Final draft ETSI EG 202 320 V1.2.1 (2005-08)

ETSI Guide

Human Factors (HF);
Duplex Universal Speech and Text (DUST) communications

Reference

DEG/HF-00057

Keywords

digital, disability, speech, text, video

ETSI

650 Route des Lucioles
F-06921 Sophia Antipolis Cedex - FRANCE

Tel.: +33 4 92 94 42 00 Fax: +33 4 93 65 47 16

Siret N° 348 623 562 00017 - NAF 742 C
Association à but non lucratif enregistrée à la
Sous-Préfecture de Grasse (06) N° 7803/88

Contents

Intellectual Property Rights

IPRs essential or potentially essential to the present document may have been declared to ETSI. The information pertaining to these essential IPRs, if any, is publicly available for **ETSI members and non-members**, and can be found in ETSI SR 000 314: *"Intellectual Property Rights (IPRs); Essential, or potentially Essential, IPRs notified to ETSI in respect of ETSI standards"*, which is available from the ETSI Secretariat. Latest updates are available on the ETSI Web server (http://webapp.etsi.org/IPR/home.asp).

Pursuant to the ETSI IPR Policy, no investigation, including IPR searches, has been carried out by ETSI. No guarantee can be given as to the existence of other IPRs not referenced in ETSI SR 000 314 (or the updates on the ETSI Web server) which are, or may be, or may become, essential to the present document.

Foreword

This ETSI Guide (EG) has been produced by ETSI Technical Committee Human Factors (HF), and is now submitted for the ETSI standards Membership Approval Procedure.

Executive summary

Text communication services such as chat, instant messaging, SMS and e-mail are now mainstream digital services, but have the disadvantage that they do not provide a fluent conversation. Real time conversational text provides a fluency of conversation, which avoids the disjointed effects of speaking out of turn that often occurs in messaging systems. Real time conversational text has up till now been largely restricted to expensive specialized terminals used mainly by the deaf community.

The present document describes a design for all solution providing real time conversational services that support text, speech and optionally video on current and future networks.

The present document identifies user requirements for a conversational text service in clause 6.1, describes the requirements for the solution in clause 6.2, and clauses 7 to 12 give details of how existing textphone systems can migrate to the new service in clause 13.

Annex A reports the study of the many different textphone systems currently in use.

Proposals for further work to deal with currently unsolved problems are contained in clauses 15 and 16.

1 Scope

The present document sets out detailed and practical requirements for Duplex Universal Speech and Text (DUST) protocols and services that provide enhanced real time text and speech conversation for all users.

The present document identifies existing textphones, chat systems and protocols, identifies user requirements for effective text and telephone conversation, and gives methods of fulfilling these requirements.

The present document identifies necessary extensions to signalling protocols and transport facilities of current multimedia systems to achieve the DUST requirements. In addition, the present document sets out a migration process from existing national textphone systems.

The present document points out where further standardization work is required to resolve any remaining problems.

The present document contains information applicable to network operators, service providers, terminal and network manufacturers.

2 References

The following documents contain provisions which, through reference in this text, constitute provisions of the present document.

- References are either specific (identified by date of publication and/or edition number or version number) or non-specific.
- For a specific reference, subsequent revisions do not apply.
- For a non-specific reference, the latest version applies.

Referenced documents which are not found to be publicly available in the expected location might be found at http://docbox.etsi.org/Reference.

[1] BT SIN 359: "BT TextDirect™ Service Description".

[2] CEPT T/CS 34-15: "Aspects techniques du terminal audiographique (Text telephone)".

[3] ETSI EG 202 116: "Human Factors (HF); Guidelines for ICT products and services; Design for All".

[4] ETSI EG 203 072: "Universal Communications Identifier (UCI); Results of a detailed study into the technical areas for identification harmonization; Recommendations on the UCI for NGN".

[5] ETSI ETR 250: "Transmission and Multiplexing (TM); Speech communication quality from mouth to ear for 3,1 kHz handset telephony across networks".

[6] ETSI ETR 297: "Human Factors (HF); Human factors in videotelephony".

[7] ETSI ETR 333: "Human Factors (HF); Text telephony; Basic user requirements and recommendations".

[8] ETSI SR 002 180: "Requirements for communication of citizens with authorities/organizations in case of distress (emergency call handling)".

