communications channel is half-duplex. The carrier is transmitted 300 ms before the first character is transmitted. The receiver shall be disabled for 300 ms when a character is transmitted to mitigate false detection of echoes (in non-V.18 devices, the carrier may remain for up to 1 s after the last character to provide this same function).

C.2 Modulation

The modulation is frequency shift-keyed modulation using Recommendation V.21(1) frequencies. The data signalling rate is 110 bits/s.

C.3 Characters in the EDT mode

The EDT must use the following character structure. The 7-bit T.50-coded character shall be preceded by one (1) START bit and shall be followed by one EVEN PARITY bit, and two STOP bits.

NOTE – Many EDT textphones use the NAK character (decimal 21) as a backspace and delete.

ANNEX D

Bell 103 mode

D.1 Mode of operation

The communication circuit for data transmission is a duplex circuit whereby data transmission in both directions simultaneously is possible at 300 bit/s or less. The frequency of the ANS used by this DCE is 2225 Hz.

D.2 Modulation

The modulation is a binary modulation obtained by frequency shift, resulting in a modulation rate being equal to the data signalling rate.

For channel No. 1, the nominal mean frequency is 1170 Hz; for channel No. 2, it is 2125 Hz.

The frequency deviation is ± 100 Hz. In each channel, the higher characteristic frequency (FA) corresponds to a binary 1 [i.e. channel No. 1 (FA = 1270 Hz and Fz = 1070 Hz); channel No. 2 (FA = 2225 Hz and Fz = 2025 Hz)].

D.3 Character code and framing

Characters shall be coded in the US 7-bit national character set according to Recommendation T.50. Characters are framed by one start bit, 7-bit data, with one even parity bit and one stop bit. Received parity should be ignored.

D.4 Presentation control

Transmitted characters are viewed through the use of local echo. Erasure of the last character is requested by BS (0/8). New line is requested by CR LF, and erased with one BS. Local word wrapping is used at the end of line, and does not cause CR LF to be sent to the line.

D.5 Usage conventions

Many existing devices have only one common window for display of both directions of transmission. Therefore, an indicator is used to indicate when a user has finished typing and wants to give turn to the other. The most commonly used indicator for this purpose is the character string "GA".

ANNEX E

V.23 Videotex terminals

There are two main types of Videotex terminals in use for text telephony, usually known as Minitel and Prestel. The modulation is asymmetric duplex conforming to Recommendation V.23 with a 1200 bit/s forward channel and a 75 bit/s backward channel.

The characters are sent in asynchronous mode, 7-bit characters framed by one start bit, one stop bit and one even parity bit, (receive parity is ignored).

Prestel and Minitel terminals use different control sequences, and it may be necessary to distinguish between them.

E.1 Minitel terminals

E.1.1 Mode of operation

Minitel terminals must follow the 40 column Videotex mode Teletel standard with coding specified in profile 2 of the CEPT Videotex Recommendation.

When used in text telephone mode, the basic C0, G0 and G2 character sets shall be supported.

A repertoire of control sequences is defined for Minitel in accordance with Profile 2 of the CEPT Videotex protocol. A subset is required for text telephone usage. After connection, the answer mode terminal takes the initiative to set the terminals into a mode suitable for text telephony by the following control sequences. This table shows only recommended initial control sequences.

Answer mode terminal sends	**Call mode terminal responds**
Reset ($1B_{16}$,39_{16},$7F_{16}$)	Acknowledge Reset (13_{16}, $5E_{16}$)
Request scroll up mode ($1B_{16}$, $3A_{16}$,69_{16},43_{16})	Acknowledge scroll mode and lowercase mode ($1B_{16}$,$3A_{16}$,73_{16},$4A_{16}$)
Clear screen ($0C_{16}$)	

The answer mode terminal echos received characters and uses local echo to view transmitted characters. Call mode terminals do not have any echo capabilities.

E.1.2 Minitel "Dialogue" terminal

Minitel Dialogue terminals are intended for text telephone use and can operate in either call mode or answer mode, with mode selection being done automatically at connection establishment.

E.1.3 Minitel "Normal" terminal

Minitel Normal terminals operate only in call mode. The control sequences described above should be initiated by the answer mode terminal to ensure that the Minitel Normal terminal is placed in the correct mode.

E.2 Prestel terminals

Prestel terminals always operate in call mode and require the remote terminal to operate in answer mode. Like Minitel terminals, the answer mode terminal echoes received characters and uses local echo to view transmitted characters. Positive identification of a Prestel terminal may be achieved by transmission of an ENQ character which will cause an identification string to be sent if one is programmed. If there is no response to an ENQ character or the Minitel control sequences listed above, it should be assumed that the answering terminal is a Prestel terminal.

ANNEX F

V.21 text telephone mode

F.1 Mode of operation

The communication connection is 300 bit/s duplex.

F.2 Modulation

The modulation is frequency shift-keyed modulation using continuous carriers according to Recommendation V.21 frequencies.

F.3 Channel selection

Existing text telephone devices use several different ways to select the mode of operation (i.e. originate or answer). The following is a list of known methods used for resolution of mode assignments:

1) The DCE starts in answer mode and then toggles at random intervals (0.6-2.4 s) between the originate and answer modes until a carrier connection is established.

2) The DCE uses stored information and chooses its mode of operation depending on whether the device has most recently dialled or detected a ring.

In other cases, where no form of resolution is provided, the assignment of the mode of operation relies on the users selecting different modes at each end by prior agreement.

F.4 Character code and framing

Characters shall be coded in 7-bit national character sets according to Recommendation T.50. Characters are framed by one start bit, 7-bit data, with one even parity bit and one stop bit. Devices should be designed to accept one or two stop bits. Received parity should be ignored.

F.5 Presentation control

Transmitted characters are viewed through the use of local echo. Erasure of the last character is requested by BS (0/8). New line is requested by CR LF, and erased with one BS. Local word wrapping is used at the end of line, and does not cause CR LF to be sent to the line.

F.6 Usage conventions

Most existing devices have only one common window for display of both directions of transmission; therefore, an indicator is used to indicate when a user has finished typing and want to give turn to the other. The most commonly used indicators for this purpose are the "*" (e.g. in the Nordic countries) and the character string "GA" (e.g. in the United Kingdom).

ANNEX G

V.18 text telephone mode

G.1 Mode of operation

The modulation in this mode shall be in accordance with Recommendation V.21 at 300 bit/s, if no other modulation is selected in the connection procedure (see clause 6).

G.2 Presentation protocol for V.18 mode

The text conversation protocol in the DTE shall be as specified in Recommendation T.140.

G.3 Framing and transmission

Each octet sent from the T.140 protocol shall be transmitted in asynchronous mode with one start bit, one stop bit and no parity bit. Characters shall not be echoed by the receiving device.

APPENDIX I

Representative ordering of automoding

The following orderings of automoding are suggested starting points for development of probing sequences for the specified countries. Any other probing sequence can be used as appropriate for the individual situation, including sequences containing only fewer, selected modes. When selecting modes and orders, the effect on connection success and connection time should be considered.

Australia, Ireland

send 5-bit code buffered message
send V.21 carrier
send V.23 carrier
send EDT code buffered message
send DTMF buffered message
send Bell 103 carrier

Germany, Switzerland, Italy, Spain Austria

send EDT code buffered message
send V.21 carrier
send V.23 carrier
send 5-bit code buffered message
send DTMF buffered message
send Bell 103 carrier

Netherlands

send DTMF buffered message
send V.21 carrier
send V.23 carrier
send 5-bit code buffered message
send EDT buffered message
send Bell 103 carrier

Nordic countries (Iceland, Norway, Sweden, Finland, Denmark)

send V.21 carrier
send DTMF buffered message
send 5-bit code buffered message
send EDT code buffered message
send V.23 carrier
send Bell 103 carrier

UK

send V.21 carrier
send 5-bit code buffered message
send V.23 carrier
send EDT code buffered message
send DTMF buffered message
send Bell 103 carrier

USA

send 5-bit code buffered message
send Bell 103 carrier
send V.21 carrier
send V.23 carrier
send EDT code buffered message
send DTMF buffered message

France, Belgium

send V.23 carrier
send EDT buffered message
send DTMF buffered message
send 5-bit code buffered message
send V.21 carrier
send Bell 103 carrier

APPENDIX II

Recommended common procedures for terminals using the V.18 DCE

II.1 Line status display

An indication of the status of the connection should be presented, including call progress information as well as the status of circuit 135, line energy present.

II.2 Connect mode

An indication of the mode in which the connection was made (e.g. V.18, V.23, Baudot, etc.) should be provided to the user.

APPENDIX III

Specification of V.18 implementation tests

Summary

This appendix to ITU-T Recommendation V.18 contains test specifications for testing implementations of Recommendation V.18 Operational and Interworking requirements for DCEs operating in the Text Telephone Mode. It contains one small section with basic interworking tests on a functional level and one larger section with implementation test cases. The interworking test is meant to give some confidence in that there is reason to perform the most elaborate implementation tests. The tests are supposed to be supported by a semi-automatic test tool called the "tester". The tests are designed so that they verify one part of the V.18 logic each. The tests do not compose a full conformance test, but are intended to give confidence in that a V.18 implementation is made according to the Recommendation.

III.1 Introduction

Tests have been defined for the majority of possible paths through the V.18 automoding states. These include calling, called and monitor automoding operation. There are tests for character conversion. There are also tests for operational functions such as provision of indications to the DTE of call status and tests for requirements of the compatibility modes described in the annexes.

There is a group of tests for exception conditions such as immunity to voice and fax machines. These are not specifically defined in V.18 but are implicit if the Textphone Under Test (TUT) is to operate correctly under typical conditions.

There are no tests for V.8 *bis*, V.61 or for other multimedia related operation as described in V.18 section 6. These may also be added at a later date.

Compliance with this suite of tests does not guarantee operation with all versions of all textphones. Although every effort has been made to test all relevant paths through V.18, it may be that some modes of operation are not covered either due to unpredictable use of V.18 or because V.18 itself does not cater for that particular mode.

Proper end-to-end interworking in the text telephone mode relies on compatibility at the presentation level. Although there are tests for implementation of the V.18 annexes, this should not be interpreted as guaranteeing end-to-end interworking at the presentation level.

The ease of use of text telephones relies on many factors including the network interface and human factors issues in the user interface. Verification against the following tests reflects only a part of the total usability.

III.2 Definitions

TUT	Textphone Under Test
Tester	The equipment used to perform the tests
Operator	The person using the tester to perform the tests

III.3 Summary of tests

It is assumed throughout the tests that a purpose built test tool, referred to as the "tester" is available for an "operator" to perform the tests. The textphone under test is referred to as the TUT. The TUT will be connected to the tester via some kind of network simulator which may be incorporated into the tester.

Only the tests that are applicable to a particular V.18 implementation should be performed, e.g. detection of RINGING is not applicable to an acoustically coupled device.

III.3.1 Interworking Tests

There are two interworking tests. They will be performed against the BT reference implementation of V.18. This is a software implementation that runs on a PC using a purpose built DSP card to provide the necessary modem functions.

1) Automode Calling Test.

2) Automode Called Test.

III.3.2 Implementation Tests

There are five groups of implementation tests:

III.3.2.1 Operational Requirements Tests

Test Description	**Identifier**	**V.18 ref**
No Disconnection Test	MISC-01	4 (1)
Automatic resumption of automoding	MISC-02	4 (2)
Retention of selected mode on loss of signal	MISC-03	4 (2)
Detection of BUSY tone	MISC-04	4 (4)
Detection of RINGING	MISC-05	4 (4)
"LOSS OF CARRIER" indication	MISC-06	4 (4)
Call progress indication	MISC-07	4 (4)

Circuit 135 test	MISC-08	4 (5)
Connection Procedures	MISC-09	

III.3.2.2 Automode Originate Tests

Test Description	**Identifier**	**V.18 ref**
CI & XCI Signal coding and cadence	ORG-01	5.1.1
ANS Signal Detection	ORG-02	5.1.2
End of ANS signal detection	ORG-03	5.1.2.2
ANS tone followed by TXP	ORG-04	5.1.2.2
ANS tone followed by 1650 Hz	ORG-05	5.1.2.3
ANS tone followed by 1300 Hz	ORG-06	5.1.2.4
ANS tone followed by no tone	ORG-07	5.1.2
Bell 103 (2225 Hz Signal) Detection	ORG-08	5.1.3
V.21 (1650 Hz Signal) Detection	ORG-09	5.1.4
V.23 (1300 Hz Signal) Detection	ORG-10	5.1.5
V.23 (390 Hz Signal) Detection	ORG-11	5.1.6
5 Bit Mode (Baudot) Detection Tests	ORG-12 a to d	5.1.7
DTMF signal detection	ORG-13	5.1.8
EDT Rate Detection	ORG-14	5.1.9.1
Rate Detection Test	ORG-15	5.1.9.1
980 Hz Detection	ORG-16	5.1.9.2
Loss of signal after 980 Hz	ORG-17	5.1.9.3
Tr Timer	ORG-18	5.1.9.3
Bell 103 (1270 Hz Signal) Detection	ORG-19	5.1.10
Immunity to Network Tones	ORG-20	-
Immunity to other non-textphone modems	ORG-21 a, b	-
Immunity to Fax Tones	ORG-22	-
Immunity to Voice	ORG-23	-
ANSam detection	ORG-24	5.1.11
V.8 originate call	ORG-25	6.1

III.3.2.3 Automode Answer Tests

Test Description	**Identifier**	**V.18 ref**
Ta timer	ANS-01	5.2.1
CI Signal Detection	ANS-02	5.2.2
Early Termination of ANS tone	ANS-03	5.2.2.1

Tt Timer	ANS-04	5.2.2.2
ANS tone followed by 980 Hz	ANS-05	5.2.3.1
ANS tone followed by 1300 Hz	ANS-06	5.2.3.2
ANS tone followed by 1650 Hz	ANS-07	5.2.3.3
980 Hz followed by 1650 Hz	ANS-08	5.2.4.1
980 Hz calling tone detection	ANS-09 a to d	5.2.4.2
V.21 Detection by Timer	ANS-10	5.2.4.3
EDT Detection by Rate	ANS-11	5.2.4.4.1
V.21 Detection by Rate	ANS-12	5.2.4.4.2
Tr Timer	ANS-13	5.2.4.4.3
Te Timer	ANS-14	5.2.4.5
5 Bit Mode (Baudot) Detection Tests	ANS-15 a to d	5.2.5
DTMF Signal Detection	ANS-16	5.2.6
Bell 103 (1270 Hz signal) detection	ANS-17	5.2.7
Bell 103 (2225 Hz signal) detection	ANS-18	5.2.8
V.21 Reverse Mode (1650 Hz) Detection	ANS-19	5.2.9
1300 Hz Calling Tone Discrimination	ANS-20 a to d	5.2.10
V.23 Reverse Mode (1300 Hz) Detection	ANS-21	5.2.11
1300 Hz with XCI Test	ANS-22	
Stimulate Mode Country Settings	ANS-23	5.2.12
Stimulate Carrierless Mode Probe Message	ANS-24	5.2.12.1
Interrupted Carrierless Mode Probe	ANS-25	5.2.12.1.1
Stimulate Carrier Mode Probe Time	ANS-26	5.2.12.2
V.23 Mode (390 Hz) Detection	ANS-27	5.2.12.2.1
Interrupted Carrier Mode Probe	ANS-28	5.2.12.2.2
Stimulate Mode Response During Probe	ANS-29	5.2.12.2.2
Immunity to Network Tones	ANS-30	
Immunity to Fax Calling Tones	ANS-31	
Immunity to Voice	ANS-32	
V.8 CM detection and V.8 Answering	ANS-33	5.2.2.1

III.3.2.4 Automode Monitor Tests

For the following tests the TUT must be set to monitor mode as defined in V.18 section 5.3 "Automode Monitor Mode".

Test Description	Identifier	V.18 ref
Repeat all answer mode tests excluding tests ANS-01, ANS-20 and ANS-23 to ANS-29	MON-01 to 20	5.3
Automode Monitor Ta timer	MON-21	5.3
Automode Monitor 1300 Hz Calling Tone Discrimination	MON-22 a to d	5.3
Automode Monitor 980 Hz Calling Tone Discrimination	MON-23 a to d	5.3

III.3.2.5 V.18 Annexes Tests

For the following tests verify the requirements specified in Annexes A to F of V.18.

Test Description	Identifier	V.18 ref
Baudot carrier timing and receiver disabling	X-01	A.1
Baudot bit rate confirmation	X-02	A.2
Baudot probe bit rate confirmation	X-03	A.3
5 Bit to T.50 Character Conversion	X-04	A.4
DTMF receiver disabling	X-05	B.1
DTMF character conversion	X-06	B.2
EDT carrier timing and receiver disabling	X-07	C.1
EDT bit rate and character structure	X-08	C.2-3
V.23 calling mode character format	X-09	E
V.23 answer mode character format	X-10	E
V.21 character structure	X-11	F.4-5
V.18 mode	X-12	G.1-3

III.4 Interworking Tests Description

III.4.1 Introduction

The interworking tests ensure that the Textphone Under Test (TUT) interworks satisfactorily with the reference V.18 Text Telephone. These tests are intended to eliminate any implementation with serious errors and/or faulty equipment and to demonstrate the interworking integrity of the TUT. Further they provide an opportunity to test the acoustic coupling and/or PSTN interface of the TUT.

No measure of quality is applied in these tests. The aim is simply to gain sufficient confidence to merit continuation of the tests.

III.4.2 Test Methodology

The TUT is set up in a working configuration and connected to the Tester possibly via a network simulator. No delays or errors are inserted into the link, so that high quality, trouble free operation should be achievable.

III.4.3 Test Cases

Only two types of tests are performed:

1) A call is made from the TUT set up in the Automode calling mode to the reference V.18 text telephone.

2) A call is made from the reference V.18 text telephone to the TUT in Automode Answer configuration.

In both cases the terminals should both arrive in V.18 mode in less than 5 seconds after the call is answered. It should then be possible to perform a text conversation correctly at least with the minimal character set and the editing operations specified in T.140.

III.5 V.18 Implementation Tests description

III.5.1 Introduction

This group of tests verifies that the TUT protocol implementation conforms to the V.18 specification.

III.5.2 Test Methodology

The TUT is set up in a working configuration and connected to the Tester via a suitable interface. This might be a direct PSTN connection or an acoustic coupler.

III.5.3 Test Case Identifier Numbers

The structure of each case identified number is as follows:

<group>- <number>

where group can be:

- *MISC, Operational Requirements and other tests.*
- *ANS, Automode Answer Tests.*
- *ORG, Automode Originate Tests.*
- *MON, Automode Monitor Tests.*
- *X, V.18 Annex Tests.*

III.5.4 Test Cases

III.5.4.1 Operational Requirements Tests

III.5.4.1.1 No Disconnection Test

Identifier: MISC-01

Purpose: To verify that the DCE does not initiate a disconnection.

Preamble: N/A

Method: A call is made to the TUT from the tester which remains off hook for 10 minutes without sending any signal.

Pass criteria: The TUT should answer the call and enter the probing state after 3 seconds. The TUT should continue to probe until the test is terminated.