[9] ETSI SR 002 299: "Emergency Communications; Collection of European Regulatory Principles".

[10] ETSI TR 101 806: "Human Factors (HF); Guidelines for Telecommunication Relay Services for Text Telephones".

[11] ETSI TS 101 331: "Telecommunications security; Lawful Interception (LI); Requirements of Law Enforcement Agencies".

[12] ETSI TS 101 882-3: " Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON) Release 4; Protocol Framework Definition; Part 3: TIPHON Simple Call service meta-protocol definition".

[13] ETSI TS 122 226: "Universal Mobile Telecommunications System (UMTS); Global text telephony; Stage 1: Service description".

[14] ETSI TS 123 226: "Universal Mobile Telecommunications System (UMTS); Global text telephony (GTT); Stage 2".

[15] ETSI TS 126 226: "Universal Mobile Telecommunications System (UMTS); Global text telephony; Transport of text in the voice channel (3GPP TS 26.226 version 5.0.0 Release 5)".

[16] ETSI TS 126 230: "Universal Mobile Telecommunications System (UMTS); Global text telephony; Cellular text telephone modem transmitter C-code description".

[17] ETSI TS 126 231: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); Global text telephony (GTT); Cellular text telephone modem minimum performance requirements".

[18] ETSI TS 126 235: "Universal Mobile Telecommunications System (UMTS); Packet switched conversational multimedia applications; Default codecs".

[19] IETF RFC 2198: "RTP Payload for Redundant Audio Data".

[20] IETF RFC 2326: "Real Time Streaming Protocol (RTSP)".

[21] IETF RFC 2778: "A Model for Presence and Instant Messaging".

[22] IETF RFC 2779: "Instant Messaging / Presence Protocol Requirements".

[23] IETF RFC 2810: "Internet Relay Chat: Architecture".

[24] IETF RFC 2811: "Internet Relay Chat: Channel Management".

[25] IETF RFC 2812: "Internet Relay Chat: Client Protocol".

[26] IETF RFC 2813: "Internet Relay Chat: Server Protocol".

[27] IETF RFC 2916: "E.164 number and DNS".

[28] IETF RFC 3102: "Realm Specific IP: Framework".

[29] IETF RFC 3235: "Network Address Translator (NAT)-Friendly Application Design Guidelines".

[30] IETF RFC 3261: "SIP: Session Initiation Protocol".

[31] IETF RFC 3303: "Middlebox communication architecture and framework".

[32] IETF RFC 3351: "User Requirements for the Session Initiation Protocol (SIP) in Support of Deaf, Hard of Hearing and Speech-impaired Individuals".

[33] IETF RFC 3401: "Dynamic Delegation Discovery System (DDDS) Part One: The Comprehensive DDDS".

[34] IETF RFC 3489: "STUN - Simple Traversal of User Datagram Protocol (UDP) Through Network Address Translators (NATs)".

[35] IETF RFC 3605: "Real Time Control Protocol (RTCP) attribute in Session Description Protocol (SDP)".

[36] IETF RFC 3761: "The E.164 to Uniform Resource Identifiers (URI) Dynamic Delegation Discovery System (DDDS) Application (ENUM)".

[37] IETF RFC 3762: "Telephone Number Mapping (ENUM) Service Registration for H.323".

[38] IETF RFC 3764: "enumservice registration for Session Initiation Protocol (SIP) Addresses-of-Record".

[39] IETF RFC 3859: "Common Profile for Presence (CPP)".

[40] IETF RFC 3953: "Telephone Number Mapping (ENUM) Service Registration for Presence Services".

[41] IETF RFC 4002: "IANA Registration for Enumservice 'web' and 'ft'".

[42] IETF RFC 4103: "RTP Payload for Text Conversation".

[43] ISO/IEC 10646-1: "Information technology -- Universal Multiple-Octet Coded Character Set (UCS) -- Part 1: Architecture and Basic Multilingual Plane".

[44] ITU-T H-series Recommendation - Supplement 1: "Application profile - Sign language and lip-reading real-time conversation using low bit-rate video communication".

[45] ITU-T Recommendation E.164: "The international public telecommunication numbering plan".

[46] ITU-T Recommendation F.700: "Framework Recommendation for multimedia services".

[47] ITU-T Recommendation F.703: "Multimedia conversational services".

[48] ITU-T Recommendation G.114: "One-way transmission time".

[49] ITU-T Recommendation H.221: "Frame structure for a 64 to 1920 kbit/s channel in audiovisual teleservices".

[50] ITU-T Recommendation H.224: "A real time control protocol for simplex applications using the H.221 LSD/HSD/MLP channels".

[51] ITU-T Recommendation H.225.0: "Call signalling protocols and media stream packetization for packet based multimedia communication systems".

[52] ITU-T Recommendation H.245: "Control protocols for multimedia communication".

[53] ITU-T Recommendation H.248.2: "Gateway control protocol: Facsimile, text conversation and call discrimination packages".

[54] ITU-T Recommendation H.310: " Broadband audiovisual communication systems and terminals".

[55] ITU-T Recommendation H.320: "Narrow-band visual telephone systems and terminal equipment".

[56] ITU-T Recommendation H.321: "Adaptation of H.320 visual telephone terminals to B-ISDN environments".

[57] ITU-T Recommendation H.323: "Packet-based multimedia communications systems".

[58] ITU-T Recommendation H.324: "Terminal for low bit-rate multimedia communication".

[59] ITU-T Recommendation I.130: "Method for the characterization of telecommunication services supported by an ISDN and network capabilities of an ISDN".

[60] ITU-T Recommendation T.50: International Reference Alphabet (IRA) (Formerly International Alphabet No. 5 or IA5) - Information technology - 7-bit coded character set for information interchange.

[61] ITU-T Recommendation T.120: "Data protocols for multimedia conferencing".

[62] ITU-T Recommendation T.125: "Multipoint communication service protocol specification".

[63] ITU-T Recommendation T.134: "Text chat application entity".

[64] ITU-T Recommendation T.140 (including Addendum 1): "Protocol for multimedia application text conversation".

[65] ITU-T Recommendation V.18: "Operational and interworking requirements for DCEs operating in the text telephone mode".

[66] ITU-T Recommendation V.21: "300 bits per second duplex modem standardized for use in the general switched telephone network".

[67] ITU-T Recommendation V.23: "600/1200-baud modem standardized for use in the general switched telephone network".

[68] ITU-T Recommendation V.61: "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".

[69] ITU-T Recommendation V.150.0: "Modem-over-IP networks: Foundation".

[70] ITU-T Recommendation V.152: "Procedures for supporting voice-band data over IP networks".

[71] Council Directive 2000/78/EC of 27 November 2000 establishing a general framework for equal treatment in employment and occupation (OJ L 303/16).

[72] Directive 2002/21/EC of the European Parliament and of the Council of 7 March 2002 on a common regulatory framework for electronic communications networks and services (Framework Directive) (OJ L 108/33).

[73] Directive 2002/22/EC of the European Parliament and of the Council of 7 March 2002 on universal service and users' rights relating to electronic communications networks and services (Universal Service Directive) (OJ L 108/51).

[74] Directive 2004/18/EC of the European Parliament and of the Council of 31 March 2004 on the coordination of procedures for the award of public works contracts, public supply contracts and public service contracts (OJ L 134/114).

[75] ITU-T Recommendation H.248.19: "Gateway control protocol: Decomposed multipoint control unit, audio, video and data conferencing packages".

[76] ITU-T Recommendation H.263: "Video coding for low bit rate communication".

[77] ITU-T Recommendation G.723.1: "Speech coders: Dual rate speech coder for multimedia communications transmitting at 5.3 and 6.3 kbit/s".

[78] ITU-T Recommendation G.728: "Coding of speech at 16 kbit/s using low-delay code excited linear prediction".

[79] ITU-T Recommendation T.124: "Generic Conference Control".

[80] IETF RFC 2396: "Uniform Resource Identifiers (URI): Generic Syntax".