Comments: This feature should also be verified by observation during the automoding tests.

III.5.4.1.2 Automatic Resumption of Automoding

Identifier: MISC-02

Purpose: To ensure that the DCE can be configured to automatically re-assume the automode calling state after 10 s of no valid signal.

Preamble: The TUT should be configured to automatically re-assume the initial automoding state.

Method: The tester should set up a call to the TUT in V.21 mode and then drop the carrier. The tester will then transmit silence for 11 seconds followed by a 1300 Hz tone for 5 seconds (i.e. V.23).

Pass criteria:
1) Ten seconds after dropping the carrier the TUT should return to state Monitor 1.
2) After 2.7±0.3 seconds the TUT should select V.23 mode and send a 390 Hz tone.

Comments: The TUT should indicate that carrier has been lost at some time after the 1650 Hz signal is lost.

III.5.4.1.3 Retention of selected mode on loss of signal

Identifier: MISC-03

Purpose: To ensure that the DCE stays in the selected transmission mode if it is not configured to automatically re-assume the initial automoding state.

Preamble: The TUT should be configured to remain in the selected transmission mode when the carrier is lost.

Method: The tester should set up a call to the TUT in V.21 mode, for example. It will drop the carrier for 9 seconds and then re-start transmission of the same carrier for 1 second followed by a short message.

Pass criteria:
1) The TUT should resume operation in V.21 mode and capture the entire test message.

Comments: The TUT should indicate that carrier has been lost at some time after the carrier signal is removed and not disconnect.

III.5.4.1.4 Detection of BUSY tone

Identifier: MISC-04

Purpose: To ensure that the DCE provides the call progress indication "BUSY" in presence of the national busy tone.

Preamble: N/A

Method: The TUT should be configured to dial out and then be presented with the appropriate national busy tone.

Pass criteria: Detection of busy tone should be displayed by the TUT.

Comments: V.18 specifies that the DCE should not hang up, but that is intended to apply to the case where a connection is established and then lost. A terminal may automatically hang up when busy tone is detected. PABX busy tones may differ in frequency and cadence from national parameters.

III.5.4.1.5 Detection of RINGING

Identifier: MISC-05

Purpose: To ensure that the DCE provides the call progress indication "RINGING" in presence of the national ringing tone.

Preamble: N/A

Method: The tester will make a call to the TUT using the nationally recommended cadence and the minimum recommended ring voltage/current.

Pass criteria: The RINGING condition should be visually indicated by the TUT.

Comments: This test should be repeated across a range of valid timings and ring voltages.

III.5.4.1.6 "LOSS OF CARRIER" indication

Identifier: MISC-06

Purpose: To ensure that the DCE provides the call progress indication "LOSS OF CARRIER" upon a loss of carrier in full duplex modes, i.e. V.21, V.23, Bell 103.

Preamble: N/A

Method: Set up a call in each of the full duplex modes and force a carrier failure to the TUT.

Pass criteria: Loss of carrier should be indicated and disappear when the carrier is restored.

Comments: The V.18 modem should not automatically disconnect when used in a manual conversation mode. However, a V.18 equipped terminal may disconnect based on operational decisions, e.g. when it is a terminal in automatic answering machine mode. There may be other cases, e.g. where the V.18 DCE is used in a gateway, when automatic disconnection is required.

III.5.4.1.7 Call Progress Indication

Identifier: MISC-07

Purpose: To ensure that the DCE provides the call progress indication "CONNECT(x)" upon a connection.

Preamble: N/A

Method: Correct CONNECT messages should be verified during the Automode tests that follow.

Pass criteria: The relevant mode should be indicated by the DCE when automoding is complete. However, this may possibly not be indicated by the DTE.

Comments: The possible modes are: V.21, V.23, Baudot 45, Baudot 50, EDT, Bell 103, DTMF.

III.5.4.1.8 Circuit 135 Test

Identifier: MISC-08

Purpose: To ensure that the DCE implements Circuit 135 or an equivalent way of indicating presence of a signal.

Preamble: N/A

Method: A call from the TUT should be answered in voice mode after 20 seconds. The tester will transmit sampled voice messages. V.24 circuit 135 or its equivalent should be observed.

Pass criteria: The ring tone and speech shall be indicated by Circuit 135.

Comment: The response times and signal level thresholds of Circuit 135 are not specified in V.18 or V.24 and therefore the pattern indicated may vary.

III.5.4.1.9 Connection Procedures

Identifier: MISC-09

Purpose: To ensure that the TUT implements the call connect procedure described in section 6 of V.18.

Preamble: N/A

Method: TBD

Pass criteria: TBD

Comment: TBD

III.5.4.2 Automode Originate Tests

In this group of tests, the TUT is placed in the automode originate mode, while the tester emulates the operation of the answering station.

III.5.4.2.1 CI and XCI Signal Coding and Cadence

Identifier: ORG-01

Purpose: To verify that TUT correctly emits the CI and XCI signals with the ON/OFF cadence defined in section 5.1.1 of Recommendation V.18.

Preamble: N/A

Method: V.21 demodulator is used to decode the CI sequence and a timer to measure the silence intervals between them. The XCI signal is also monitored and decoded for to check for correct coding and timing of the signal.

Pass criteria:

1) No signal should be transmitted for one second after connecting to the line.
2) Four CI patterns are transmitted for each repetition.
3) No signal is transmitted for two seconds after the end of each CI.
4) Each CI must have the correct bit pattern.
5) The CI patterns followed by two seconds of silence must be repeated twice.
6) One second after every 3 blocks CI an XCI signal must be transmitted.
7) The XCI should have the structure defined in V.18 section 3.11.
8) The whole sequence should be repeated until the call is cleared.
9) When V.18 to V.18, the XCI must not force V.23 or Minitel mode.

Comments:

III.5.4.2.2 ANS Signal Detection

Identifier: ORG-02

Purpose: To verify that TUT correctly detects the ANS (2100 Hz) signal during the two-second interval (T_{off}) between transmission of CI sequences.

Preamble: Make a V.18 call from the TUT.

Method: The Test System waits for the TUT to stop transmitting a CI and responds with an ANS signal. The V.21 demodulator is used to decode the TXP sequence and a timer measures the silence intervals between them. ANS should be transmitted for 2 seconds.

Pass criteria:
1) No signal should be transmitted by TUT for 0.5 seconds from detection of ANS.
2) The TUT should reply with transmission of TXP as defined in section 5.1.2 of V.18.
3) Verify that TXP sequence has correct bit pattern.

Comments:

III.5.4.2.3 End of ANS Signal Detection

Identifier: ORG-03

Purpose: The TUT should stop sending TXP at the end of the current sequence when the ANS tone ceases.

Preamble: Test ORG-02 should be successfully completed immediately prior to this test.

Method: The tester sends ANS for 2 seconds followed by silence. The tester will then monitor for cessation of TXP at the end of the answer tone.

Pass criteria: The TUT should stop sending TXP at the end of the current sequence when ANS tone ceases.

Comments:

III.5.4.2.4 ANS Tone Followed by TXP

Identifier: ORG-04

Purpose: To check correct detection of V.18 modem.

Preamble: Tests ORG-02 and ORG-03 should be successfully completed prior to this test.

Method: Tester transmits ANS for 2.5 seconds followed by 75 ms of no tone then transmits 3 TXP sequences using V.21 (2) and starts a 1 s timer. It will then transmit 1650 Hz for 5 seconds.

Pass criteria:
1) TUT should initially respond with TXP.
2) TUT should stop sending TXP within 0.2 seconds of end of ANS.
3) TUT should respond with 980 Hz carrier within 1 second of end of 3 TXP sequences.
4) Data should be transmitted and received according to T.140 to comply with the V.18 operational requirements.

Comments: The TUT should indicate that V.18 mode has been selected.

III.5.4.2.5 ANS Tone Followed by 1650 Hz

Identifier: ORG-05

Purpose: To check correct detection of V.21 modem upper channel when preceded by answer tone and to confirm discrimination between V.21 and V.18 modes.

Preamble: Tests ORG-02 and ORG-03 should be successfully completed prior to this test.

Method: Tester transmits ANS for 2.5 seconds followed by 75 ms of no tone then transmits 1650 Hz and starts a 0.7 second timer.

Pass criteria:

1) TUT should initially respond with TXP.
2) TUT should stop sending TXP within 0.2 seconds of end of ANS.
3) TUT should respond with 980 Hz at 0.5(+0.2-0.0) seconds of start of 1650 Hz.
4) Data should be transmitted and received at 300 bit/s complying with Annex F.

Comments: Selection of V.21 as opposed to V.18 should be confirmed by examination of TUT. If there is no visual indication, verify by use of T.50 for V.21 as opposed to UTF-8 coded ISO 10646 character set for V.18.

III.5.4.2.6 ANS Tone Followed by 1300 Hz

Identifier: ORG-06

Purpose: To check correct detection of V.23 modem upper channel when preceded by answer tone.

Preamble: Tests ORG-02 and ORG-03 should be successfully completed prior to this test.

Method: Tester transmits ANS for 2.5 seconds followed by 75 ms of no tone then transmits 1300 Hz and starts a 2.7 s timer.

Pass criteria:

1) TUT should initially respond with TXP.
2) TUT should stop sending TXP within 0.2 seconds of end of ANS.
3) TUT should respond with 390 Hz after 1.7(+0.2-0.0) seconds of start of 1300 Hz.
4) Data should be transmitted and received at 75 bit/s and 1200 bit/s respectively by the TUT to comply with Annex E.

Comments: The TUT should indicate that V.23 mode has been selected.

III.5.4.2.7 ANS Tone Followed by No Tone

Identifier: ORG-07

Purpose: To confirm that TUT does not lock up under this condition.

Preamble: Tests ORG-02 and ORG-03 should be successfully completed prior to this test.

Method: Tester transmits ANS for 2.5 seconds followed by no tone for 10 s. It then transmits DTMF tones for 2 seconds.

Pass criteria: 1) TUT should initially respond with TXP.

2) TUT should stop sending TXP within 0.2 seconds of end of ANS.

3) TUT should return to Monitor 1 state and then connect in DTMF mode within 12 seconds of the end of ANS tone.

Comments: This condition would cause the terminal to lock up if the V.18 standard is followed literally. It may however, occur when connected to certain Swedish textphones if the handset is lifted just after the start of an automatically answered incoming call.

III.5.4.2.8 Bell 103 (2225 Hz Signal) Detection

Identifier: ORG-08

Purpose: To verify that the TUT correctly detects the Bell 103 upper channel signal during the 2-second interval between transmission of CI sequences.

Preamble: N/A

Method: The tester waits for a CI and then sends a 2225 Hz signal for 5 seconds.

Pass criteria: 1) The TUT should respond with a 1270 Hz tone in 0.5±0.1 seconds.

2) Data should be transmitted and received at 300 bit/s to comply with Annex D.

Comments: The TUT should indicate that Bell 103 mode has been selected.

III.5.4.2.9 V.21 (1650 Hz Signal) Detection

Identifier: ORG-09

Purpose: To verify that the TUT correctly detects the V.21 upper channel signal during the 2-second interval between transmission of CI sequences.

Preamble: N/A

Method: The tester waits for a CI and then sends a 1650 Hz signal for 5 seconds.

Pass criteria: 1) The TUT should respond with a 980 Hz tone in 0.5±0.1 seconds.

2) Data should be transmitted and received at 300 bit/s to comply with Annex F.

Comments: The TUT should indicate that V.21 mode has been selected.

III.5.4.2.10 V.23 (1300 Hz Signal) Detection

Identifier: ORG-10

Purpose: To verify that the TUT correctly detects the V.23 upper channel signal during the 2-second interval between transmission of CI sequences.

Preamble: N/A

Method: The tester waits for a CI and then sends a 1300 Hz signal for 5 seconds.

Pass criteria: 1) The TUT should respond with a 390 Hz tone in 1.7±0.1 seconds.

2) Data should be transmitted and received at 75 bit/s and 1200 bit/s respectively by the TUT to comply with Annex E.

Comments: The TUT should indicate that V.23 mode has been selected.

III.5.4.2.11 V.23 (390 Hz Signal) Detection

Identifier: ORG-11

Purpose: To confirm correct selection of V.23 reverse mode during sending of XCI.

Preamble: N/A

Method: The tester should wait for the start of the XCI signal and then send 390 Hz to TUT for 5 seconds.

Pass criteria:

1) The TUT should complete the XCI as normal.
2) The TUT should then maintain the 1300 Hz tone while the 390 Hz test tone is present.
3) Data should be transmitted and received at 1200 bit/s and 75 bit/s respectively by the TUT to comply with Annex E when connection is indicated.

Comments: The TUT should indicate that V.23 mode has been selected at least 3 seconds after the start of the 390 Hz tone.

III.5.4.2.12 5 Bit Mode (Baudot) Detection Tests

Identifier: ORG-12 (a) to (d)

Purpose: To confirm detection of Baudot modulation at various bit rates that may be encountered.

Preamble: N/A

Method: The tester transmits the 5-bit coded characters "0" to "9" followed by "abcdef" at (a) 45.45, (b) 47.6, (c) 50 and (d) 100 bits per second. When TUT indicates a connection, type at least 5 characters back to the tester so that correct selection of bit rate can be confirmed.

Pass criteria:

1) TUT should select Baudot mode and the appropriate bit rate.
2) The tester will analyse the bit rate of received characters, which should be at either 45.45 or 50 bits per second as appropriate.

Comments: 45.45 and 50 bit/s are the commonly used Baudot bit rates. However, certain textphones can operate at higher rates (e.g. 100 bit/s). Responding at either 45.45 or 50 bit/s is acceptable to these devices which normally fall back to the selected rate.

47.6 bit/s may possibly be encountered from another V.18 textphone in the automode answer state. The TUT may then select either 45.45 or 50 bit/s for the transmission.

III.5.4.2.13 DTMF signal detection

Identifier: ORG-13

Purpose: To verify whether the TUT correctly recognizes DTMF signals during the 2-second interval between transmission of CI.

Preamble: N/A

Method: The tester will send a single DTMF tone of 40 ms duration to TUT. When TUT indicates a connection, type at least 5 characters back to the tester so that correct selection of mode can be confirmed.

Pass criteria: The tester will analyse the received characters to confirm DTMF mode selection.

Comments: TUT should indicate that it has selected DTMF mode. The DTMF capabilities of the TUT should comply with ITU-T Recommendation Q.24 for the Danish Administration while receiving for best possible performance.

III.5.4.2.14 EDT Rate Detection

Identifier: ORG-14

Purpose: To confirm detection of EDT modems by detecting the transmission rate of received characters.

Preamble: N/A

Method: The tester transmits EDT characters "abcdef" to TUT at 110 bit/s. When TUT indicates that the connection is established, type characters "abcdef<CR>" back to the tester. The same characters will then be transmitted back to the TUT.

Pass criteria: Ensure correct reception of characters by tester and TUT.

Comments: The TUT should be able to determine the rate on the six characters given. If it takes more than this then performance is probably inadequate as too many characters would be lost. Some characters may be lost during the detection process. However, the number lost should be minimal. The data bits and parity are specified in V.18 Annex C.

III.5.4.2.15 Rate Detection Test

Identifier: ORG-15

Purpose: To verify the presence of 980/1180 Hz at a different signalling rate than 110 bit/s returns the TUT modem to the "monitor A" state.

Preamble:

Method: The tester transmits 980/1180 Hz signals at 300 bit/s for 2 seconds.

Pass criteria: The TUT should not select EDT or any other mode and should continue to transmit the CI signal.

Comments: Echoes of the CI sequences may be detected at 300 bit/s.

III.5.4.2.16 980 Hz Detection

Identifier: ORG-16

Purpose: To confirm correct selection of V.21 reverse mode.

Preamble: N/A

Method: The tester sends 980 Hz to TUT for 5 seconds.

Pass criteria: 1) TUT should respond with 1650 Hz tone after 1.5±0.1 seconds after start of 980 Hz tone.

2) Data should be transmitted and received at 300 bit/s complying with Annex F.

Comments: The TUT should indicate that V.21 mode has been selected.

III.5.4.2.17 Loss of signal after 980 Hz

Identifier: ORG-17

Purpose: To confirm that TUT returns to the Monitor 1 state if 980 Hz signal disappears.

Preamble: N/A

Method: The tester sends 980 Hz to TUT for 1.2 seconds followed by silence for 5 seconds.

Pass criteria: TUT should not respond to the 980 Hz tone and resume sending CI signals after a maximum of 2.4 seconds from the end of the 980 Hz tone.

Comments:

III.5.4.2.18 Tr Timer

Identifier: ORG-18

Purpose: To confirm that TUT returns to the Monitor 1 state if Timer Tr expires.

Preamble: N/A

Method: The tester sends 980 Hz to TUT for 1.2 seconds followed by 1650 Hz for 5 seconds with no pause.

Pass criteria: TUT should respond with 980 Hz after 1.3±0.1 seconds of 1650 Hz.

Comments: This implies timer Tr has expired 2 seconds after the start of the 980 Hz tone and then 1650 Hz has been detected for 0.5 seconds.

III.5.4.2.19 Bell 103 (1270 Hz Signal) Detection

Identifier: ORG-19

Purpose: To confirm correct selection of Bell 103 reverse mode.

Preamble: N/A

Method: The tester sends 1270 Hz to TUT for 5 seconds.

Pass criteria:
1) TUT should respond with 2225 Hz tone after 0.7±0.1 s.
2) Data should be transmitted and received at 300 bit/s complying with Annex D.

Comments: The TUT should indicate that Bell 103 mode has been selected.

III.5.4.2.20 Immunity to Network Tones

Identifier: ORG-20

Purpose: To ensure that the TUT does not interpret network tones as valid signals.

Preamble: N/A

Method: The tester will first send a dial tone to the TUT, this will be followed by a ringing tone and a network congestion tone. The frequencies and cadences of the tones will vary according to the country setting. The tester must be configured for the same country as the TUT.

Pass criteria: The countries supported by the TUT should be noted along with the response to each tone. The tones should either be ignored or reported as the relevant network tone to the user.

Comments: V.18 is required to recognize and report RINGING and BUSY tones. Other network tones may be ignored. Some devices may only provide a visual indication of the presence and cadence of the tones for instance by a flashing light. The TUT may disconnect on reception of tones indicating a failed call attempt.

III.5.4.2.21 Immunity to non-textphone modems

Identifier: ORG-21 (a) and (b)

Purpose: To ensure that the TUT does not interpret modem tones not supported by V.18 as valid text telephone tones.

Preamble: N/A

Method: The tester will respond with an ANS tone (2100 Hz) followed by simulated (a) V.32 *bis* and (b) V.34 modem training sequences.

Pass criteria: The tones should either be ignored or reported back to the user. No textphone modem should be selected.

Comments: Some high speed modems may fall back to a compatibility mode, e.g. V.21 or V.23 that should be correctly detected by the TUT.

III.5.4.2.22 Immunity to Fax Tones

Identifier: ORG-22

Purpose: To ensure that the TUT will not interpret a called fax machine as being a textphone.

Preamble: N/A

Method: The tester will respond as if it were a typical group 3 fax machine in automatic answer mode. It should send a CED tone (2100 Hz) plus Digital Identification Signal (DIS) as defined in T.30.