3 Definitions and abbreviations

3.1 Definitions

For the purposes of the present document, the following terms and definitions apply:

buddy list: list of people a user wishes to communicate with and know their presence status

EXAMPLE: offline, online, online but busy, online but away, online but unavailable, online but on the phone etc. (Used in e.g. instant messaging (IM)) (see also presence).

call stateful proxy: proxy which retains state for a dialog from the initiating INVITE to the terminating BYE request

calling line identification (presentation): supplementary service which provides the called party with the possibility of receiving identification of the calling party

chat: Internet service whereby two or more users can exchange text messages in substantially real time

duplex communication: mode of operation by which information can be transferred in both directions simultaneously between two points

firewall: means of preventing external parties from directly accessing internal network resources

NOTE: All signalling to internal network resources are directed via an entity dedicated to that purpose.

gateway: node in a communications network equipped for interfacing with another network that uses different protocols

NOTE: A gateway may contain devices such as protocol translators, impedance matching devices, rate convertors, fault isolators or signal translators as necessary to provide system interoperability. It also requires that mutually acceptable administrative procedures be established between the two networks.

host environment: session environment where the text component is added

EXAMPLE: Circuit switched voice, IP multimedia, etc.

indirect access: access to services provided by a public network operator or public service provider via the infrastructure of another operator

lag: failing behind or retardation of one phenomenon with respect to another to which it is closely related; time delay

legacy textphone: textphone communicating by means of a V.18 compatible modem over an analogue network

multimedia: availability of more than one medium of communication (text, graphics, video and/or audio)

netlag: time delay over the Internet (see also lag)

presence: instantaneous knowledge that someone is available, online and reachable via instant messaging

presentity (presence entity): any uniquely identifiable entity that is capable of providing presence information to presence service

relay service: telecommunications service that enables users of different modes of communication to interact by providing conversion between the modes of communication

signing: use of sign language

sign language: language that uses a system of manual, facial, and other body movements as the means of communication

simplex communication: mode of communication by which information can be transmitted in either direction but not simultaneously, between two points

stateful proxy: logical entity that maintains the client and server transaction state machines defined by this specification during the processing of a request

NOTE 1: Also known as a transaction stateful proxy.

NOTE 2: A stateful proxy is not the same as a call stateful proxy.

stateless proxy: logical entity that does not maintain the client or server transaction state machines defined in this specification when it processes requests

NOTE: A stateless proxy forwards every request it receives downstream and every response it receives upstream.

text conversation: exchange of text between users in two or more locations perceived as being transmitted and displayed in real time on a character by character basis

text telephone service: audiovisual conversation service providing bi-directional real time transfer of text and optionally audio between users in two locations

NOTE: Audio may be transmitted alternating with text or simultaneously with text.

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

videotex: interactive service generally as described in ITU-T Recommendation F.300 (see Bibliography) which provides communication with data bases and other computer based applications via telecommunications networks

videotex terminal: terminal in a Videotex service which typically includes a keypad and a visual display unit or a suitably modified television receiver

3.2 Abbreviations

For the purposes of the present document, the following abbreviations apply:

3GPP	Third Generation Partnership Project
ALGs	Application Layer Gateways
CEPT	Conférence des Administrations Européennes des Postes et Télécommunications
CLI	Calling Line Identity
CLIP/CLIR	Calling Line Identification Presentation/Calling Party Identification Restriction
COST	European Cooperation in the field of Scientific and Technical research
CPIM	Common Presence and Instant Messaging
CPP	Common Profile for Presence
CTM	Cellular Text telephone Modem
DDDS	Dynamic Delegation Discovery System
DNS	Domain Name Service
DTMF	Dual Tone Multi Frequency
DUNDi	Distributed Universal Number Discovery
DUST	Duplex Universal Speech and Text
E2E	End to End
EDT	European Deaf Telephone
EIA	Electronic Industries Association (USA)
ENUM	Electronic NUMbering
FE	Functional Entity
FSK	Frequency Shift Keying
GPRS	General Packet Radio Service
GSM	Global System for Mobile communications
GTT-IPMM	Global Text Telephony-IP MultiMedia
ICE	Interactive Connectivity Establishment
ICT	Information and Communications Technology
IETF	Internet Engineering Task Force
IM	Instant Messaging
IMPP	Instant Messaging and Presence Protocol
IP	Internet Protocol
IRC	Internet Relay Chat
ISDN	Integrated Services Digital Network
MCID	Malicious Caller ID
MGCP	Media Gateway Control Protocol
MIME	Multipurpose Internet Mail Extensions
NAT	Network Address Translation (also Network Address Translator)
NNI	Network to Network Interface
PABX	Private Automated Branch eXchange
PDA	Personal Digital Assistant
PSTN	Public Switched Telephone Network
PUA	Personal User Agent
QoS	Quality of Service
RTCP	Real Time Control Protocol
RTP	Real Time Protocol
RTSP	Real Time Streaming Protocol
SCN	Switched Circuit Network
SIMPLE	SIP forInstant Messaging and Presence Leveraging Extensions
SIP	Session Initiation Protocol
SRTP	Secure Real Time Protocol

STUN	Simple Traversal of UDP through NAT
TCP	Transmission Control Protocol
TE	Terminal Equipment
Text SET	Text Simple Endpoint Type
TURN	Traversal Using Relay NAT
UCI	Universal Communications Identifier
UDP	User Datagram Protocol
UML	Unified Modelling Language
UNI	User-Network Interface
UPnP	Universal Plug and Play
URI	Uniform Resource Identifier

4 General

4.1 Basic considerations

In an ordinary telephone system, any user wishing to converse with another user requires the means to set up a connection from a terminal in one network to a second terminal in the same or any other network and then to be able to have a conversation with a user at that terminal. The user need no knowledge of the network or handset used by the person he is calling in order to achieve this is end-to-end interoperability. Setting up a connection requires a means of inputting the address of the desired terminal and a means of drawing attention at the desired terminal that a connection is desired. Having set up the connection it is then required to be able to converse in any desired mode that is within the capabilities of both terminals.

The most important characteristics of a conversation are that it is both way and that it occurs in real time. In a normal spoken conversation, any sound is heard as soon as it is uttered. There is no need for a turn taking process as is necessary in some primitive radio communications. Each party perceives the other party as being present and that he has the ability to interrupt at any time. Any noticeable delay can affect the normal conversational flow and interfere with the perception of continuous contact.

Human communications takes many forms, including speech, text, gesture and visual expression and so a user may wish to have access to multimedia facilities permitting a conversation in more than one mode which may also include the video mode. The user may be connected to the PSTN, to a private network, to a mobile service or to the Internet. All of the combinations of these factors that are within the capabilities of a user's terminal and that of the called party should be possible.

In multimedia telephony, a number of modes are available. Speech only, text only, speech plus text, video only, speech plus video, text plus video, and speech/text/video simultaneously. Which modes are possible are dependent on the terminal capabilities and the characteristics of the network. Which mode is preferred is dependent on the user, his abilities, whether he is making or receiving the call, his personal attitude to privacy, and the knowledge that he has available about the other party on the call. A user may wish to answer the call in one mode and add or change to another mode once the identity of the caller and/or the purpose of the call has been established.

4.2 Additional facilities

In addition to the basic considerations described in clause 4.1 there are a number of other facilities that users expect from any telecommunications service to which they are connected. The Universal Service Directive [73], which is called up by the Framework Directive [72] requires that all users are able to call the emergency services free of charge by using the single European emergency call number "112" and also must have the facility selectively to bar outgoing calls. All users must be able to access operator assistance and directory enquiry services and calling line identification should also be available.

Most networks offer supplementary services such as abbreviated dialling, call transfer, call forwarding, incoming call barring, anonymous call rejection and many others.

4.3 Special provisions

As users who are deaf or who are speech-impaired cannot make use of the speech only mode in any multimedia conversation it is necessary to provide some other conversational mode, e.g. conversational text. Such conversational text has some special requirements which are described in more detail in clause 6.1.3.

4.4 Current situation

Real time interactive communication with speech uses the voice telephony service, with global interoperability between many forms of networks and terminals. The corresponding text feature is a real time interactive conversational text feature, where text is sent and displayed in visual or Braille form substantially character by character.

Text telephone systems were implemented mainly for distant conversation with deaf, hard-of-hearing, speech-impaired and deaf-blind users. The text telephone systems offer a real time, character by character, conversation in text, optionally combinable with voice. The text telephone service is described in general terms in ITU-T Recommendation F.703 [47].

Text telephony is used with dedicated text telephone terminals and, with software and modems, with computers, handheld computers and programmable communication devices.

In the PSTN, seven different, openly specified systems for text telephony exist, and are used in different regions. The text feature is thus not harmonized or globally supported even though international standards exist. Some proprietary modes are also used.

Mobile and IP networks are now in common use for voice and video telephony, but it is rare for them to offer interoperable conversational text.

5 The DUST concept

The present PSTN legacy text communication systems all have limitations and can only satisfy less than half of a user's requirements that are described in clause 6.