Pass criteria: The TUT should ignore the received tones.

Comments: Ideally the TUT should detect the presence of a fax machine and report it back to the user.

III.5.4.2.23 Immunity to Voice

Identifier: ORG-23

Purpose: To ensure that the TUT does not misinterpret speech as a valid textphone signal.

Preamble: N/A

Method: The tester will respond with sampled speech. A number of phrases recorded from typical male and female speakers will be transmitted. This will include a typical network announcement.

Pass criteria: The TUT should ignore the speech.

Comments: Ideally the TUT should report the presence of speech back to the user, e.g. via Circuit 135.

III.5.4.2.24 ANSam Signal Detection

Identifier: ORG-24

Purpose: To verify that TUT correctly detects the ANSam (2100 Hz modulated) signal during the two-second interval (T_{off}) between transmission of CI sequences.

Preamble: Make a V.18 call from the TUT.

Method: The Test System waits for the TUT to stop transmitting a CI and responds with an ANSam signal. The V.21 demodulator is used to decode the CM sequence. ANSam should be transmitted for 2 seconds.

Pass criteria:
1) No signal should be transmitted by TUT for 0.5 seconds from detection of ANSam.
2) The TUT should reply with transmission of CM as defined in section 5.1.13 of V.18.
3) Verify that CM sequence has correct bit pattern.

Comments:

III.5.4.2.25 V.8 calling procedure

Identifier: ORG-25

Purpose: To verify that TUT correctly performs a V.8 call negotiation.

Preamble: Make a V.18 call from the TUT. Answer with ANSam from the Tester and with JM for V.21 on the CM.

Method: The Test System waits for the TUT to start transmitting V.21 carrier (1).

Pass criteria: 1) The TUT should connect by sending V.21 carrier (1).

Comments:

III.5.4.3 Automode Answer Tests

For the tests in this section a call must be established from the tester to the TUT. All tests, except where stated otherwise, will commence 0.5 seconds after the call is answered to ensure that the actions are begun before timer Ta expires within the TUT. This implies that the tester must detect when the TUT goes off hook.

III.5.4.3.1 Ta Timer

Identifier: ANS-01

Purpose: To ensure that on connecting the call, the DCE starts timer **Ta** (3 seconds) and on expiry begins probing.

Preamble: N/A

Method: The tester makes a call to the TUT and attempts to determine when the TUT answers the call. It will then monitor for any signal.

Pass criteria: The TUT should start probing 3 seconds after answering the call.

Comments:

III.5.4.3.2 CI Signal Detection

Identifier: ANS-02

Purpose: To confirm the correct detection and response to the V.18 CI signal.

Preamble: N/A

Method: The tester will transmit 2 sequences of 4 CI patterns separated by 2 seconds. It will monitor for ANS and measure duration.

Pass criteria:
1) The TUT should respond after either the first or second CI with ANSam tone.
2) ANSam tone should remain for 3 seconds ±0.5 s followed by silence.

Comments: The ANSam tone is a modulated 2100 Hz tone. It may have phase reversals. The XCI signal is tested in a separate test.

III.5.4.3.3 Early Termination of ANSam tone

Identifier: ANS-03

Purpose: To confirm that the TUT will respond correctly to TXP signals, i.e. by stopping ANSam tone on reception of TXP signal.

Preamble: N/A

Method: The tester will transmit 2 sequences of 4 CI patterns separated by 2 seconds. On reception of the ANSam tone the tester will wait 0.5 seconds and then begin transmitting the TXP signal in V.21 (1) mode.

Pass criteria:
1) On reception of the TXP signal, the TUT should remain silent for 75±5 ms.
2) The TUT should then transmit 3 TXP sequences in V.21(2) mode.
3) The 3 TXPs should be followed by continuous 1650 Hz.
4) Correct transmission and reception of T.140 data should be verified after the V.18 mode connection is completed.

Comments: The TUT should indicate V.18 mode.

III.5.4.3.4 Tt Timer

Identifier: ANS-04

Purpose: To ensure that after detection of ANSam the TUT will return to Monitor A after timer Tt expires.

Preamble: Successful completion of test ANS-03.

Method: After completion of test ANS-03 the tester will continue to monitor for signals.

Pass criteria: The TUT should start probing 3 seconds after ANSam disappears.

Comments: It is assumed that timer Ta is restarted on return to Monitor A.

III.5.4.3.5 ANS Tone Followed by 980 Hz

Identifier: ANS-05

Purpose: To check correct detection of V.21 modem lower channel when preceded by answer tone.

Preamble: N/A

Method: Tester transmits ANS for 2.5 seconds followed by 75 ms of no tone then transmits 980 Hz and starts a 1 s timer.

Pass criteria: TUT should respond with 1650 Hz within 400±100 ms of start of 980 Hz.

Comments: The TUT should indicate that V.21 mode has been selected.

III.5.4.3.6 ANS Tone Followed by 1300 Hz

Identifier: ANS-06

Purpose: To check correct detection of V.23 modem upper channel when preceded by answer tone.

Preamble: N/A

Method: Tester transmits ANS for 2.5 seconds followed by 75 ms of no tone then transmits 1300 Hz and starts a 2-s timer.

Pass criteria: TUT should respond with 390 Hz after 1.7(+0.2-0.0) seconds of start of 1300 Hz.

Comments: The TUT should indicate that V.23 mode has been selected.

III.5.4.3.7 ANS Tone Followed by 1650 Hz

Identifier: ANS-07

Purpose: To check correct detection of V.21 modem upper channel when preceded by answer tone and to confirm discrimination between V.21 and V.18 modes.

Preamble: N/A

Method: Tester transmits ANS for 2.5 seconds followed by 75 ms of no tone then transmits 1650 Hz and starts a 1-second timer.

Pass criteria: TUT should respond with 980 Hz within 400±100 ms of start of 1650 Hz.

Comments: The TUT should indicate that V.21 mode has been selected.

III.5.4.3.8 980 Hz followed by 1650 Hz

Identifier: ANS-08

Purpose: To ensure the correct selection of V.21 modem channel when certain types of Swedish textphones are encountered.

Preamble: N/A

Method: The tester will simulate a call from a Diatext2 textphone that alternates between 980 Hz and 1650 Hz until a connection is made.

Pass criteria: The TUT should respond with the appropriate carrier depending on when it connects.

Comments: The TUT should indicate a V.21 connection. The time for which each frequency is transmitted is random and varies between 0.64 and 2.56 seconds.

III.5.4.3.9 980 Hz calling tone detection

Identifier: ANS-09 (a) to (d)

Purpose: To confirm correct detection of 980 Hz calling tones as defined in V.25.

Preamble: N/A

Method: The tester will send bursts of 980 Hz signals (a) 400 ms, (b) 500 ms, (c) 700 ms and (d) 800 ms followed by 1 second of silence.

Pass criteria:
1) The TUT should not respond to bursts of 400 or 800 ms.
2) The TUT should immediately begin probing after a burst of 980 Hz for 500 or 700 ms followed by 1 second of silence.

Comments: The probe sent by the TUT will depend on the country setting.

III.5.4.3.10 V.21 detection by timer

Identifier: ANS-10

Purpose: To confirm correct selection of V.21 calling modem when the received signal is not modulated, i.e. there is no 1180 Hz.

Preamble: N/A

Method: The tester sends 980 Hz to TUT for 2 seconds.

Pass criteria: The TUT should respond with a 1650 Hz tone in 1.5±0.1 seconds.

Comments: The TUT should indicate that V.21 mode has been selected.

III.5.4.3.11 EDT Detection by Rate

Identifier: ANS-11

Purpose: To confirm detection of EDT modems by detecting the transmission rate of received characters.

Preamble: N/A

Method: The tester transmits EDT characters "abcdef" to TUT at 110 bit/s. When TUT indicates that the connection is established, type characters "abcdef<CR>" back to the tester. The same characters will then be transmitted back to the TUT.

Pass criteria: Ensure correct reception of characters by tester and TUT.

Comments: The TUT should indicate that EDT mode has been selected. Some characters may be lost during the detection process. However, the number lost should be minimal. The data bits and parity are specified in V.18 Annex C.

III.5.4.3.12 V.21 Detection by Rate

Identifier: ANS-12

Purpose: To confirm detection of V.21 modem low channel by detecting the transmission rate of received characters and to ensure correct discrimination between V.18 and V.21 modes.

Preamble: N/A

Method: The tester transmits characters "abcdef" to TUT using V.21(1) at 300 bit/s. When TUT indicates that the connection is established, type characters "abcdef<CR>" back to the tester. The same characters will then be transmitted back to the TUT.

Pass criteria: Ensure correct reception of characters by tester and TUT.

Comments: This situation is unlikely to occur in practice unless the DCE is sending a V.21 (1650 Hz) probe. However, it is catered for in V.18. It is more likely that this is where CI or TXP characters would be detected (see test ANS-02).

III.5.4.3.13 Tr Timer

Identifier: ANS-13

Purpose: To ensure that the TUT returns to the Monitor A state on expiry of timer Tr (2 seconds). Timer Tr is started when a modulated V.21(1) signal is detected.

Preamble: N/A

Method: The tester will transmit 980 Hz for 200 ms followed by alternating 980 Hz/1180 Hz at 110 bit/s for 100 ms followed by 980 Hz for 1 second.

Pass criteria: The TUT should begin probing 4±0.5 seconds after the 980 Hz signal is removed.

Comments: It is not possible to be precise on timings for this test since the definition of a "modulated signal" as in V.18 section 5.2.4.4 is not specified. Therefore it is not known exactly when timer Tr will start. It is assumed that timer Ta is restarted on re-entering the Monitor A state.

III.5.4.3.14 Te Timer

Identifier: ANS-14

Purpose: To ensure that the TUT returns to the Monitor A on expiry of timer Te (2.7 seconds). Timer Te is started when a 980 Hz signal is detected.

Preamble: N/A

Method: The tester will transmit 980 Hz for 200 ms followed silence for 7 s.

Pass criteria: The TUT should begin probing 5.5±0.5 seconds after the 980 Hz signal is removed.

Comments: It is assumed that timer Ta (3 seconds) is restarted on re-entering the Monitor A state.

III.5.4.3.15 5 Bit Mode (Baudot) Detection Tests

Identifier: ANS-15 (a) to (d)

Purpose: To confirm detection of Baudot modulation at various bit rates that may be encountered.

Preamble: N/A

Method: The tester transmits the 5-bit coded characters "0" to "9" followed by "abcdef" at (a) 45.45, (b) 47.6, (c) 50 and (d) 100 bits per second. When TUT indicates a connection, type at least 5 characters back to the tester so that correct selection of bit rate can be confirmed.

Pass criteria: 1) The TUT should select Baudot mode and the appropriate bit rate.

2) The tester will analyse the bit rate of received characters, which should be at an appropriate rate, and confirm the carrier on/off times before and after the characters.

Comments: 45.45 and 50 bit/s are the commonly used Baudot bit rates. However, some textphones can transmit at higher rates e.g. 100 bit/s. Responding at either 45.45 or 50 bit/s is acceptable to these devices which then fall back to the selected rate.

A rate of 47.6 bit/s may be encountered from another V.18 textphone in the automode answer state. The TUT may then select either 45.45 or 50 bit/s for the transmission.

III.5.4.3.16 DTMF Signal Detection

Identifier: ANS-16

Purpose: To verify whether the TUT correctly recognizes DTMF signals.

Preamble: N/A

Method: The tester will send a single DTMF tone of 40 ms duration to TUT. When TUT indicates a connection, type at least 5 characters back to the tester so that correct selection of mode can be confirmed.

Pass criteria: Tester will analyse the received characters to confirm DTMF mode selection.

Comments: The TUT should indicate that it has selected DTMF mode. The DTMF capabilities of the TUT should comply with ITU-T Recommendation Q.24 for the Danish Administration.

III.5.4.3.17 Bell 103 (1270 Hz signal) detection

Identifier: ANS-17

Purpose: To ensure correct detection and selection of Bell 103 modems.

Preamble: N/A

Method: The tester sends 1270 Hz to TUT for 5 seconds.

Pass criteria: TUT should respond with 2225 Hz tone after 0.7±0.1 s.

Comments: The TUT should indicate that Bell 103 mode has been selected.

III.5.4.3.18 Bell 103 (2225 Hz signal) detection

Identifier: ANS-18

Purpose: To ensure correct detection and selection of Bell 103 modems in reverse mode.

Preamble: N/A

Method: The tester sends 2225 Hz to TUT for 5 seconds.

Pass criteria: The TUT should respond with 1270 Hz after 1±0.2 seconds.

Comments: The TUT should indicate that Bell 103 mode has been selected. Bell 103 modems use 2225 Hz as both answer tone and higher frequency of the upper channel.

III.5.4.3.19 V.21 Reverse Mode (1650 Hz) Detection

Identifier: ANS-19

Purpose: To ensure correct detection and selection of V.21 reverse mode.

Preamble: N/A

Method: The tester sends 1650 Hz to TUT for 5 seconds.

Pass criteria: The TUT should respond with 980 Hz after 0.4±0.2 seconds.

Comments: The TUT should indicate that V.21 mode has been selected.

III.5.4.3.20 1300 Hz Calling Tone Discrimination

Identifier: ANS-20 (a) to (d)

Purpose: To confirm correct detection of 1300 Hz calling tones as defined in V.25.

Preamble: N/A

Method: The tester will send 1300 Hz bursts of (a) 400 ms, (b) 500 ms, (c) 700 ms and (d) 800 ms followed by 1 second of silence.

Pass criteria:
1) The TUT should not respond to bursts of 400 or 800 ms.
2) The TUT should immediately begin probing after a burst of 1300 Hz for 500 or 700 ms followed by 1 second of silence.

Comments: The probe sent by the TUT will depend on the country setting.

III.5.4.3.21 V.23 Reverse Mode (1300 Hz) Detection

Identifier: ANS-21

Purpose: To ensure correct detection and selection of V.23 reverse mode.

Preamble: N/A

Method: The tester sends 1300 Hz only, with no XCI signals, to TUT for 5 seconds.

Pass criteria: The TUT should respond with 390 Hz after 1.7±0.1 seconds.

Comments: The TUT should indicate that V.23 mode has been selected.

III.5.4.3.22 1300 Hz with XCI Test

Identifier: ANS-22

Purpose: To ensure correct detection of the XCI signal and selection of V.18 mode.

Preamble: N/A

Method: The tester sends XCI signal as defined in V.18 section 3.11. On reception of ANS it will become silent for 500 ms then transmit the TXP signal in V.21(1) mode.

Pass criteria: The TUT should respond with TXPusing V.21(2) and select V.18 mode.

Comments:

III.5.4.3.23 Stimulate Mode Country Settings

Identifier: ANS-23

Purpose: To ensure that the TUT steps through the probes in the specified order for the country selected.

Preamble: The TUT should be configured for each of the possible probe orders specified in V.18 Appendix 1 in turn.

Method: The tester will call the TUT, wait for Ta to expire and then monitor the probes sent by the TUT.

Pass criteria: The TUT should use the orders described in Appendix 1.

Comments: The order of the probes is not mandatory.

III.5.4.3.24 Stimulate Carrierless Mode Probe Message

Identifier: ANS-24

Purpose: To ensure that the TUT sends the correct probe message for each of the carrierless modes.

Preamble:

Method: The tester will call the TUT, wait for Ta to expire and then monitor the probes sent by the TUT.

Pass criteria: The TUT should send the user defined probe message for Annexes A, B, and C modes followed by a pause of Tm (default 3) seconds.

Comments: The carrierless modes are those described in Annexes A, B and C.

III.5.4.3.25 Interrupted Carrierless Mode Probe

Identifier: ANS-25

Purpose: To ensure that the TUT continues probing from the point of interruption a maximum of 20 s after a failed connect attempt.

Preamble: The TUT should be configured for the UK country setting.

Method: The tester will call the TUT, wait for Ta to expire and then during the pause after the first Baudot probe it will send a 200 ms burst of 1270 Hz followed by silence for 30 s.

Pass criteria: The TUT should transmit silence on detecting the 1270 Hz tone and then continue probing starting with the V.23 probe 20 seconds after the end of the 1270 Hz signal.

Comments:

III.5.4.3.26 Stimulate Carrier Mode Probe Time

Identifier: ANS-26

Purpose: To ensure that the TUT sends each carrier mode for time Tc (default 6 seconds) preceded by the correct answer tone.

Preamble: None.

Method: The tester will call the TUT, wait for Ta to expire and then monitor the probes sent by the TUT.

Pass criteria: The TUT should send the ANS tone (2100 Hz) for 1 second followed by silence for 75±5 ms and then the 1650 Hz, 1300 Hz and 2225 Hz probes for time Tc.

Comments: The carrier modes are those described in Annexes D, E, and F.

III.5.4.3.27 V.23 Mode (390 Hz) Detection

Identifier: ANS-27

Purpose: To confirm correct selection of V.23 mode.

Preamble: N/A

Method: The tester waits until the 1300 Hz probe is detected from the TUT and then transmits 390 Hz for 11 seconds.

Pass criteria:
1) After 3 seconds of the 390 Hz signal the TUT should indicate that V.23 has been selected.
2) The tester will confirm that the 1300 Hz carrier is maintained for at least 4 seconds beyond the normal probe duration, i.e. Tc (=6 s default) + 4 s = 10 seconds total.

Comments: All known V.23 devices need to receive 1300 Hz tone before they will respond with 390 Hz. When the 1300 Hz probe is not being transmitted, a 390 Hz tone may be interpreted as a 400 Hz network tone.

III.5.4.3.28 Interrupted Carrier Mode Probe

Identifier: ANS-28

Purpose: To ensure that the TUT continues probing from the point of interruption a maximum of 4 s after a failed connect attempt.

Preamble: The TUT should be configured for the UK country setting.

Method: The tester will call the TUT, wait for Ta to expire and then during the first V.21 probe it will send a 200 ms burst of 1270 Hz followed by silence for 30 s.

Pass criteria: The TUT should transmit silence on detecting the 1270 Hz tone and then continue probing with the Baudot stored message 4 seconds after the end of the 1270 Hz burst.

Comments: It is most likely that the TUT will return to probing time Ta (3 seconds) after the 1270 Hz tone ceases. This condition needs further clarification.

III.5.4.3.29 Stimulate Mode Response During Probe

Identifier: ANS-29

Purpose: To ensure that the TUT is able to detect an incoming signal while transmitting a carrier mode probe.

Preamble:

Method: The tester will step through each possible response as defined in tests ANS-08 to ANS-23 for each of the carrier mode probes and for each pause after a carrierless mode probe message.

Pass criteria: The TUT should respond as described in the appropriate test above.

Comments: The TUT may not respond to any signals while a carrierless mode probe is being sent since these modes are half duplex.

III.5.4.3.30 Immunity to Network Tones

Identifier: ANS-30

Purpose: To ensure that the TUT does not interpret network tones as valid signals.

Preamble: N/A

Method: The tester will first send a busy tone to the TUT this will be followed by a number unobtainable tone. The frequencies and cadences of the tones will vary according to the country setting. The tester must be configured for the same country as the TUT.

Pass criteria: The countries supported by the TUT should be noted along with the response to each tone. The tones should either be ignored or reported as the relevant network tone to the user.