The best method of achieving the required text conversational function is to offer a system that has facilities equal to those offered to mainstream voice users. This can be done by offering a digital network based text communication service that is designed for all EG 202 116 [3], based upon existing mainstream solutions where possible. The use of mainstream protocols facilitates multi-platform provision of text conversation services and permits interoperable European and world wide provision of services.

Clauses 7 to 13 set out the characteristic of the proposed new system which is capable of fulfilling the requirements of the user.

The title **Duplex Universal Speech and Text** with the acronym **DUST**, conveys the main special features of the communication proposed.

- **D** - The communication should be **D**uplex, i.e. it should have the ability to send and receive simultaneously. This allows interruptions to occur as in a real live conversation.
- **U** - The communication should be a **U**niversal means of communication available to all users on a worldwide basis.
- **S** - The communication should provide the facility for **S**peech or voice communications in both directions.
- **T** - The communication should provide a real time **T**ext flow.

It is evident that a move to internationally standardized protocols could increase the number of potential terminals and at the same time prevent the exclusion of textphone users from mainstream text communication.

In order for a user to have the full capabilities of DUST as described above it is necessary that a service provider offers such a service. Such a DUST service can be provided by digitally based communication with text, voice and optionally video as described in clause 9. This is readily achieved in the form of PC software, and also may be embedded in dedicated terminals. Such terminals can fulfil all user requirements listed in clause 6, and users, appreciating the higher functionality provided, can be attracted to these new services and form a pioneering user group for DUST services.

Examples of the higher functionality as compared to most PSTN text telephony implementations that should attract users are:

- Simultaneous text and voice - no cumbersome mode switching actions required.
- Simultaneous transmission in both directions, with no need for formal turn-taking.
- Full typing speed transmission.
- Support for text in any language.
- The possibility to use DUST in Wireless LANs and other wireless environments.
- The possibility to use text in video calls - or video in text calls.
- Easy to find access possibilities when travelling.
- Multi-party calls.
- Call transfer.

Nevertheless it is recognized that legacy systems and terminals as described in annex A will remain in existence for many years and so suitable migration steps are described in clause 13 which will permit coexistence with legacy systems and the effective management of the transition from old to new.

Simple small scale gateways between this form of DUST terminals and legacy text telephony can be deployed. ITU-T Recommendation H.248.2 [53] describes procedures for such gateways that are applicable even if the gateway itself is not designed according to the H.248 architecture.

In countries where text relay services and emergency services are available for PSTN text telephone users, access to these services can be provided through gateways.

In countries where text relay services and emergency services are not yet available, new services can be provided with access directly through digital networks using the same DUST standards for access to these services.

The products and methods for firewall and NAT router traversal, commonly used for SIP calls, can be validated and used for text services as well as other SIP calls.

6 User requirements

6.1 General considerations

This clause describes the requirements of text communications from the user's point of view without any technical detail. Clause 6.2 describes the technical considerations necessary to support such requirements.

6.1.1 Universality

Any voice telephone user can pick up a telephone and talk to another voice telephone user on any network anywhere in the world. Telephony services have been implemented in such a way as to ensure this end-to-end interoperability, It can be expected that a modern telephone system will offer a good quality of speech so that the resultant conversation will be quite natural and is often described as "as if they were in the same room".

Some people prefer to communicate by text and others who are deaf or who are speech-impaired are not able to take a full part in a spoken telephone conversation and so use text or a combination of text and voice. These text users need means to communicate with any other telecommunications users worldwide to a similar extent as voice telephony users.

User requirement 1: Text communication should be available on all networks on a universal basis.

6.1.2 Call setup

Users require to be able to set up a text communication as easily as setting up a speech conversation using similar means of addressing.

User requirement 2: Text communication should be as easily set up as an ordinary speech call.

6.1.3 Conversational text

One of the most disturbing effects occurring on a poor quality speech call is end-to-end delay, where there is a perceptible delay between the spoken word and it being heard at the other end. If the delay is too great, it affects the normal conversational flow and in the extreme, can lead to both parties in the call speaking at once.

This illustrates that one of the most important characteristics of a telephone conversation that makes it seem natural is that it occurs in real time just like a normal spoken conversation, where any sound is heard by both parties as soon as it is uttered. Because simultaneous two way (duplex) communication is possible, each party perceives the other party as being present and that he has the ability to interrupt or comment at any time. Any noticeable delay can affect the normal interactive conversational flow and interfere with the perception of continuous contact.

Similarly, text communication can be made conversational by making each letter appear on the screen at both ends when it is typed. In this way the two parties to the call have the feeling of being in continuous contact and are able to carry on a normal interactive two way conversation in a similar manner to a spoken conversation. As occurs with a speech call, any noticeable delay can affect the normal interactive conversational flow and interfere with the perception of continuous contact, but with text, the delays can be greater than for speech before they become a problem.

User requirement 3: Live two way conversational text communication should be available to all users with delays that do not impair normal interactive conversational flow.

6.1.4 Conversational quality

It is normal for the speech quality of a call set up on a modern telephone network to be rated "good" or "excellent" although with some terminals on some connections with IP telephony or mobile telephony, the quality may fall to levels only considered "acceptable". Nevertheless there is a normal expectation that speech will be perceived as undistorted without any gaps in transmission.

Similar qualities of transmission should be provided on a conversational text call. Both the terminal and the network should be able to support normal human typing speeds and also the speed of automatic speech to text processing.

User requirement 4: Error free live conversational text communication should be available to all users.

6.1.5 Speech communication

Many users will wish to use text and voice in various ways, some will, for example, wish to use text in one direction and speech in the other. Some will wish to use speech with text available to use for clarification where necessary.

User requirement 5: Speech communication of good quality should be available simultaneously with the text communication.

6.1.6 Display

It is easier to follow a conversation if the terminal displays the dialogue in such a way that it is easy to read the text of both parties in the conversation and also if there is sufficient text displayed to permit the conversation to be easily followed. This is a feature of a terminal, independent of any network considerations. Terminals should support a range of character sets so as to permit a number of natural languages to be displayed.

User requirement 6: Terminals should be able to display the text of both parties in the character set in which they are typed.

6.1.7 Loss in transmission

If some characters are lost in transmission it is helpful if there is an indication in the text marking the loss.

User requirement 7: Missing text should be detected and an indication should be given in the display.

6.1.8 Editing

It is of assistance if a simple set of editing functions is available, including erase last character and insert a new line.

User requirement 8: Simple editing functions should be provided.

6.1.9 Service accessibility

A text user needs to be able to receive all of the basic and supplementary services normally supplied by a network operator including any supplementary services such as abbreviated dialling, call transfer, call forwarding, incoming call barring, anonymous call rejection and many others. Some of these services are required under the provisions of European law.

The Universal Service Directive [73] requires that all users "are able to call the emergency services free of charge by using the single European emergency call number "112" " and must have the facility of selective call barring for outgoing calls. Users must be able to access operator assistance and directory enquiry services and calling line identification should also be available.

This accessibility should also be extended to any new services such as presence service.

User requirement 9: All services should operate with text in addition to other media.

6.1.10 Call progress information

The provision of services normally requires call progress indications often provided by special tones and spoken messages.

All of these services should be accessible to all users and their provision may require special arrangements for deaf users. This may require the provision of text alternatives for spoken messages or tones or arrangements in a terminal to permit the connection of some assistive device such as a flash display or a vibrator. Alternative modes of communication should be available for all of the call progress information that is normally provided in audio form in both basic and supplementary services.

User requirement 10: Alternative modes of communication should be available for all of the call progress information that is normally provided in audio form in both basic and supplementary services to make it accessible to text users.

6.1.11 Conferencing

Where conference facilities are offered it should be possible to use text conversation in such multi party sessions. The necessary text support should be provided and it should be possible for the text from any user to be provided to all other participants in the conference. In the case of a controlled conference, it should be possible for any delegate to invoke control commands. In a video conference the input from a text user should be displayed as a subtitle under his image on the screen. Such a facility can also be provided for all speakers to provide subtitling of speeches.

User requirement 11: Conferencing services should support DUST capable terminals.

6.1.12 Multimedia telephony

A number of modes are available in multimedia telephony: speech only, text only, speech plus text, video only, speech plus video, text plus video, and speech/text/video simultaneously. Which modes are available to the user depends upon the terminal and network capabilities.

User requirement 12: A DUST compliant service should offer service in all modes within the capabilities of the terminals and networks engaged on a call.

6.1.13 Terminal configurability

A user should be able to configure his terminal to satisfy his communication preferences for e.g. font size, colour and text windows.