Comments: V.18 is required to recognize and report RINGING and BUSY tones. Other network tones may be ignored. Some devices may only provide a visual indication of the presence and cadence of the tones for instance by a flashing light.

III.5.4.3.31 Immunity to Fax Calling Tones

Identifier: ANS-31

Purpose: To determine whether the TUT can discriminate fax calling tones.

Preamble:

Method: The tester will call the TUT and send the fax calling tone, CNG. This is an 1100 Hz tone with cadence of 0.5 seconds ON and 3 seconds OFF as defined in T.30.

Pass criteria: The TUT should not respond to this signal and may report it as being a calling fax machine.

Comments: This is an optional test as detection of the fax calling tone is not required by V.18.

III.5.4.3.32 Immunity to Voice

Identifier: ANS-32

Purpose: To ensure that the TUT does not misinterpret speech as a valid textphone signal.

Preamble: N/A

Method: The tester will respond with sampled speech. A number of phrases recorded from typical male and female speakers will be transmitted. This will include a typical network announcement.

Pass criteria: The TUT should ignore the speech.

Comments: Ideally the TUT should report the presence of speech back to the user. This is an optional test.

III.5.4.3.33 CM Detection and V.8 Answering

Identifier: ANS-33

Purpose: To confirm that the TUT will respond correctly to CM signals and connect according to V.8 procedures.

Preamble: N/A

Method: The tester will transmit 2 sequences of 4 CI patterns separated by 2 seconds. On reception of the ANSam tone the tester will wait 0.5 seconds and then begin transmitting the CM signal with textphone and V.21 specified.

Pass criteria:

1) On reception of the CM signal, the TUT should transmit JM with textphone and V.21.
2) The TUT should then transmit in V.21(2) mode.
3) The JM should be followed by continuous 1650 Hz.
4) Correct transmission and reception of T.140 data should be verified after the V.18 mode connection is completed.

Comments: The TUT should indicate V.18 mode.

III.5.4.4 Automode Monitor Tests

For the following tests the TUT must be set to monitor mode as defined in V.18 section 5.3 "Automoding Monitor Mode". These tests should be the same as the Automode Answer case except that the stimulate tests are not applicable and the calling tone test should result only in the TUT reporting the fact that a calling tone has been detected.

III.5.4.4.1 Automode Monitor Ta Timer Test

Identifier: MON-21

Purpose: To ensure that on entering monitor mode, timer Ta (3 seconds) is not active and that the TUT does not enter the probing state.

Preamble: N/A

Method: The TUT should be put into monitor state. The tester will then monitor for signals for 1 minute.

Pass criteria: The TUT should not start probing.

Comments:

III.5.4.4.2 Automode Monitor 1300 Hz Calling Tone Discrimination

Identifier: MON-22 (a) to (d)

Purpose: To confirm correct detection and reporting of 1300 Hz calling tones as defined in V.25.

Preamble: N/A

Method: The tester will send 1300 Hz bursts of (a) 400 ms, (b) 500 ms, (c) 700 ms and (d) 800 ms followed by 1 second of silence.

Pass criteria:
1) The TUT should not respond to bursts of 400 or 800 ms.
2) The TUT should report detection of calling tones to the DTE after a burst of 1300 Hz for 500 or 700 ms followed by 1 second of silence.

Comments: In automode answer, the 1300 Hz calling causes the DCE to start probing. In monitor mode it should only report detection to the DTE.

III.5.4.4.3 Automode Monitor 980 Hz Calling Tone Discrimination

Identifier: MON-23 (a) to (d)

Purpose: To confirm correct detection and reporting of 980 Hz calling tones as defined in V.25.

Preamble: N/A

Method: The tester will send 980 Hz bursts of (a) 400 ms, (b) 500 ms, (c) 700 ms and (d) 800 ms followed by 1 second of silence.

Pass criteria:
1) The TUT should not respond to bursts of 400 or 800 ms.
2) The TUT should report detection of calling tones to the DTE after a burst of 980 Hz for 500 or 700 ms followed by 1 second of silence.

Comments: In automode answer, the 980 Hz calling causes the DCE to start probing. In monitor mode it should only report detection to the DTE.

III.5.4.5 V.18 Annexes Tests

The following tests verify features required in Annexes A to F of V.18.

III.5.4.5.1 Baudot carrier timing and receiver disabling

Identifier: X-1

Purpose: To verify that the TUT sends unmodulated carrier for 150 ms before a new character and disables its receiver for 300 ms after a character is transmitted.

Preamble: Establish a call between the tester and TUT in Baudot mode.

Method: The operator should send a single character from the TUT. The tester will immediately start sending a unique character sequence. Examination of the TUT display will show when its receiver is re-enabled.

Pass criteria:
1) The TUT should send unmodulated carrier for 150 ms before the beginning of the start bit.
2) The receiver should be re-enabled after 300 ms.

3) The tester will confirm that 1 start bit and at least 1.5 stop bits are used.

Comments: The carrier should be maintained during the 300 ms after a character.

III.5.4.5.2 Baudot bit rate confirmation

Identifier: X-2 (a) and (b)

Purpose: To verify that the TUT uses the correct bit rates in the Baudot mode.

Preamble: Establish a call between the tester and TUT in Baudot mode for each of the two tests.

Method: The operator should select Baudot (a) 45 bit/s followed by (b) 50 bit/s modes and transmit the string "abcdef" at each rate.

Pass criteria: The tester will measure the bit timings and confirm the rates.

Comments:

III.5.4.5.3 Baudot probe bit rate confirmation

Identifier: X-3

Purpose: To verify that the TUT uses the correct bit rates in the Baudot mode probe during automoding.

Preamble: Set the user defined carrierless mode probe message to the string "abcdef" if possible. Set the TUT country setting to "United States". A call should be initiated from the tester to the TUT.

Method: The tester will wait for the Baudot mode probe and measure the bit rate.

Pass criteria: The tester will measure the bit timings and confirm the rate of 47.6 bit/s.

Comments: The probe message must be long enough for the tester to establish the bit rate. "GA" may not be sufficient.

III.5.4.5.4 5 Bit to T.50 Character Conversion

Identifier: X-4

Purpose: To check that the character conversion tables in Annex A of V.18 have been correctly implemented.

Preamble: Establish a call between the tester and TUT in Baudot mode at 45 bit/s.

Method: The tester will send all possible characters preceded by the relevant case shift command one at a time and wait for a response from the TUT operator. Each character should be responded to at the TUT by typing the received character or <CR> if the character is not available.

Pass criteria:

1) The tester will verify that each character is correctly echoed back by the TUT. The operator should verify that each character is correctly displayed on the TUT.

2) The TUT will send the LTRS symbol before its first character and the appropriate mode character (either LTRS or FIGS) after every 72 subsequent characters.

Comments: The tester should indicate which character has been sent in each case. Some of the characters may not be available from the TUT keyboard and can be ignored. It is assumed that the character conversion is the same for Baudot at 50 bit/s and any other supported speed.

III.5.4.5.5 DTMF receiver disabling

Identifier: X-5

Purpose: To verify that the TUT disables its DTMF receiver for 300 ms when a character is transmitted.

Preamble: Establish a call between the tester and TUT in DTMF mode.

Method: The operator should send a single "e" character from the TUT which will result in sending a single DTMF tone to the tester. The tester will immediately start sending a unique character sequence using single DTMF tones. Examination of the TUT display will show when its receiver is re-enabled.

Pass criteria: The receiver should be re-enabled after 300 ms.

Comments:

III.5.4.5.6 DTMF character conversion

Identifier: X-6

Purpose: To check that the character conversion tables in Annex B of V.18 have been correctly implemented.

Preamble: Establish a call between the tester and TUT in DTMF mode.

Method: The tester will send each character from the set in Annex B, waiting for a response after each one. Each character should be responded to at the TUT by typing the same character.

Pass criteria: The tester will verify that each character is correctly echoed back by the TUT.

Comments: The conversion table is specified in Annex B of V.18. The receiver at the tester may be re-enabled 100 ms after transmission of each character to maximize likelihood of receiving character from the TUT. It is assumed that the echo delay in the test system is negligible.

III.5.4.5.7 EDT carrier timing and receiver disabling

Identifier: X-7

Purpose: To verify that the TUT sends unmodulated carrier for 300 ms before a character and disables its receiver for 300 ms after a character is transmitted.

Preamble: Establish a call between the tester and TUT in EDT mode.

Method: The operator should send a single character from the TUT. The tester will immediately start sending a unique character sequence. Examination of the TUT display will show when its receiver is re-enabled.

Pass criteria:

1) The TUT should send unmodulated carrier for 300 ms before the beginning of the start bit.

2) The receiver should be re-enabled after 300 ms.

3) The tester will confirm that 1 start bit and at least 1.5 stop bits are used.

Comments: The carrier should be maintained during the 300 ms after a character.

III.5.4.5.8 EDT bit rate and character structure

Identifier: X-8

Purpose: To verify that the TUT uses the correct bit rate and character structure in the EDT mode.

Preamble: Establish a call between the tester and TUT in EDT mode.

Method: The operator should transmit the string "abcdef" from the TUT.

Pass criteria:

1) The tester should measure the bit timings and confirm that the rate is 110 bit/s.
2) The tester should confirm that 1 start bit, 7 data bits, 1 even parity bit and 2 stop bits are used.

Comments:

III.5.4.5.9 V.23 calling mode character format

Identifier: X-9

Purpose: To verify that the TUT uses the correct character format in the V.23 calling mode.

Preamble: Establish a call from the TUT to the tester in V.23 mode.

Method: The operator should transmit the string "abcdef" from the TUT. The tester will echo characters back to the TUT as they are received. The tester will then transmit the string "abcdef" with ODD parity to the TUT.

Pass criteria:

1) Confirm that 1 start bit, 7 data bits, 1 even parity bit and 2 stop bits are transmitted.
2) The operator should confirm that there is no local echo at the TUT by checking that there are no duplicate characters on the TUT display.
3) The received string should be correctly displayed despite the incorrect parity.

Comments:

III.5.4.5.10 V.23 answer mode character format

Identifier: X-10

Purpose: To verify that the TUT uses the correct character format in the V.23 answer mode.

Preamble: Establish a call from the tester to the TUT in V.23 mode.

Method: The tester will transmit the string "abcdef" with ODD parity. The TUT should echo characters back to the tester as they are received. The operator should then transmit the string "abcdef" from the TUT.

Pass criteria:

1) The received string should be correctly displayed at the TUT despite the incorrect parity.
2) Confirm that 1 start bit, 7 data bits, 1 even parity bit and 2 stop bits are transmitted by the TUT.
3) The tester should confirm that there is remote echo from TUT.

4) The operator should confirm that there is local echo on the TUT.

Comments: This test is only applicable to Minitel *Dialogue* terminals. Prestel and Minitel *Normal* terminals cannot operate in this mode.

III.5.4.5.11 V.21 character structure

Identifier: X-11

Purpose: To verify that the TUT uses the character structure in the V.21 mode.

Preamble: Establish a call from the TUT to the tester in V.21 mode.

Method: The operator should transmit a string from the TUT that is long enough to cause the display to word wrap followed by "abcdef", new line (CR+LF). The tester will then transmit the string "123456", BACKSPACE (0/8) with ODD parity to the TUT.

Pass criteria:

1) The tester should confirm that 1 start bit, 7 data bits, 1 even parity bit and 1 stop bits are transmitted.
2) The word wrap should not result in CR+LF.
3) The forced new line should be indicated by CR+LF.
4) The last five characters on the TUT display should be "12345" (no "6") correctly displayed despite the incorrect parity.

Comments:

III.5.4.5.12 V.18 mode

Identifier: X-12

Purpose: To verify that the TUT uses the protocol defined in T.140.

Preamble: Establish a call from the TUT to the tester in V.18 mode.

Method: The operator should transmit a string from the TUT that is long enough to cause the display to word wrap followed by "abcdef", new line (CR+LF), new line (UNICODE preferred). The tester will then transmit the string "123456", BACKSPACE.

Pass criteria: The tester should confirm UTF8 encoded UNICODE characters are used with the controls specified in T.140.

Comments:

Source: **Ericsson**

Title: **H.248 Annex F with latest corrections**

Document for: **Information**

Agenda Item: **4**

This document is referenced from both CTM and the proposed 23.002 Annex B because it describes methods to handle interworking between text telephony in PSTN and text conversation in other networks.

The reference is informative because it is rather the procedures described than the H.248 environment that is important for the application

The basic document was approved in ITU-T in November 2000. This copy contains minor corrections in parameter values made in 2001.

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INTERNATIONAL TELECOMMUNICATION UNION

TELECOMMUNICATION STANDARDIZATION SECTOR

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STUDY GROUP 16

TITLE: H.248 ANNEX F: FACSIMILE, TEXT CONVERSATION AND CALL DISCRIMINATION PACKAGES

H.248 ANNEX F

Facsimile, Text Conversation and Call Discrimination packages

CONTENTS

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F.1 Summary

H.248 Annex F describes packages for fax, text telephone, call type discrimination, and data call detection for use with the H.248 Gateway Control Protocol. As defined in H.248, a "package" is an extension to H.248 that supports specific behaviour.

The packages are intended for control over gateway functions for transport of facsimile or text conversation between different network environments. Extensions can be made for other kinds of data transport.

- **The Call Type Discrimination package** defines control and monitoring of a PSTN line for the signalling protocols used in the beginning of a session of data transmission for fax, text telephony or data.
- **The Text Telephone package** defines control of a PSTN text telephone session in any of the modes supported by the automoding text telephone Recommendation V.18.
- **The Fax package** defines control of a PSTN fax transmission.
- **The Fax/Textphone/Modem Tones Detection package** defines control over a termination for detection of any signals from a fax, text telephone or data modem during a connection in voice mode.
- **The Text Conversation package** defines control over a real time interactive text conversation session using a universal presentation format and transferred with a transport method from a multimedia protocol in any network environment.
- **The IP Fax package** defines control over facsimile transmission in a packet network.

F.2 Scope

H.248 Annex F describes packages for the ITU-T H.248 gateway protocol related to data or telematic services. With terminations implementing these packages, a gateway is expected to handle initial modem negotiations, and the communication in voice, fax and text telephone call types. It contains:

Package "ftmd" for general detection of signals on a fixed telephone line indicating a possible request to enter some data related mode.

Package "ctyp" for general call discrimination to sort out if a call should be handled as voice, fax, text telephone or modem data, and perform the initial negotiation.

Package "txp" for communicating with text telephones in the telephone network.

Package "fax" for communication with facsimile in the telephone network.

Package "txc" for general text conversation in other environments.

Package "ipfax" for fax transmission in IP networks.

F.3 Definitions

F.3.1 Hexadecimal octet coding

Hexadecimal octet coding is a means for representing a string of octets as a string of hexadecimal digits, with two digits representing each octet.

Each octet is issued by the DTE or DCE in the same time sequence as transmitted on the GSTN line, with no intervening characters.

For each octet, the 8-bit sequence is encoded as two hexadecimal digits. Bit 0 is the first transmitted; bit 7 is the last.

Bits 7-4 are encoded as the first hexadecimal digit, with bit 7 as MSB and bit 4 as LSB. Bits 3-0 are encoded as the second hexadecimal digit, with bit 3 as MSB and bit 0 as LSB.

Examples:

Octet bit pattern (time order as specified in V.8 and V.8bis)	**Hexadecimal coding**	**T.50 codes**
00011011	D8	4/4, 3/8
11100100	27	3/2, 3/7
10000011 10100010 11001000 00001001	C1451390	4/3, 3/1, 3/4, 3/5, 3/1, 3/3, 3/9, 3/0

F.3.2 Hexadecimal octet sequence

A hexadecimal octet sequence is an even number of hexadecimal digits, terminated by a <CR> (T.50 0/13) character.

F.4 References

[1] IETF RFC 1889: Schulzrinne, H., *et al*, RTP: *A Transport Protocol for Real-Time Applications*, January 1996.

[2] ITU-T Recommendation T.30 (07/96) - *Procedures for document facsimile transmission in the general switched telephone network.*

[3] ITU-T Recommendation T.38 (06/98) - *Procedures for real-time Group 3 facsimile communication over IP networks*.

[4] ITU-T Recommendation V.8 (2000) - *Procedures for starting sessions of data transmission over the public switched telephone network*.

[5] ITU-T Recommendation V.8bis (2000) - *Procedures for the identification and selection of common modes of operation between data circuit-termination equipments (DCEs).*

[6] ITU-T Recommendation V.18 (2000) - *Operational and interworking requirements for DCES operating in the text telephone mode*.

[7] ITU-T Recommendation V.25 (1996) - *Automatic answering equipment and/or parallel automatic calling equipment on the general switched telephone network.*

[8] ITU-T Recommendation T.140 (1998) - *Text conversation protocol for multimedia application. With Amendment 1 (2000).*

[9] ITU-T Recommendation H.323 Annex G (2000) - *Text Conversation and Text SET (2000).*

[10] IETF G. Hellström, *"RTP Payload for Text Conversation", Internet Engineering Task Force, RFC 2793*. (2000)

[11] ITU-T Recommendation T.134 (1998) - *Text Chat Application Entity*.

[12] ITU-T Recommendation V.17 (02/91), Recommendation V.17 (02/91) - *A 2-wire modem for facsimile applications with rates up to 14 400 bit/s*.

[13] ITU-T Recommendation V.27ter (11/88) - *4800/2400 bits per second modem standardized for use in the general switched telephone network*.

[14] ITU-T Recommendation V.21 (11/88) - *300 bits per second duplex modem standardized for use in the general switched telephone network*.

[15] ITU-T Recommendation V.23 (11/88) - *600/1200-baud modem standardized for use in the general switched telephone network.*

[16] ITU-T Recommendation V.34 (02/91) - *A duplex modem operating at data signalling rates of up to 14 400 bit/s for use on the general switched telephone network and on leased point-to-point 2-wire telephone-type circuit*.

[17] ITU-T Recommendation V.90 (09/98) - *A digital modem and analogue modem pair for use on the Public Switched Telephone Network (PSTN) at data signalling rates of up to 56 000 bit/s downstream and up to 33 600 bit/s upstream*.

[18] ITU-T Recommendation V.61 (08/96) - *A simultaneous voice plus data modem, operating at a voice plus data signalling rate of 4 800 bit/s, with optional automatic switching to data-only signalling rates of up to 14 400 bit/s, for use on the General Switched ...*

[19] ITU-T Recommendation T.37 (06/98) - *Procedures for the transfer of facsimile data via store and forward on the Internet*.

[20] ISO/IEC 10646-1 (1993) - *Universal Multiple Octet Coded Character Set*.

[21] ITU-T Recommendation T.50 (1992) - *International Reference Alphabet (IRA) (formerly International Alphabet No. 5 or IA5) - Information technology - 7-bit coded character set for information interchange*.

[22] ITU-T H.323 Annex D (1998) - *Facsimile*.

F.4.1 Non-normative references

– RFC 2532, *Extended Facsimile Using Internet Mail.*, IETF

– RFC 2530, *Indicating Supported Media Features Using Extensions to DSN and MDN.*, IETF

– RFC 2531, *Content Feature Schema for Internet Fax.*, IETF

– RFC 2301, *File Format for Internet Fax*., IETF

– RFC 2302, *Tag Image File Format (TIFF) - image/tiff MIME Sub-type Registration*, IETF

– RFC 2303, *Minimal PSTN address format in Internet Mail*., IETF

– RFC 2304, *Minimal FAX address format in Internet Mail*., IETF

– RFC 2305, *A Simple Mode of Facsimile Using Internet Mail*., IETF

F.5 FAX/Textphone/Modem Tones Detection Package

PackageID: ftmd, 0x000E

Version: 1

Extends: tonedet version 1

This package defines an event to detect the presence of data traffic (fax, textphone or modem) on a line. The main intention of this event may be used to effect the compression option on the line so that an audio codec capable of transmitting modem signals can be invoked to handle the connection when needed. This Package extends the possible values of tone id in the "start tone detected" event. Note that there is no discrimination between tones from this package. If discrimination is desired, the Call Type Discrimination package should be invoked.

F.5.1 Properties

None.

F.5.2 Events

Events are defined as for the tone detection package.

F.5.2.1 Additional tone id value

dtfm, 0x0039

This tone id is generated when any of the following tones are detected.

"Tone"	Description	Applicable to
CNG	a T.30 fax calling	Fax
V21flag	a V21 tone and flags	Fax
CIV18	a V.8 CI with V.18 call function	Textphone
XCI	a V.18 XCI	Textphone
V18txp	a V.18 "txp"	Textphone
Belltone	a Bell 103 carrier, either the high or the low frequency channel (as defined in V.18)	Textphone
Baudot	a Baudot initial tone and character (as def. in V.18)	Textphone
Edt	an EDT initial tone and character (as def. in V.18)	Textphone
CIdata	a V.8 CI with any data call function	Data
CT	a V.25 calling tone	Text and Data
CIfax	a V.8 CI with facsimile call function	Fax
V21tone	a V.21 carrier, either the high or the low frequency channel	Text and Data
V23tone	a V.23 carrier, either the high or the low frequency channel	Text and data
V8bis	a V.8bis modem handshaking signal	Fax, Text and Data
ANS	V.25 ANS, equivalent to T.30 CED from answering terminal	Fax, Text and Data
ANSAM	V.8 ANSam	Fax, Text and Data

F.5.3 Signals

None.

F.5.4 Statistics

None.

F.5.5 Procedures

None.

F.6 Text Conversation package

Package Name: Text Conversation

PackageID: txc (0x00F)

Version: 1

Extends: None

Description:

The Text Conversation package is intended for enabling real time text conversation between terminals in different networks or multimedia environments. This package includes the mechanisms needed to transport T.140 text conversation streams [8] in multimedia environments. The transport mechanism will be different for each environment where the package is used.

F.6.1 Properties

F.6.1.1 Text buffering time

PropertyID: bufftime (0x0001)

Type: Integer

Possible values: 0-500

Defined in: LocalControl

Characteristics: Read/Write

Description:

This property indicates the time in ms that T.140 [8] data shall be collected before transmission in order to keep overhead from text low. In low bit rate IP networks, a value of 300 ms is recommended. In environments with low overhead or high bit rates this property should have the value 0 enabling immediate transmission of entered characters.

F.6.1.2 Text termination connection state

PropertyID: connstate (0x0002)

Type: Enumeration

Possible values:

Idle	(0x0001)	for no connection efforts
Prepare	(0x0002)	for being known in the termination and ready to accept connections. (the text capability is offered in session requests)
Initiate	(0x0003)	for taking the initiative to establish a text connection opening a text channel
Accept	(0x0004)	for accepting an incoming request for a text session
Deny	(0x0005)	for denying an incoming text connect request
Connected	(0x0006)	for established connection in text mode

Defined in: TerminationState

Characteristics: Read/Write

Description:

The connection state property is used to register text capability, request a text connection, and reflect details of the achieved text connection. For transport methods having separate channel control procedures, managed by the MGC, only a subset of the values is used: Idle, Prepare, Connected.

F.6.1.3 Text User Identity

PropertyID: txuserid (0x0003)

Type: String

Possible value: String of up to 64 characters in Unicode UTF-8 [20]

Defined in: LocalControl

Characteristics: Read/Write

Description:

This parameter holds the optional remote User Identity parameter of a T.140 [8] text conversation session, retrieved from the session.

F.6.1.4 Text Transport

PropertyID: trpt (0x0004)

Type: Enumeration

Possible values:

H224	(0x0001)	for H.224 Client ID=2 in H.320
AL1	(0x0002)	for AL1 in H.324
TCP	(0x0003)	for TCP as in H.323 Annex G [9]
RTP/T.140	(0x0004)	for RTP with T.140 [8] as in H.323 Annex G [9] or IETF SIP
RTP/RED/T.140	(0x0005)	for RTP with T.140 and redundancy coding RED as in H.323 Annex G or IETF SIP
T.134	(0x0006)	for T.134 in the T.120 environment [11]
Unassigned	(0x0007)	When no transport protocol is assigned

Defined in: LocalControl

Characteristics: Read/Write

Description:

The Transport parameter reflects the transport mechanism selected for the Text Conversation termination. When the media description has the full capability of describing sessions including the transport mechanism, this parameter is implied by the media descriptor.

F.6.1.5 Text Protocol Version

PropertyID: TextProto (0x0005)

Type: Integer

Possible values: Any integer corresponding to a T.140 version number. (currently 1)

Defined in: LocalControl

Characteristics: Read/Write

Description:

The version of the T.140 protocol used in the connection.

F.6.1.6 Redundancy Level

PropertyID: red (0x0006)

Type: Integer

Possible values: 0-6

0 = use default or automatic decision on redundancy level (default)

1 = use no redundancy

2-6 = use specified number of generations of data

Defined in: LocalControl

Characteristics: Read/Write

Description:

The number of generations to use in RTP redundancy coding including the Primary.

F.6.1.7 Txc request timer

PropertyID: txctim (0x0007)

Type: integer

Possible values: 0-6000

Default: 0

Defined in: LocalControl

Characteristics: Read/Write

Description:

The txctim property is a timer value in tenths of seconds for the requested operation. If the requested operation is not completed within this time, the state is returned to Idle and the result reported in the connchange event. An initial timer value of 0 indicates that no timer control is requested.

F.6.2 Events

F.6.2.1 Connection State Change

Event Id: connchange (0x0001)

EventDescriptorParameters

none

ObservedEventDescriptorParameters

ParameterName: Connection Change

ParameterID: connchng (0X0001)

Type: Enumeration

Possible Value: As property txc/connstate

Description:

This event will occur when the text connection state for the termination has changed. Its parameter is the new contents of the Connection State property.

If a request timed out, the state is returned to Idle.

F.6.3 Signals

None.

F.6.4 Statistics

F.6.4.1 Characters Transferred

StatisticsID: chartrans (0x0001)

Units: count

Description:

No. of bytes of T.140 data transferred through the termination.

F.6.4.2 Lost Packets

StatisticsID: packlost (0x0002)

Units: count

Description:

Number of T.140 packets lost as counted by the receiving T.140 termination.

F.6.5 Procedures

The following are standard transport mechanisms for text conversation in different environments.

– In H.320: H.224 with Client ID=2.

– In H.324: AL1 channel connected with H.245 procedures.

– In T.120: T.134 transport in T.125 communication channel environment.

– In H.323: RTP/T.140 or TCP as selected with H.245 messages.

– In IETF SIP: RTP/T.140 as initiated with SDP.

Note that the T.140 text media is also used together with V.18 [6] modems for text telephony, specified in a separate package: Text_Telephone (txp).

The Text Conversation package is intended to be added to a multimedia termination, handling appropriate multiplexing and control.

F.6.5.1 Function

A termination with Text Conversation adds capability declaration for a text conversation channel in the call setup according to procedures defined for each environment. When matching capabilities exist, a T.140 channel can be established according to the transport protocol used in the current environment. T.140 text stream contents received from one termination is transferred for transmission to other T.140 capable terminations in the context. The T.140 contents may be buffered for a short moment for possible collection of more text in the same transmission according to the buffer time property.

F.6.5.2 Informative description

Real time text conversation allows telecom users to carry out a written conversation. The presentation and coding aspects of standardized text conversation are defined in ITU-T T.140. Text is transmitted character by character (or in small blocks) so that the users experience a close interaction. The text and basic editing control is ISO 10646-1, UTF-8 [20] coded. The figure gives an example of how a text conversation can be displayed to the user.

ANNE	EVE
Hi, this is Anne. Yes, have you heard that I will come to Paris in November?	Oh, hello Anne, I am glad you are calling! It was long since we met! No, that was new to me. What brings you here?

FIGURE

Possible display of a one-to-one text conversation

For each transport environment, a suitable transport protocol must be selected to carry the text. Currently defined and recommended transport environments for T.140 text media streams that can be supported by this package are:

1) Packet networks, where the procedures described in H.323 Annex G [9] can be used for setting up and conducting text conversation sessions, using TCP or RTP/T.140 for the transport of T.140.

2) Packet networks, where the IETF Session Initiation Protocol SIP can be used for setting up and conducting text conversation sessions using RTP/T.140 for the transport of T.140.

3) The H.324 multimedia environment in PSTN, ISDN and Mobile networks, where an AL1 channel connected by H.245 procedures is used for T.140.

4) The H.320 multimedia environment, where a H.224 channel with client ID=2 is specified for transport of T.140.

5) The T.120 data conferencing environment, that can be used alone or in conjunction with any of the environments above, where T.134 specifies the application entity and T.125 the data channel for T.140.

A separate Text Telephone package (txp) supports text telephony in the PSTN using the ITU-T V.18 modem in native and legacy modes and T.140 for communication with terminations using this package.

Interworking between these forms of Text Conversation can be achieved through the use of gateways with packages defined here.

F.6.5.3 Total Conversation

Most text conversation transport environments are part of multimedia communication systems. With the introduction of text, they enable conversation in video, text and voice simultaneously, called Total Conversation. The total set of communication modes that people tend to use locally can be offered on a distance through Total Conversation. Since the text part is built on the unified presentation level T.140, the task to arrange interoperability of Total Conversation in different network environments through a gateway is simplified.

Video is optional in the multimedia systems. Therefore compatible text-and-voice conversation can also be established within the same framework.

F.6.5.4 Descriptor to use for text conversation

One descriptor value is of specific interest for the Text Conversation and Text Telephone packages. That is the text conversation media stream. It is described here for information.

Text conversation stream

This descriptor is used for the text conversation stream, according to ITU-T T.140 [8]. T.140 gives a general presentation level description for a termination supporting real time text conversation. The text and basic editing control is UTF-8 coded [20]. For each transport environment, a suitable transport protocol must be selected to carry the text.

T.140 is a registered MIME text stream name, that can be specified to be used as it is or in RTP embedding of RFC 2793 [10].

Examples:

From MGC to MG in an ADD command, the T.140 stream could be specified as this example shows:

```
Media { Stream = 4 { LocalControl {
        Mode = ReceiveOnly,
        g/NetworkType = RTP/IP4,
        g/PreferredCodecs=T140}}}
```

The MG would return the SDP specification for the media stream:

```
Media { Stream = 4 {Local = SDP {
        v=0
        c=IN IP4 125.125.125.111
        m=text 1111 RTP/AVP 98
        a=rtpmap:96 red
        a=fmtp: 98 96/96
        a=rtpmap: 96 t140}}}
```

F.7 Text Telephone package

PackageID: txp (0x0010~~6~~)

Version: 1

Extends: None

Description:

The text telephone package is used on a line termination in a Media Gateway, to handle text telephone calls. It includes V.18 [6] text telephone modem functionality that adapts to different legacy text telephone systems in the PSTN as well as it provides communication with V.18 equipped text telephones. The text media stream is UTF-8 coded [20] with a few editing functions as specified in ITU-T T.140 [8]. The text telephone package is intended to be operated together with the Call Type Discrimination package (ctyp) to perform V.18 automoding functions.

Text Telephony

Text Telephony offers a real time conversation in text between two parties. It may be combined with voice conversation. Text telephony in PSTN existed in at least six incompatible legacy modes before the automoding modem Recommendation for text telephony V.18 was introduced by ITU. V.18 is suitable for use in PSTN text telephones, but also in gateways for connection to the PSTN text telephones. When connected, it can operate in one of its native V.18 modes, or in any of the six legacy modes described in V.18 annexes. The legacy modes are Baudot, EDT, DTMF, V.21, Minitel and Bell103. The mode detection and adjustment of the transmission to the selected mode is automatic.

The native modes use ITU-T T.140 for the text coding and control and V.21 [14] or optionally V.61 [19] for the modulation. The legacy modes use different character coding schemes, but when used in a gateway, the text stream to and from the textphone termination is T.140 coded for all modes. The text telephone package described here includes character conversion, filtering and other adaptation needed for conversation with the legacy mode text telephones.

Carrier modes and carrier-less modes

Three of the legacy textphone modes are carrier-less. This means that they do not send any signal at all when there are no characters to transmit. Three legacy modes and the native V.18 modes use a carrier tone transmitted as long as the connection is maintained. If the carrier stops, it is detected but the line is not disconnected, because this is normal behaviour during call transfer and alternating voice and text usage.

Text telephone package considerations above the V.18 modem level

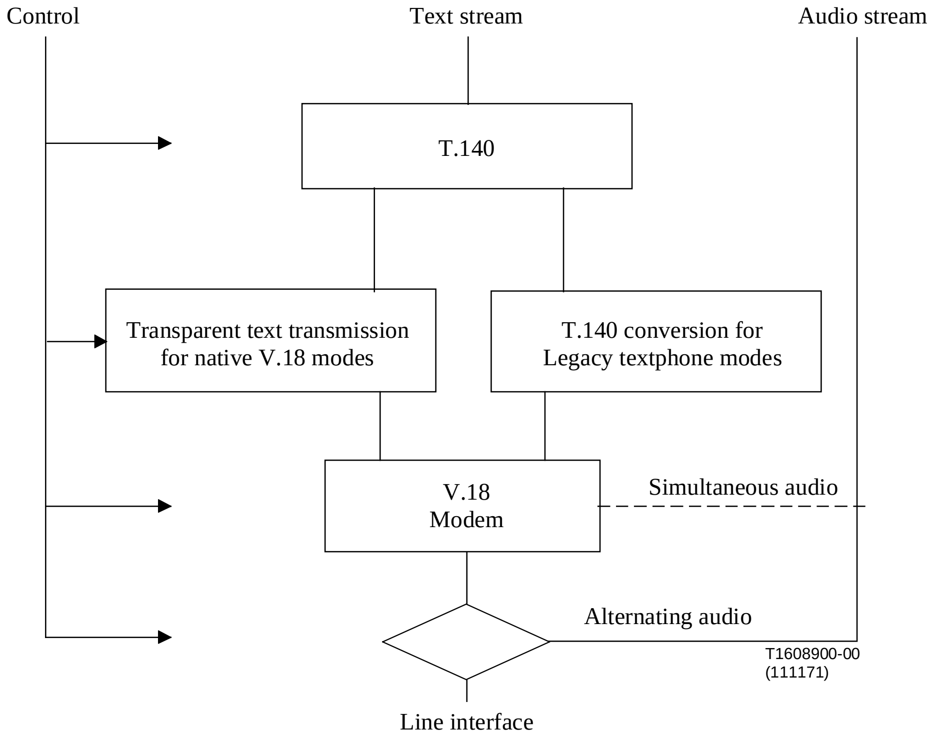
V.18 only specifies an automoding modem and the requirement to use T.140 when V.18 native mode is achieved in a connection. When used in a gateway, there are some general issues that must be handled above the V.18 level.

Character set

The legacy modes have limited character sets. For all legacy modes, appropriate character conversion, filtering and control interception is included in the package functionality, so that the communication with other T140 text terminations in the context is equalized to a T140 text stream.

Embedded termination functionality

There is no need to open all details of the use of V.18 and T.140 to be accessible from the MGC in a gateway. V.18, T.140, character conversion methods and other automated methods are therefore combined in the text telephone package that can be added to suitable terminations of a gateway. This figure describes the text telephone package components.



FIGURE

Text telephone package functional view

F.7.1 Properties

F.7.1.1 Conversation mode

PropertyID: convmode (0x0001)

Type: Sub-list

Possible values:

Text-only (0x0001) Basic text only mode, not possible to combine with voice

Alternating	(0x0002) Text and voice may be alternating
Simultaneous	(0x0003) Simultaneous text and voice mode

Defined in: Termination state

Characteristics: Read/Write

Description:

The behaviour of the termination is influenced by this property. By setting the property to a selection of the possible values, the number of ways that the conversation can be conducted can be defined. After connection the property contains the actual conversation mode used in the call.

The basic text only mode shall always be supported.

The alternating text and voice mode is most often used to enable one user to speak and read and the other to listen and type. It is used because there was no technology support for simultaneous voice and text when text telephony was introduced. It is only supported for compatibility with the legacy mode text telephone habits.

The simultaneous text and voice mode enables the users to communicate in any combination and order of the two media. No legacy mode terminals operate in this mode. V.18 equipped terminals with V.61 [18] modulation can operate in this mode.

F.7.1.2 Communication Mode

PropertyID: commode (0x0002)

Type: Enumeration

Possible values:

V18-V21Hi	(0x0001)	native V.18 mode transmitting on the high channel for text only or text and voice alternatively
V18-V21Lo	(0x0002)	native V.18 mode transmitting on the low channel for text only or text and voice alternatively
V18-V61C	(0x0003)	native V.18 mode for text and voice simultaneously, transmitting in the caller's channel
V18-V61A	(0x0004)	native V.18 mode for text and voice simultaneously, transmitting in the answering part's channel
V21Hi	(0x0005)	legacy V.21 mode transmitting on the high channel
V21Lo	(0x0006)	legacy V.21 mode transmitting on the low channel
DTMF	(0x0007)	DTMF text telephone mode
EDT	(0x0008)	EDT ("European Deaf Telephone")
Baudot 45	(0x0009)	Baudot 45.45 bit/s
Baudot 47	(0x000A)	Baudot undetermined bit rate
Baudot 50	(0x000B)	Baudot 50 bit/s

V23Hi	(0x000C)	V.23 modulation and Minitel coding transmitting on the high channel
V23Lo	(0x000D)	V.23 modulation and Minitel coding, transmitting on the low channel
BellHi	(0x000E)	Bell 103, transmitting on the high channel
BellLo	(0x000F)	Bell 103, transmitting on the low channel
None	(0x0010)	No mode achieved

Defined in: LocalControl

Characteristics: Read/Write

Description:

This property indicates what modulation and mode the V.18 modem is operating in, reflecting what type of text telephone it is in connection with. For an explanation of the different modes, see ITU-T V.18 [6].

If a specific mode of operation is wanted, this property is set before the text connection is made.

Normally it is set with the outcome of the V.18 automoding procedure performed with the Call Type Discrimination package.

When a legacy mode textphone signal is detected by the Call Type Discrimination package, the connection result is only reported, but V.18 does not transmit any signal until ordered to do so by setting this property or when probing is invoked.

F.7.1.3 Connection Mode

PropertyID: connmode (0x0003)

Type: Enumeration

Possible values:

Idle	(0x0001)	No connection established and no efforts to connect
Connecting	(0x0002)	For request of the native or legacy mode indicated in the Communication Mode property
Connected	(0x0003)	Connection established in one of the communication modes

Defined in: Termination State

Characteristics: Read/Write

Description:

This property indicates in what connection phase and mode the V.18 modem is operating. A connection effort is initiated by setting this property to connecting, with the desired mode in the Communication Mode property.

A V.18 modem can be controlled to operate in one of a set of modes for seeking contact with a counterpart. The modes available are listed as values of this property. Determination of the mode is made by the ctyp package, possibly combined with the probing action of that package.

Once connected, the termination operates in the selected mode until the text connection is lost or it is ordered to disconnect. If text connection is lost for a certain time, the automoding procedure can be restarted through the ctyp package, or the modem can stay in the achieved mode trying to reconnect.

The ctyp package may be used on a connected voice line to detect if the remote user wants to enter text mode. It must be noted that for some of the legacy modes (EDT, DTMF and Baudot), the user has to push some keys on the textphone to make the connection when V.18 is set in the automode monitor mode. This is slightly unusual for a textphone user, who normally waits for the answering side to start the conversation. Therefore, the explicit automoding modes should be used when possible, probing as answering and sending V.18 signals as calling.

If a connection request fails, the property returns to Idle state. If the connection request succeeds, the property changes value to Connected.

F.7.1.4 Action at loss of connection

PropertyID: lossconnection (0x0006)

Type: Enumeration

Possible values:

Keep	(0x0001)	keep selected communication mode
Return	(0x0002)	return to automoding

Defined in: Termination State

Characteristics: Read/Write

Description:

This property tells how the V.18 modem handles loss of text connection. When "Keep" is selected, the conversation is optimized for the alternating text - voice mode. When "Return" is selected, the communication is optimized for call forwarding between different types of text telephones. For that case, ctyp must be invoked for reconnection.

F.7.1.5 V18 options

PropertyID: v18opt (0x0007)

Type: Enumeration

Possible values: List of:

V.61 capability	(0x0001)	indicates the ability to use V.61 modulation [19]

Defined in: Termination state

Characteristics: Read/Write

Description:

This property indicates what optional capabilities the V.18 modem implementation has and is allowed to use.

F.7.1.6 Character set

PropertyID: characterset (0x0008)

Type: String

Possible values: ISO registered name for a character set

Defined in: Termination State

Characteristics: Read/Write

Description:

The legacy modes have limited character sets. For all legacy modes, appropriate character conversion, filtering and control interception is included in the package functionality, so that the communication with other T.140 text terminations in the context is equalized to a T.140 text stream. For a user-friendly conversion of received national characters in the limited character sets to ISO 10646-1 used in T.140, there is a need to specify what national translation table to use. This is valid for EDT, DTMF, V.21 and Baudot modes. The Character set parameter is the registered ISO code for the national variant of the ITU-T T.50 [21] character set used. Default is:

– German for EDT,
– Danish for DTMF (suitable also for the Netherlands),
– Swedish/Finnish for V.21 (suitable also for UK),
– International Reference Version for Baudot.

Example: In Norway, the letter "Æ" (A and E together) is used in the same location of the 7-bit character table as used for letter "Ä" (A with umlaut) in Finland and Sweden. The international reference version has the character "[" (left bracket) in the same position. In T.140 these characters have unique positions.

F.7.2 Events

F.7.2.1 Connection mode changed

EventID: connchng (0x0001)

EventDescriptorParameters

none

ObservedEventDescriptorParameters

Same as the property txp/commode

Description:

This event reports the change of communication mode, as a result of a connection effort, or a disconnection.

F.7.3 Signals

None.

F.7.4 Statistics

F.7.4.1 Number of characters transferred

StatisticsID: chartrans (0x0001)

Units: count

Description:

Number of bytes of T.140 data transferred (sent and received).

F.7.4.2 Number of alternating turns

StatisticsID: altturns (0x0002)

Units: count

Description:

Number of alternating turns when using alternating conversation mode.

F.7.5 Procedures

F.7.5.1 Basic operation

After line connection, the termination where the Text Telephone package is implemented should be requested to try a text telephone connection using the functionality of the Call Type Discrimination Package for the modem signalling according to ITU-T V.18 in a selected mode. Once the connection is established, the text telephone package is used for the text communication in the established mode.

After connection in text mode, the result is a gateway context with one textphone termination and one voice line termination connected to the same line. In the same context, the normal case is to have other terminations with audio and text conversation media.

In the simplest text-only case, the audio streams are not used and may be released.

Text received through the V.18 modem is converted if necessary to T.140 [8]. It is embedded in the RTP/T.140 format according to the rules in T.140 and RFC 2793 [10], specifying RTP/T.140. Text received from other text conversation terminations is transmitted through the text telephone termination after extraction from the RTP packets. This process continues until any end disconnects.

F.7.5.2 Informative description

Descriptors to use for text telephony:

Two descriptor values are of specific interest for the Text Telephone package. That is the text conversation media stream and the V.18 modem. The text conversation media stream is described in the Text Conversation package. The V.18 modem descriptor is described here for information.

F.7.5.3 V.18 Modem

Modem name V18.

This modem type is intended for communication with text telephones in the PSTN. Its operational modes are implemented in the textphone package. The logic for setting and detecting the mode according to V.18 is handled by the ctyp package. Some properties of the text telephone package and the Call Type Discrimination package directly reflect parameters for control of the V.18 modem. V.18 modem implementations may have different capabilities reflected in the property values.

A V.18 modem may be operated in automode monitor mode, when it listens on a voice line for text telephone signals. This mode can be used to detect that the user wish to transit from voice to text during a voice call. That is done entirely with the ctyp package.

Alternatively, a V.18 modem may be operated in modes where it actively tries to establish a text telephone connection. The procedure includes transmission of text telephone specific signals on the line. For calling modems, it is done by the CI signal in the ctyp package. For an answering modem it is done with the ctyp package combined with probing from the textphone package by setting the commode property to the probing mode.

When the mode is discriminated, the commode property should be set to request communication in that mode.

After successful connection in a text telephone mode, the text session is conducted in the specific mode as controlled with the commode property, and the text stream is made available in T.140 format for other text terminations in the context.

The text telephone package only contains the text connection and text media aspects of the termination. It is supposed to be combined with appropriate call control packages, line interface packages and voice channel packages.

F.7.5.4 Operation with alternating text and voice mode

If the involved gateways have the alternating text and voice capability, the following procedure can be applied to give the users a possibility to go back and forth between using text and voice. Between the terminals in the context, two streams are members of the context during the call, the text stream and the audio stream. The procedure is slightly dependent on the terminal type as described in the following section.

F.7.5.5 Alternating text and voice mode with legacy, carrier-less textphones

For the carrier-less types Baudot, DTMF and EDT the following way to operate should be used: When V.18 detects text, the textphone termination stops feeding the audio stream into the audio stream of the context, and instead inserts the detected and T.140 converted characters into the text stream. This mode is continued as long as characters keep coming from the PSTN textphone.

When no more characters arrive, and no textphone signal is received within 1 second, the audio channel is again fed to the Audio-stream channel. If new text comes from the V.18 side, the process is repeated.

It is important that the implementation of V.18 can retrieve characters from the first detected text telephone signals after each mode shift. The leading tones before the characters can be as short as 150 ms.

If text is received from the context through the Text stream, when V.18 is not active receiving text, the voice path is muted, and the characters are sent to the V.18 modem for transmission. When all text is transmitted and no more is received for two seconds, the audio channels are enabled again.

Since the carrier-less systems are one way alternate transmission systems, transmission of characters is possible only in one direction at a time. Once started, reception is given priority.

In the Context, two way simultaneous transmission is possible. Therefore, characters received from the context while V.18 is busy receiving should be buffered (up to a reasonable limit).

All these actions after the initial connections are automatic and are handled within the textphone termination.

F.7.5.6 Alternating voice and text conversation in carrier mode

After a carrier mode text connection is established, loss of carrier can be taken as the indication that the audio stream shall be connected with audio interface of the line. When the remote end is a V.21, Bell or V.18 device, the text communication can be full duplex, so the gateway can just let the text streams flow between the terminations.

When carrier reappears, or text is received through the text system, the audio stream shall be muted, and text transmission noted.

Minitel does not support any voice interworking mode.

F.7.5.7 Simultaneous voice and text mode

In case the simultaneous voice and text method is enabled, the handling of the voice and text channels is trivial. Once connected, the text stream can stay connected with the remote text stream all the time to serve a two-way simultaneous text conversation, and the audio channel can be connected with the remote audio stream to support a two-way simultaneous audio channel. This mode can be supported by V.18 with V.61 modulation.

F.8 Call Type Discrimination package

PackageID: ctyp (0x0011)

Version: 1

Extends: none

Description:

This package monitors the termination for signals indicating presence of a T.30 telefax terminal [2], a V.18 or legacy mode text telephone [6] or data modem. In cooperation with the MGC and the remote MG or endpoint, it can perform exchange of signals until the call type is determined and an appropriate mode for the call can be established.

The package contain modem negotiation functions of ITU-T V.25 [7], V.8 [4], V.8bis [5], V.18 [6] and T.30 [2].

F.8.1 Properties

F.8.1.1 Call Types

PropertyID: calltyp (0x0001)

Type: sub-list

Possible values:

FAX (0x0001)

TEXT (0x0002)

DATA (0x0003)

Defined in: Termination State

Characteristics: Read/Write

Description:

The Call Types property selects the types of calls for which the termination is monitored. Note that the connection is by default regarded to be capable of handling audio and therefore no specific value is included for that.

F.8.1.2 Text Call Types

PropertyID: ttyp (0x0002)

Type: Sub-list

Possible values:

V21	(0x0001)
DTMF	(0x0002)
Baudot45	(0x0003)
Baudot50	(0x0004)
Bell	(0x0005)
EDT	(0x0006)
Minitel	(0x0007)
V18	(0x0008)

Description:

This parameter indicates for what text telephone modes the termination is monitored, used in TEXT mode.

F.8.1.3 V8bissupport

PropertyID:	v8bsup (0x0003)
Type:	Boolean
Possible values:	True V.8bis is supported by the package
	False V.8bis is not supported by the package
Defined in:	Termination State
Characteristics:	Read

Description:

Support of the V.8bis [5] modem negotiating procedure is optional.
The V8bissupport property indicates if V.8bis is supported. It can be used in TEXT, FAX and DATA modes.

F.8.1.4 Probe message

PropertyID:	probemsg (0x0004)
Type:	String
Possible Value:	Any string, not more than 20 characters long
Defined in:	Termination State
Characteristics:	Read/Write

Description:

This property holds a short string that the termination transmits as a stimulating probe message for the carrierless communication modes in the answering modes. The far end user will see this message when it is transmitted in the mode matching the counterpart’s textphone, and type a response back, enabling the V.18 modem to detect the type of carrierless text telephone in the connection.

When issued, it is automatically followed by "GA" in Baudot probing, and with "+" in EDT and DTMF probing to reflect the turntaking signal habit in the different user communities. The string could be customized to briefly inform the called user about what service that is reached.

Note that the string is not issued in the carrier modes.

F.8.1.5 Probe order

PropertyID: probeorder (0x0005)

Type: Sub-List

Possible values: (for recommended orders, see V.18)

Any combination of none to six of the type indicators

V21	(0x0001)
DTMF	(0x0002)
Baudot	(0x0003)
EDT	(0x0004)
MINITEL	(0x0005)
BELL	(0x0006)

in any desired order

Defined in: Termination state

Characteristics: Read/Write

Description:

This property holds an indication on what modes to probe for, and the order the probes will be transmitted. Probing is a time-consuming procedure and it is important that the most likely modes are probed first. The order to select depends on if any legacy mode textphones are on the market in the area where the gateway is installed. An optimized order can be composed by enumerating the desired specific type indicators. Note that leaving out a type from probing may cause connection problems for connection with textphones of that type.

F.8.1.6 PhasereversalDetect

PropertyID: v8bsup (0x0006)

Type: Boolean

Possible values:
True Phase reversal detection is supported by the termination
False Phase reversal detection is not supported by the termination

Defined in: Termination State

Characteristics: Read

Description:

This property indicates support of detection of the phase reversals within ANS or ANSam signals. If this property has the value "False", ANS with phase reversals (ANSBAR) will be reported as ANS and ANSam with phase reversals (ANSAMBAR) will be reported as ANSam in the dtone event.

F.8.2 Events

F.8.2.1 Discriminating tone detected

EventID: dtone (0x0001)

Description:

This event indicates that a signal valid for detection and discrimination of mode was detected. The signal name is given as a parameter. Further logic is needed in some cases to discriminate the call type from this information. The V.8bis related parameters are returned only when V.8bis is supported [5].

Note that some textphones operate with DTMF tones. This package decodes initial DTMF signals according to the specification for text telephones in V.18 [6]. DTMF detection may be indicated also from the "dd" package if that is active.

EventsDescriptor parameters:

none

ObservedEventDescriptor parameters:

DiscriminatingToneType

ParameterID: dtt (0x0001)

Type: Enumeration

Possible values:

For FAX

CNG	(0x0001)	a T.30 fax calling tone
V21flag	(0x0002)	V21 tone and flags for fax answering

For TEXT

XCI	(0x0003)	a V.18 XCI
V18txp1	(0x0004)	a V.18 txp signal in channel V.21(1)
V18txp2	(0x0005)	a V.18 txp signal in channel V.21(2)
BellHi	(0x0006)	a Bell 103 carrier on the high channel
BellLo	(0x0007)	a Bell 103 low channel
Baudot45	(0x0008)	a Baudot45 initial carrier and characters
Baudot50	(0x0009)	a Baudot50 initial carrier and characters
Edt	(0x000A)	an EDT initial tone and characters
DTMF	(0x000B)	DTMF signals

For DATA

Sig	(0x000~~B~~C)	Modulation signal from a mode only used for data, i.e. not V.21, V.23 nor Bell 103

Common to TEXT and DATA:

CT	(0x000~~C~~D)	a V.25 calling tone
V21hi	(0x000~~D~~E)	a V.21 carrier on the higher frequency channel
V21lo	(0x000~~E~~F)	a V.21 carrier on the low frequency channel
V23hi	(0x0010~~0F~~)	a V.23 high carrier
V23lo	(0x0011~~0~~)	a V.23 low carrier
CI	(0x0012~~1~~)	a V.8 CI with contents in "dtvalue"

Common to FAX, TEXT and DATA:

ANS	(0x0013~~2~~)	V.25 ANS, equivalent to T.30 CED from answering terminal
ANSbar	(0x0014~~3~~)	V.25 ANS with phase reversals
ANSAM	(0x0015~~4~~)	V.8 ANSam
ANSAMbar	(0x0016~~5~~)	V.8 ANSam with phase reversals
CM	(0x0017~~6~~)	V.8 CM with contents in "dtvalue"
CJ	(0x0018~~7~~)	V.8 CJ
JM	(0x0019~~8~~)	V.8 JM with contents in "dtvalue"
ENDOFSIG	(0x001A~~9~~)	End of reported signal detected reported for continuous or repeated signals
V8BIS	(0x001B~~20~~)	V.8bis signal, with signal type in parameter V8bistype and value in "dtvalue"

DiscriminatingToneValue

ParameterID: dtvalue (0x0002)

Type: string

Possible values:

When used for V.8 and V.8bis related messages, the following coding rules applies:

If a V.8bis message is detected without a preceding V.8bis signal, the preamble is reported as a 0 <signal> value.

The contents of valid V.8bis message(s), if detected, are reported using hexadecimal octet coded string(s) (F.3.1.1). Flag detection and consumption, flag transparency 0-bit deletion and FCS checking are performed by the MG. The MG shall not report invalid messages (e.g. bad FCS). If two consecutive messages are detected but the first is invalid, the MG shall indicate this with a comma in front of the second message (e.g. <2nd message>). Two concatenated V.8bis messages are reported with two consecutive <message> indications.

V8bistype

ParameterID: v8bist (0x0004)

Type: enumeration

Possible values:

ESi	(0x0001)	V.8bis signal ESi
ESr	(0x0002)	V.8bis signal ESr
MRe	(0x0003)	V.8bis signal MRe
MRdi	(0x0004)	V.8bis signal MRd from initiator
MRdr	(0x0005)	V.8bis signal MRd from responder
CRe	(0x0006)	V.8bis signal CRe
CRdi	(0x0007)	V.8bis signal CRd from initiator
CRdr	(0x0008)	V.8bis signal CRd from responder
MS	(0x0009)	V.8bis message MS with contents in "dtvalue"

CL (0x000A) V.8bis message CL with contents in "dtvalue"
CLR (0x000B) V.8bis message CLR with contents in "dtvalue"
ACK (0x000C) V.8bis message ACK with contents in "dtvalue"
NAK (0x000D~~E~~) V.8bis message NAK with contents in "dtvalue"

Description: A detected V.8bis [5] signal. V.8bis can be used for all modes.

F.8.3 Signals

F.8.3.1 V8Signal

SignalID: v8sig (0x0001)

SignalType: OO

Parameters:

V.8 Signal Type

Parameter ID: v8styp (0x0001)

Type: Enumeration

Possible values:

CM (0x0001)
CJ (0x0002)
JM (0x0003)
CI (0x0004)
v8nosig (0x0005) no signal - used to stop the V.8 signal

Default may be provisioned

V8SigCont

Parameter ID: v8scont (0x0002)

Type: string

Possible values:

Allowed contents of the signals, coded as hexadecimal
octet coded string

Default is empty

Description: The V.8 [4] signals carry data for call type and modulation modes. These parameters can be supplied through the v8cont parameter. V.8 can be used for FAX, TEXT and DATA modes.

V18XCIEnable

Parameter ID: v18xcien (x0003)

Type: Boolean

Possible values:

True XCI transmission enabled during V.18 CI transmission

False XCI transmission disabled

Default is True

Description: XCI can be sent intermixed with CI transmission as specified in V.18 to stimulate plainMinitel terminals to respond as text telephones. Used in TEXT mode.

F.8.3.2 AnswerSignal

SignalID: ans (0x0002)

Signal Type: OO

Parameters:

AnsType

ParameterID: AnsType (0x0001)

Type: Enumeration

Possible values:

ANS	(0x0001)	V.25 ANS (equivalent to T.30 CED) for all modes
ANSBAR	(0x0002)	V.25 ANS with phase reversals for all modes
ANSAM	(0x0003)	V.8 ANSam for all modes
ANSAMBAR	(0x0004)	V.8 ANSam with phase reversals for all modes
V18txp1	(0x0005)	a V.18 txp signal played in V.21 channel (1) for TEXT
V18txp2	(0x0006)	a V.18 txp signal played in V.21 channel (2) for TEXT
ansnosig	(0x0007)	no signal - used to turn off the signal

Default may be provisioned

F.8.3.3 CallingSignal

SignalID: callsig (0x0003)

SignalType: OO

Parameters:

callSigname

Parameter ID: cSn (0x0001)

Type: Enumeration

Possible values:

CT	(0x0001)	V.25 Calling Tone used for TEXT and DATA
CNG	(0x0002)	T.30 Calling tone used for FAX with defined cadence
callnosig	(0x0003)	no signal - used to turn off the signal

Default may be provisioned

F.8.3.4 V8bisSignal

SignalID: v8bs (0x0004)

Signaltype: BR

Parameters:

V8bisSigname

ParameterID: V8bsn (0x0001)

Type: Enumeration

Possible values:

ESi	(0x0001)	V.8bis signal ESi
ESr	(0x0002)	V.8bis signal ESr
MRe	(0x0003)	V.8bis signal MRe
MRdi	(0x0004)	V.8bis signal MRd from initiator
~~MRdrh~~	~~(0x0005)~~	~~V.8bis signal MRd from responder on high power~~
MRdrl	(0x00~~6~~5)	V.8bis signal MRd from responder on low power
~~CReh~~	~~(0x0007)~~	~~V.8bis signal CRe on high power~~
CRel	(0x0006)	V.8bis signal CRe on low power
CRdi	(0x0007)	V.8bis signal CRd from initiator
CRdr	(0x0008)	V.8bis signal CRd from responder
MS	(0x0009)	V.8bis message MS with contents in signalvalue
CL	(0x000A)	V.8bis message CL with contents in signalvalue
CLR	(0x000B)	V.8bis message CLR with contents in signalvalue
ACK	(0x000C)	V.8bis message ACK with contents in signalvalue
NAK	(0x000D)	V.8bis message NAK with contents in signalvalue
MRdrh	(0x000E)	V.8bis signal MRd from responder on high power
CReh	(0x000F)	V.8bis signal CRe on high power

Default may be provisioned

Description: V.8bis [5] signals can be used in all modes. Some V.8bis signals contain data messages, supplied in V8bisSigContents.

V8bisSigContents

ParameterID: V8bscont (0x0002)

Type: string

Possible values: Valid contents for the V.8bis signals

Default is empty

Description: Some of the V.8bis signals are messages. Their contents can be defined with theV8biscont parameter.

V.8bis can be used in TEXT, FAX and DATA modes.

The transmitted V.8bis message frame(s) is specified as hexadecimal octet coded string (F.3.1). Additional messages are delimited by comma characters. Flag generation, flag transparency 0-bit insertion and FCS generation are performed by the MG. If no data is provided by the MGC, no V.21 carrier is generated beyond that used in segment 2. For two concatenated messages, the MG shall insert the required preamble between the first and second messages.

F.8.3.5 V18probe

SignalID: v18prob (0x0005)

SignalType: OO

Parameters: none

Description:

This signal transmits the v18 probes in order to stimulate possible text telephones to transmit connect establishing signals. The probes are sent according to the specification in Recommendation V.18. For carrierless probes, the string in the "probemsg" property is transmitted. The probes are sent in the order specified in the property "probeorder".

F.8.4 Statistics

None.

F.8.5 Procedures

The Call Type Discrimination package is invoked for cases when the network connection is established and the call may enter one of the types of voice, fax, text telephone and modem. The package contains functionality to support the decision and connection processes. Once discriminated and the modem handshaking completed, an appropriate specific call type package should be invoked to complete the connection establishment on the modulation level and perform the session.

When used for active modem negotiation, by means of commands from the MGC, the termination shall be made to operate according to the Recommendations for modem negotiation; V.25 [7], V.8 [4], V.8bis [5], V.18 [6] and T.30 [2]. For probing according to V.18 during the negotiating process, the probing mechanism may be applied as defined in this package by turning the signal v18prob ON.

The package may also be used for monitoring and reporting on data activity on the termination.

F.8.5.1 Informative description

If the desired call type is known from the beginning, the call type discrimination package should be invoked in order to actively try to establish a connection by sending out stimulating signals. By contrast this package is also used to monitor the line to detect signals which are to be relayed to the Media Gateway Controller as input to a discrimination decision. In principle, when tones are reported to the MGC as events by an MG, the MG should avoid passing these tones via the media stream where possible, to reduce the possibility of unwanted duplicate tones. Since the Call Type Discrimination package can be invoked to initially only monitor the line, it can be invoked on lines where voice calls are the most common mode of operation. There may be situations where this passive way of working results in less efficient or less reliable connection in fax/text/data mode.

F.8.5.2 Operation

The package is activated on a termination of a line in an outgoing or incoming call where fax, text or data mode may be wanted. The properties are set to the enabled call types.

F.8.5.3 Operation for incoming calls

The call is answered, the destination is evaluated and the remote call initiated using packages and gateway functions outside the scope of this package.

The MGC may order stimulating signals defined in this package to be sent on the line.

The line is monitored for signals for the selected modes as defined in the "dtone" event descriptor.

The MGC is expected to evaluate call type indications of all types; registered type of the destination, offered capabilities of the endpoint, invoked connection efforts of specific types from the endpoint and discriminating events from a call type discriminating package active in setting up the connection with the other endpoint.

As soon as the modem handshaking is complete and a condition is reached that is valid for only one call type, a package for handling that call type should be invoked by the MGC, thus placing the MG into the desired mode of operation.

The package contains components for conducting a negotiation procedure according to the different connection procedures defined in Recommendations V.25 [7], V.8 [4], V.8bis [5], T.30 [2], T.38 [3] and V.18 [6]. (V.8bis support is optional and its availability can be interrogated through the property V8bissupport.)

F.8.5.4 Operation for transit calls, coming from and going to the switched network

If no fax/text/data indication is present in the incoming call, the outgoing call is placed in voice mode, with the Call Type Discrimination package active.

If a valid tone is detected, it is reported to the MGC as an event. By actions of the MGC it can be signalled to be replayed at the other end.

The process continues according to the rules of the connection procedures until the call type can be determined and the mode of operation can be established.

F.8.5.5 Operation for calls having one endpoint in the packet network

If no fax/text/data indication is present in the incoming call, the outgoing call in the packet network is placed in voice mode.

If a request to open a text channel, a fax channel or a data channel is made from the packet endpoint, the corresponding call type is tried on the switched network connection.

If a signal indicating presence of a fax, textphone or a modem is received from the circuit switched network, and the call type can be evaluated, a corresponding channel is requested to the remote packet endpoint. If that request is acknowledged, the connection in the fax/text/data mode is completed on the switched side.

If the call type cannot be evaluated, further signal exchange is performed on the switched interface until the call type is determined, and then the channel establishment continues on the packet side.

F.8.5.6 Cases when the call type cannot be determined from the signals

For cases when the call type cannot be determined by the signal exchange, a decision must be taken by other means, or a transparent transport can be selected.

The other means to make the decision may be a number analysis and comparison to registered user preferences or network defaults.

Cases when the decision is not possible by signal analysis but need to be taken by external means:

V.21: Used both for text telephony and for credit card transactions. The decision is recommended to be based on regional preferences and registering preference for data per destination number in regions where the default preference is for text telephony.

V.23: Used both for Minitel-based text telephones and for the Minitel information retrieval system. The conflict is only when an answering endpoint transmits the v23hi signal. A transparent data transport is recommended for this case.

F.8.5.7 Scenarios and call flows

Signal sequence scenarios can be derived from the different connection protocols, with T.38 being the main protocol for fax, V.18 for text telephony and V.8/V.25 for data.

The typical fax scenario is discriminated when CNG is detected from the calling end and a corresponding CED (ANS) and/or V.21flags are detected at the answering end. For instances when either a CNG or ANS is not reported to the MGC, V21flags detection is sufficient for fax discrimination. Alternatively, a V.8 CM or JM signal with a fax call type may be detected at either end.

The text telephone scenario is discriminated when a text telephone call type is detected in V.8, a text telephone function is negotiated in V.8bis, or a signal valid for text only is detected.

The data scenario is discriminated when a data call type is detected in V.8, a data function is negotiated in V.8bis, or a data mode (not text) is entered by any part.

In all cases the handshaking protocol should be completed using the Call Type Discrimination package, before entering the selected data mode.

F.8.5.8 Initial characters

For carrierless text telephones of the Baudot, EDT and DTMF types the text transmission itself is needed for mode determination, and therefore the characters received during determination shall be stored. They shall be made available by local actions in the MGto be used by the txp package as initially received text for a seamless takeover of a connection.

F.8.5.9 Time critical handling

The default way of handling connection requests should be to propagate the connection request to the remote endpoint, and verify capabilities before positively responding to an incoming connection request for fax, text or data mode. It can however be very time-consuming to verify the endpoint capabilities, and connect appropriate channels. The caller may timeout between detection of off-hook, and receiving a positive signal. Similar time critical steps exist in the V.8, V.8bis, V.18, T.30 and V.25 procedures. The MGC must take action to compromise between the risk of one party timing out because of long waiting for a signal, and the risk of connecting a fax/text/data call before the capabilities of the endpoints are verified and the appropriate channels connected. One possible way to handle this risk is to define default actions to take before any party in the call times out. The ctyp package gives the MGC all necessary control to handle the connection process including such actions.

F.9 Fax package

Package Name: Fax

PackageID: fax (0x0012)

Version: 1

Extends: None

Description:

The fax package is intended for enabling fax communication between terminals/applications in different networks or messaging environments. This package includes the mechanisms needed to identify T.30 [2] fax sessions (signals and data).

F.9.1 Properties

F.9.1.1 Fax connection state

PropertyID: faxstate (0x0001)

Type: Enumeration

Possible values:

Idle	(0x0001)	no connection efforts
Prepare	(0x0002)	known in the termination and ready to accept connections
Negotiating	(0x0003)	taking the initiative to establish a fax connection
TrainR	(0x0004)	Fax Phase B or later training as Receiver
TrainT	(0x0005)	Fax Phase B or later training as Transmitting
Connected	(0x0006)	completed connection
EOP	(0x0007)	Procedures Complete
ProcInterrupt	(0x0008)	Procedure Interrupt Processing
Disconnect	(0x0009)	Premature Disconnect

Characteristics: Read/Write

Defined in: Termination State

Description:

After successful phase A connection with the ctyp package, the connection state property is used to request a fax connection. When placing a termination into a fax mode, the initial state shall be set to "Negotiating".

When this property is interrogated, it shall reflect the state of the achieved fax connection.
A connection effort can be cancelled by setting the faxstate property to Idle.

F.9.1.2 Fax Transport

PropertyID: ftrpt (0x0001)

Type: Enumeration

Possible values:

T30	(0x0001)	for T.30 PSTN sessions without ECM
T30ECM	(0x0002)	for T.30 PSTN sessions with ECM (non-V.34)
T.30V34	(0x0003)	for T.30 PSTN sessions with V.34 (half-duplex)

Characteristics: Read/Write

Defined in: Termination State

Description:

The Transport parameter reflects the transport mechanism selected for the fax termination.

F.9.1.3 TransmissionSpeed

PropertyID: trspd (0x0002)

Type: Integer

Possible values: 1200-33600

Defined in: Termination State

Characteristics: Read/Write

Description:

The Transport parameter reflects the transmission speed seen at the analogue interface for the fax relay or the transmission speed used by the FAX termination (T.30 PSTN).

F.9.1.4 PSTN Interface

PropertyID: pstnif (0x0003)

Type: Enumeration

Possible values:

NA	(0x0001)	not applicable
V17	(0x0002)	
V27TER	(0x0003)	
V29	(0x0004)	
V21	(0x0005)	
V34	(0x0006)	

Defined in: Termination State

Characteristics: Read/Write

Description:

The PSTN Interface parameter reflects the interface used to connect to a physical FAX machine.

F.9.2 Events

F.9.2.1 Fax Connection State Change

Event ID: faxconnchange (0x0001)

EventDescriptor Parameters:

none

ObservedEventDescriptorParameters

Fax Connection Change

ParameterID: faxconnchng (0x0001)

Type: Enumeration

Possible Value:

Idle	(0x0001)	no connection efforts
Prepare	(0x0002)	known in the termination and ready to accept connections
Negotiating	(0x0003)	taking the initiative to establish a fax connection
TrainR	(0x0004)	Fax Phase B or later training as Receiver
TrainT	(0x0005)	Fax Phase B or later training as Transmitting
Connected	(0x0006)	completed connection
EOP	(0x0007)	Procedures Complete
ProcInterrupt	(0x0008)	Procedure Interrupt Processing
EOF	(0x0009)	end of fax session, call terminating
PI	(0x000A)	Priority Interrupt ; Switch to Voice
Disconnect	(0x000B)	Premature Disconnect

Description:

This event will occur when the fax connection state for the termination has changed. Its parameter is the new Fax Connection State. A connection effort that timed out returns the termination to the Idle state.

F.9.3 Signals

None.

F.9.4 Statistics

F.9.4.1 Pages Transferred

StatisticsID: pagestrans (0x0001)

Type: integer

Description:

No. of pages of fax image data transferred through the termination.

F.9.4.2 Train Downs

StatisticsID: traindowns (0x0002)

Units: count

Description:

Number of times FAX trained down during transmission.

F.9.5 Procedures

The following are standard transport mechanisms for fax in different environments.

- In T.30: Use T.30 [2] procedures with and without ECM.
- In T.30 Annex C/F: Use T.30 procedures selected via V.8 (Used for V.34 fax).

F.9.5.1 Function

A termination with Fax provides a method for transfer of fax pages preceded by negotiations in the call setup according to procedures defined for each environment. When matching capabilities exist, the appropriate sessions can be established in order to transfer pages of image or binary data.

Real time fax allows telecom users to transfer fax pages in real time. The procedural aspects of GSTN fax are defined in ITU-T T.30 [2]. The compression methods used in transporting fax images are defined in T.4, T.6, T.81, T.82, T.85 and T.44. In traditional T.30 without error correction, images are transferred in a stream one page at a time. In T.30 with error correction, images are transferred in blocks that are also known as partial pages. Numerous examples of fax sessions are contained in Appendix IV to T.30.

1) For each transport environment, a suitable transport protocol must be selected to carry the image. Currently defined and recommended transport environments for T.30 media streams that can be supported by this package are GSTN networks, where the procedures are defined in T.30, T.30 Annex A (for error correction), T.30 Annex C (duplex protocol) and Annex F (half duplex V.34 protocol).

F.9.5.2 Process of Adding Fax Capable Terminations

The MGs are responsible for detecting fax tones and relaying the related events to the MGC. The MGC should conduct Call Discrimination as defined within the Call Type Discrimination Package in order to determine whether a fax or other mode is applicable. The MGC may choose to skip this if the MG is not capable of the Call Type Discrimination Package. Once the MGC evaluates the tones and determines that the incoming call is fax, the MGC shall execute appropriate Modify commands to place the termination into a "Negotiating" state.

F.9.5.3 Process of Ending a Fax Call

The MGs are responsible for detecting events that would cause the interruption of a fax call. The MGC is responsible for making the determination if this switch can be made and instruct the MGs to switch. It is also responsible for switching it back to fax.

The MGC should receive indication that the fax call ends from the MG before receiving typical call termination indications.

F.10 IP Fax package

Package Name: IPFax

PackageID: ipfax (0x0013)

Version: 1

Extends: None

Description:

The fax package is intended for enabling real time or store and forward fax communication between terminals/applications in different networks or messaging environments. This package includes the mechanisms needed to transport T.30 fax sessions (signals and data) in a real time IP environment. The transport mechanism will be different for each environment where the package is used.

F.10.1 Properties

F.10.1.1 Fax connection state

PropertyID: faxstate (0x0001)

Type: Enumeration

Possible values:

Idle	(0x0001)	no connection efforts
Prepare	(0x0002)	known in the termination and ready to accept connections
Negotiating	(0x0003)	taking the initiative to establish a fax connection
TrainR	(0x0004)	Fax Phase B or later training as Receiver
TrainT	(0x0005)	Fax Phase B or later training as Transmitter
Connected	(0x0006)	for completed connection
EOP	(0x0007)	Procedures Complete
ProcInterrupt	(0x0008)	Procedure Interrupt Processing
Disconnect	(0x0009)	Premature Disconnect

Characteristics: Read/Write

Defined in: Termination State

Description:

After successful phase A connection with the ctyp package, the connection state property is used to request a fax connection. When placing a termination into a fax mode, the initial state shall be set to "Negotiating".

When this property is interrogated, it shall reflect the state of the achieved fax connection.

F.10.1.2 IPFaxTransport

PropertyID: ipftrpt (0x0001)

Type: Enumeration

Possible values:

T38UDPTL	(0x0001)	for T.38 [3] using UDPTL
T38TCP	(0x0002)	for T.38 using TCP
T37	(0x0003)	for T.37 [19]
AUDIO	(0x0004)	for audio codec (e.g. G.711 over RTP [1])

Characteristics: Read/Write

Defined in: Termination State

Description:

The IP Fax Transport parameter reflects the transport mechanism selected for the fax termination.

F.10.1.3 TransmissionSpeed

PropertyID: trspd (0x0002)

Type: Integer

Possible values: 0-33600

Characteristics: Read/Write

Defined in: Termination State

Description:

The Transport property reflects the transmission speed seen at the IP interface for the fax relay. A value of zero (0) indicates that there is no speed set.

F.10.1.4 T.38 Capabilities

PropertyID: T38Capabilities (0x0003)

Type: sub-list

Possible values:

FaxFillBitRemoval	(0x0001)	indication of fill bit removal
FaxTranscodingMMR	(0x0002)	for MMR transcoding availability
FaxTranscodingJBIG	(0x0003)	for JBIG transcoding availability
UDPFEC	(0x0004)	UDP Forward Error Correction
UDPRedundancy	(0x0005)	UDP Redundancy Error Correction

Characteristics: Read/Write

Defined in: Termination State

Description:

These capabilities describe the T.38 [3] fax termination. They are defined in Recommendation T.38 Annex B. Their SDP equivalents are defined in Recommendation T.38 Annex D.

F.10.1.5 T38MaximumBufferSize

PropertyID: T38MaxBufferSize (0x0004)

Type: Integer

Possible values: 0-

Characteristics: Read/Write

Defined in: Termination State

Description:

This capability describes the T.38 fax termination. They are defined in Recommendation T.38 Annex B. Their SDP equivalents are defined in Recommendation T.38 Annex D.

F.10.1.6 T38MaximumDatagramSize

PropertyID: T38MaxDatagramSize (0x0005)

Type: Integer

Possible values: 0-

Characteristics: Read/Write

Defined in: Termination State

Description:

This capability describes the T.38 fax termination. They are defined in Recommendation T.38 Annex B. Their SDP equivalents are defined in Recommendation T.38 Annex D.

F.10.1.7 T38Version

PropertyID: T38Version (0x0006)

Type: Integer

Possible values: 0-

Characteristics: Read/Write

Defined in: Termination State

Description:

This is the T.38 version number.

F.10.2 Events

F.10.2.1 Fax Connection State Change

Event ID: faxconnchange (0x0001)

EventDescriptor Parameters:

none

ObservedEventDescriptorParameters

Fax Connection Change

ParameterID: faxconnchng (0x0001)

Type: Enumeration

Possible Value:	Idle	(0x0001)	no connection efforts
	Prepare	(0x0002)	known in the termination and ready to accept connections
	Negotiating	(0x0003)	taking the initiative to establish a fax connection
	TrainR	(0x0004)	Fax Phase B or later training as Receiver

TrainT	(0x0005)	Fax Phase B or later training as Transmitter
Connected	(0x0006)	for completed connection
EOP	(0x0007)	Procedures Complete
ProcInterrupt	(0x0008)	Procedure Interrupt Processing
EOF	(0x0009)	- end of fax session, call terminating
PI	(0x000A)	- Priority Interrupt ; Switch to Voice
Disconnect	(0x000B)	Premature Disconnect

Description:

This event will occur when the fax connection state for the termination has changed. Its parameter reflects the new state. If a connection effort times out, it is reported in this event, with the faxconnchng parameter set to Idle.

F.10.3 Signals

None.

F.10.4 Statistics

F.10.4.1 Pages Transferred

StatisticsID: pagestrans (0x0001)

Type: integer

Description:

No. of pages of fax image data transferred through the termination.

F.10.4.2 Train Downs

StatisticsID: traindowns (0x0002)

Units: count

Description:

Number of times FAX trained down during transmission.

F.10.5 Procedures

The following are standard transport mechanisms for fax in different environments.

- In T.38 Annex B: UDPTL or TCP in T.38 fax only communication channel environment.
- In H.323 Annex D [22]: UDPTL or TCP as selected with H.245 messages.
- In T.38 Annex D (SIP): UDPTL or TCP as initiated with SDP.
- In T.38 Annex E: UDPTL or TCP as initiated with H.248.
- In T.37: SMTP (MIME)/TCP.

F.10.5.1 Function

A termination with Fax provides a method for transfer of fax pages preceded by negotiations in the call setup according to procedures defined for each environment. When matching capabilities exist, the appropriate sessions can be established in order to transfer pages of image or binary data.

Real time fax allows telecom users to transfer fax pages in real time. For each transport environment, a suitable transport protocol must be selected to carry the image. Currently defined and recommended transport environments for T.30 media streams that can be supported by this package are:

1) Packet networks, where the procedures described in T.38 Annex B [3] can be used for setting up and conducting fax sessions, using TCP or UDPTL for the transport of T.30 signals and data.

2) Packet networks, where the procedures described in H.323 Annex D [22] can be used for setting up and conducting fax and voice sessions, using TCP or UDPTL as negotiated via H.245.

3) Packet networks, where the IETF Session Initiation Protocol SIP can be used for setting up and conducting fax sessions as defined in T.38 Annex D using UDPTL or TCP for the transport of T.30 signals and data.

4) Packet networks, where H.248 can be used for setting up and conducting fax sessions as defined in T.38 Annex E using UDPTL or TCP for the transport of T.30 signals and data.

5) Packet networks, where the packets of G.711 coded data (with T.30 signals and data embedded) can be transported via RTP.

The Extended Simple Mail Transport Protocol messaging environment over packet, that can be used alone or in conjunction with any of the environments above, where T.37 [19] specify the methods for transporting image/tiff files using the same compression methods as specified for use in T.30. For information it can be noted that RFC 2301-2305 and RFCs 2530-2532 specify these transport mechanisms.

Interworking between these forms of fax can be achieved through the use of gateways with packages defined here.

F.10.5.2 Process of Adding IP Fax Capable Terminations

The MGs are responsible for detecting fax tones and relaying the related events to the MGC. The MGC should conduct Call Discrimination as defined within the Call Type Discrimination Package in order to determine whether a fax or other mode is applicable. The MGC may choose to skip this if the MG is not capable of the Call Type Discrimination Package. Once the MGC evaluates the tones and determines that the incoming call is fax, the MGC shall execute appropriate Modify commands to place the IP fax capable termination into a "Negotiating" state.

F.10.5.3 Process of Ending a Fax Call

The MGs are responsible for detecting events that would cause the interruption of a fax call. The MGC is responsible for making the determination if this switch can be made and instruct the MGs to switch. It is also responsible for switching it back to fax.

The MGC should receive indication that the fax call ends from the MG before receiving typical call termination indications.

F.10.5.4 Informative Example

One possible instruction from an MGC to an MG to modify an existing context to a T.38 media stream:

```
MGC to MG:
MEGACO/1.0 [123.123.123.4]:55555
Transaction = 14 {
 Context = 2000 {
  Modify = RTP/1 {
   Media {
    Stream = 1 {
     Local {
v=0
c=IN IP4 124.124.124.222
m=image 1111 udptl t38
a=T38FaxRateManagement:transferredTCF
a=T38UdpEC:t38UDPFEC
     }
    }
   }
  }
 }
}
```

Addendum 1 to Recommendation T.140

1. Introduction

The amendments described in this document are essential for implementations of T.140, especially when used over unreliable channels.

2. Contact Information

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3. Document History

Version	Date	Description
1	Feb 18, 2000	Addition of "Missing text marker"

4. References

- ITU-T Recommendation T.140 (1998), Text Conversation Protocol for Multimedia Application.

5. Additions to Recommendation T.140. Text Conversation Protocol for Multimedia Application.

[Begin addition to T.140]

5.1 Addition of a marker for missing text to Recommendation T.140

In Table 2, add:

Missing text marker	Marks missing text		

In Section 7, Code elements, add in the table:
Replacement Character $FFFD_{16}$ Marker for missing text.

In Section 8, detailed coding and procedures, add:
8.9   Missing text marker
Purpose: To mark in the text the position of lost character(s).
Coding: 0xFFFD
Graphical representation: A white question mark in a black diamond. (see ISO 10646-1)
Procedure: The Missing Text Marker "Replacement Character" should be inserted in the T.140 data stream by any protocol entity who has discovered a data loss affecting some T.140 data.

[End Addition to T.140]

INTERNATIONAL TELECOMMUNICATION UNION

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STUDY GROUP 16 – CONTRIBUTION 48 CORRECTED

SOURCE*: RAPPORTEUR

TITLE: ITU-T RECOMMENDATION T.140 - TEXT CONVERSATION PROTOCOL FOR MULTIMEDIA APPLICATION

Summary

This Recommendation specifies a text conversation protocol. The intention of this protocol is to be a common presentation level suitable for straightforward real time text conversation in multimedia services and in text telephony.

It is based on ISO 10646 Universal Character Set 16 bit characters and features character by character transmission and a limited set of presentation controls.

Its application to the data conferencing environment is specified in Recommendation T.134.

Its application for plain text telephony in the PSTN is specified in Recommendation V.18.

Its application in video telephony is specified in Recommendation H.324 and H.245.

It is meant to be easily applied wherever there is a data channel available to carry the protocol.

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(54830)

1 Background

This is a brief description of the application environment where the protocol is intended to be used.

1) Text entered at one terminal is distributed to other terminals participating in the same session.
2) Text entered by one terminal is displayed in a window on that terminal and on others included in the session.
3) The character set is UTF-8 coded ISO 10646-1 level 3, used in order to be useful on all markets with no or little extra configuration problems. The intention to use a specific subset of ISO 10646 can be signalled in the protocol.
4) The normal case is to transmit and display character by character as they are entered. Short time buffering (0.5 sec) may be introduced by the transport mechanism in some configurations to reduce overhead. If used, the buffering is part of the transport mechanism used and is beyond the scope of this Recommendation.
5) The text entries from the different participants should be displayed in such a way that they can be easily read and the order of the entries perceived.
6) In case of multipoint use, the origin of the text should be displayed in connection to the text. The supporting multipoint protocol is assumed to supply the identification.
7) If characters from languages with writing directions right to left are supported, the implicit writing direction should also be supported according to rules in ISO 10646-1.
8) Editing and control functions from ISO 6429 and ISO 10646-1 are included for:
   - New line.
   - Erase last character.
   - Alert the user during a session.
   - Select graphic rendition.
9) A mechanism is defined for extending the protocol without disturbing communication with terminals implementing only limited versions.
10) Session control and transmission functions are needed from the implementation environment for:
   - Initiating and identifying a session.
   - Alerting on incoming calls (activate external signals; visual, audible or tactile).
   - Accepting a session.
   - Ending a session.
   - Transport the protocol data.
11) The protocol is suitable for application in:
   - Point-to-point situations.
   - Multipoint conferencing when the session control functions provide the multipoint distribution services.
12) Support for use of the protocol is foreseen:
   - in pure text conversation situations or in combination with.
   - voice.

- video.
- data conferencing.
- any combination of these modes.

2 Scope

This Recommendation specifies a simple text conversation protocol. Its purpose is to offer a standardized way to perform conversations in text mode between terminals.

The described protocol is intended to be used in the following environments:

- As the text conversation protocol used between two devices using Recommendation V.18 modems for text telephony.
- As the text conversation protocol between nodes in a point-to-point or multipoint multimedia conference as specified in Recommendation T.134.
- As the point-to-point text conversation protocol between multimedia terminals using a logical data channel transport mechanism in cases when T.120 functionality is not available.

3 References

The following Recommendations and other references contain provisions which, through reference in this text, constitute provisions of this Recommendation. At the time of publication, the editions indicated were valid. All Recommendations and other references are subject to revision; all users of this Recommendation are therefore encouraged to investigate the possibility of applying the most recent editions of the Recommendations and other references listed below. A list of the currently valid ITU-T Recommendations is regularly published.

ISO/IEC 10646-1: (1993), Universal Multiple Octet Coded Character Set

ISO/IEC 6429 (1992), Control functions for coded character sets

ITU-T Recommendation V.18 (1998), Operational and interworking requirements for DCEs operating in the text telephone mode

ITU-T Recommendation T.134 (1998), Text Chat Application Protocol Entity

4 Definitions

4.1 Session

For the purpose of this Recommendation, a session is a logical connection between two or more user terminals for the purpose of exchanging information in text format on a real-time basis.

4.2 Node

A terminal or group of terminals. A node can also incorporate a MCU for the purpose of coordinating multipoint sessions.

4.3 Data channels

A communication path used to carry text and presentation control information.

4.4 Origin identification

The user terminal may have an origin identification that can be used to identify the display of text from that terminal. The origin is specified by the surrounding transport protocol.

5 Abbreviations

UCS Universal Multiple-Octet Coded Character Set

UTF UCS Transformation Format

MCU Multipoint Control Unit

6 Protocol

This Recommendation only addresses the session contents. The procedures used to establish a session are beyond the scope of this Recommendation.

6.1 Required Session control functions

The session control functions are implemented with functions external to the channel and may be different for each transport mechanism. Although not part of this Recommendation the following parameters and functions are required and should be specified for each implementation environment.

Protocol identity	Registered value for the initial version of the T.140 protocol.
Destination node	An address valid in the environment where the protocol is used.
Originating node	An address for the node where the text origins.
User identity	Name and other valid identifications for an end user.
Data	Data for transmission from the text conversation protocol.

During session establishment, the protocol identity connected to Recommendation T.140 should be signalled.

The following conceptual session functions are needed to support a T.140 session.

TABLE 1

Functions in the session layer

Function title	Purpose	Parameters
Prepare session	Announce readiness to accept invitations to sessions (may be a local function)	Supported protocol T.140
Initiate session	Ask for a session with a specified node	Destination node, originating node and user identity, protocol T.140
Accept session	Accept to enter a session	Accepted protocol, accepting user identity
Deny session	Refuse to enter a session	Refusing user identity
Disconnect session	Leave a session	
Data	Transmit data to one or all members of a session	Data from the presentation protocol, destination node or all

6.1.1 Requirements on the data transmission functions

A data channel is set up with mechanisms specific to each environment where the protocol is implemented. Text is transmitted to the data channel from the text conversation protocol character by character.

The requirements and the specific mechanisms for buffering are beyond the scope of this Recommendation. However if buffering is provided in the data channel, it should not delay transmission more than 0.5 seconds and not be related to complete lines of input. Any grouping of data into blocks for transmission should be transparent to the text protocol.

Data from one node shall be delivered in the same order as it was transmitted.

6.2 Presentation Protocol functions

The protocol functions of the protocol are invoked by transmission of protocol data elements in the channel established by the session control functions.

The following parameters are used in specific protocol functions.

Parameter	Purpose
Text content	Text from one source in a session.
Display characteristics	As defined in SGR of ISO 6429.
UCS subset	An ISO registered subset of ISO 10646.
Function	Protocol function.

The following table gives an overview over the presentation protocol functions.

TABLE 2

Protocol functions

Function	Purpose	Parameters	
Alert user in session	Intended to cause an alerting signal from the user terminal during a session		
Erase last character	Used to erase the last character		
Identify UCS subset	Indicate intended subset within ISO 10646	UCS subset	
Interrupt	Initiate mode change		
New line	Move current display position to the next line		
Text	Text to display in conversation	A character of text	
Application protocol function	Extended control function	Function and parameters (to be defined)	
Select graphic rendition	Suggests display attributes for the following text	Colours, fonts and other display characteristics	

7 Code elements

Characters shall be sent, ordered in octets. If implemented in a user terminal, the terminal should also provide for local display of transmitted characters. The character set shall conform to the two octet version, of ISO 10646-1.

All terminals implementing this recommendation shall support the characters in the "IRV" and "Latin-1 supplement" in ISO 10646. Support for other parts of ISO 10646 is optional.

Presentation control functions are coded according to the principles of ISO 6429.

ISO 6429 control functions shall be padded with 00 characters as specified in ISO 10646-1.

In transmission, all code elements shall be transformed to the UTF-8 form of ISO 10 646-1.

The following control sequences are included in this protocol.

Name	**Code**	**Usage**
BEL	0007	Bell, provides for alerting during an active session.
BS	0008	Back Space, erases the last entered character.
NEW LINE	2028	Line separator.
CR LF	000D 000A	A supported, but not preferred way of requesting a new line.
INT	ESC 0061	Interrupt (used to initiate mode negotiation procedure).

SGR	009B Ps 006D	Select graphic rendition. Ps is rendition parameters specified in ISO 6429.
SOS	0098	Start of string, used as a general protocol element introducer.
ST	009C	String terminator, end of SOS string.
ESC	001B	Escape - used in control strings.
Byte order mark	FEFF	Zero width, no break space, used for synchronization.

7.1 Code signature and synchronization

The ZERO WIDTH NO-BREAK SPACE character (FEFF) shall be inserted in the beginning of the session.

The reception of ZERO WIDTH NO-BREAK SPACE shall be used as a tool to verify the right byte-order within the characters. (See ISO 10 646-1).

8 Detailed coding and procedures

8.1 Text

Purpose: Transfer text to display as text in the receiving window of the peer terminal(s).

Code: Character according to ISO 10646-1, level 3.

Procedure: Text contents is the default protocol element in a session. Thus, if characters are received, that are not recognized as part of other protocol elements, they shall be regarded as text from the peer partner in a session and be decoded and displayed as text.

When supported by the terminal, the received character shall be displayed in the receiving window, using the rules for displaying ISO 10646 characters and the rules for implicit and explicit writing direction.

A receiving user terminal shall display some representation of a character even if the character is not supported by the terminal.

8.2 Erase last character

Purpose: Erase the last character sent from the display at the receiving end.

Code: BS: 0008.

Procedure: On the receiving end: Move the insertion point to the last character and erase it.

Combined characters are erased as a unit, with one BS erasing the whole character even if it is combined from more than one component.

Control sequences (like CR LF) are erased in one operation.

NOTE - The same action shall be taken on the local display.

8.3 New line

Purpose: Move the insertion point for text to the beginning of the next line in the display window.

Preferred Code: LINE SEPARATOR: 2028.

Accepted code: CR LF: 000D 000A.

8.4 Alert user in session

Purpose: Intended to cause an alerting indication at the receiving terminal during a session.

NOTE 1 - Users may have a need to indicate the receipt of an alerting signal through such mechanisms as flashing the screen or through the triggering external devices such as lights or vibrators.

NOTE 2 - This function should not be intermixed with the need for external alerting caused by incoming calls or incoming connect requests.
Code: BEL: 0007.

Procedure: Sent during a session to activate alerting signals.

8.5 Interrupt

Purpose: To initiate a request for a mode change.

Coding: INT: ESC 0061.

Procedure: After receiving INT, the terminal stops the data transmission, disconnects the data session and prepares for connection in a new mode as requested by the user. Can be used to revert to voice telephone mode when appropriate.

8.6 Identify UCS subset

Purpose: Announce an intention to use a standardized subset of ISO 10646.

Coding: Defined by ISO for each UCS subset.

Procedure: A sending terminal entering a specific language area can send the "Identify UCS Subset" indication to the receiving terminal. The intention of this function is to prepare the terminal for display of characters from a specific subset of ISO 10646.

8.7 Application protocol function

Purpose: Identified coding of extensions to the protocol, so that they can be introduced unilaterally without disturbing the display.

Coding: SOS, function code, parameter string, ST.

Where: The function code is one ISO 10646 character uniquely identifying the function.

The parameter string shall not be more than 255 ISO 10646 characters in length and shall not include the ST character.

Procedure: The receiving terminal shall function according to the request. The whole function shall be ignored by a terminal not supporting it. For terminals supporting the extended function they will have an effect specified for that function.

If no trailing ST is received after the maximum length of the parameter string, the protocol reverts to normal element decoding mode.

Currently defined application protocol functions are:

- Unsupported request, with function code "?" = 003F.
- Indicate ENHANCED profile, with function code "_" = 005F.

8.8 Select graphic rendition

Purpose: Propose display attributes for the following text.

Coding: 009B, Ps, 006D, where Ps indicates the suggested display attribute(s) according to ISO 6429.

Procedure: The transmitting terminal sends the "select graphics rendition" on user command. The normal action is to set the graphics rendition of subsequently received text according to the rules of the cumulative graphics rendition combination mode of ISO 6429. The receiving terminal may obey the display attribute or not, depending on both its capabilities and the preferences of the user. It should present some distinguishable representation of the display attribute.

NOTE - The default parameter value 0 requests a return to default rendition.

APPENDIX I/T.140

Display arrangements

Display arrangements

This is an informative appendix. It is not a formal part of the Recommendation.

Display of text is suggested to be horizontal.

The display of text from the members of the conversation should be arranged so that the text from each participant is clearly readable, and its source and the relative timing of entered text is visualized in the display. Mechanisms for looking back in the contents from the current session should be provided. The text should be displayed as soon as it is received.

Examples of display arrangements

Two examples of possible display arrangements are given here.

One window per source

One possible way of arranging the display is to have one window per source, including one window for the user of the terminal itself. The identity of the source can be displayed in the window header. The windows can be placed side by side. At end of line, word wrapping should be used.

The window contents can be chronologically ordered so that its position relative to lines in the other windows indicates when it was received relative to the other.

ANNE	EVE
Hi, this is Anne.	Oh, hello Anne, I am glad you are calling! It was long since we met!
Yes, have you heard that I will come to Paris in November?	No, that was new to me. What brings you here?

FIGURE I-1

A possible way to display a conversation with one window each

One window for the whole session

Another way to arrange the display is to have one common window. The text from each participant can be displayed with its source identity as a label. With two participants, the character flow can be character by character, thus maintaining two logical insertion points. With more participants it can be acceptable to buffer text until a NEW LINE is received, and then display it in the window with a label indicating the source.

```
<EVE>This is Eve.
<ANN>Hi this is Anne.
<EVE>Oh hello Anne, I am glad you are calling!
<ANN>Have you heard that I am coming to Paris in November?
<EVE>No, that was new to me. What brings you here?
```

FIGURE I-2

A possible way to display a conversation in a common window