Privacy considerations dictate that camera off and microphone-off facilities should always be available and the user should be given the choice of default mode when making or answering a call.

User requirement 13: Terminals supporting the DUST service capabilities should be configurable by the user to suit the communication preferences and abilities of the user.

6.1.14 Signing and lipreading

Some people, particularly those who are pre-lingually deaf, identify themselves as being members of the Deaf community that recognizes sign language as its primary/native language and prefers to communicate by signing rather than by text. Others use lipreading. These users need multimedia telephony capable of offering a video mode suitable for the display of signing and lipreading.

User requirement 14: Any service offering a video mode should provide a video display with a quality that is sufficient for signing and lipreading.

6.1.15 Service configurability

A user may have a preference for a particular mode dependent on his abilities, whether he is making or receiving the call, his personal attitude to privacy, and the knowledge that he has available about the other party on the call.

When a video mode is available an indication should be given to the called party when an incoming call is in video mode. More detailed information can be found in ETR 297 [6].

User requirement 15: A DUST compliant service should be configurable by the user to suit the preferences and abilities of the user.

6.1.16 Call configurability

A user should be provided with the means to request connection of text conversation from the beginning of a call and it should also be possible to add the text component at a later stage in a call that was originally established with other media connected. A user may wish to answer the call in one mode and add or change to another mode once the identity of the caller and/or the purpose of the call has been established.

User requirement 16: A user of the DUST service should be provided with the means to alter the terminal or call configuration during a call.

6.1.17 Relay services

In order to carry on a conversation between a user in one communication mode (e.g. text) with a user in a different mode (e.g. voice), a relay service is needed to convert between the two modes. Information on the requirements for such services can be found in TR 101 806 [10]. Such relay services should also provide a speech path in both directions so as to permit voice connection in either direction when desired. They can also provide the necessary protocol conversion between different text protocols and some, with video capabilities, can enable a deaf signer to interact with a voice or text user, sometimes with the text mode being used as a supplementary information channel.

User requirement 17: A capability should be provided within the network to enable communication between users of terminals that do not share common modes of communication.

6.1.18 Masquerade

Participants in a speech conversation have the opportunity to validate the identity of a caller (and possibly their age and sex) by recognizing characteristics of their voice. This facility is denied to participants in a text conversation and the risk of masquerade is consequently higher. Some means of overcoming this problem is therefore highly desirable.

User requirement 18: Means should be provided to minimize the possibility of masquerade.

6.1.19 Integrity

A telecommunications service user has the right to expect his communications to be protected against unauthorized modification.

User requirement 19: The integrity of all communication should be maintained.

6.1.20 Confidentiality

A telecommunications service user has the right to expect his communications to be treated as confidential.

User requirement 20: All communication should be treated as confidential.

6.2 Technical considerations

This clause sets out the technical considerations necessary to meet the user requirements set out in clause 5.1.

6.2.1 Universality

To provide **user requirement 1**, to ensure that Duplex Universal Speech and Text (DUST) communication is available on a universal world wide basis, it is necessary that text transmission methods used must be fully compatible so as to ensure proper interoperability over all available transmission networks in a similar manner to the existing provision of world wide compatible speech networks.

Technical requirement 1: Global Standards that are compatible with international transmission networks are needed to ensure interconnection and interoperability of conversational text communication terminals.

6.2.2 Call setup

To meet **user requirement 2**, that text communication should be as easily set up as an ordinary speech call, the means of setting up a text call on any terminal on any network should follow similar procedures to those for setting up a speech call to another speech terminal on its own network or on any network throughout the world.

Technical requirement 2: DUST compliant services should be addressable within the international telecommunications numbering space.

6.2.3 Conversational text

To satisfy **user requirement 3**, to provide a normal interactive conversational flow, it is necessary to control the delay between data entry and data display for each character in a conversation. As is readily understood, interactive telephone conversations may be affected by end to end delays lower than 100 ms (ITU-T Recommendation G.114 [48]), and within the range 150 ms to 400 ms difficulties may arise for interruptability and normal flow of conversation (ETR 250 [5]). Delays greater than 400 ms are normally considered unacceptable for speech.

In a manner analogous to the grading of voice services, ITU-T Recommendation F.700 [46] describes two levels of subjective quality for text conversation: "usable" and "good", where the quality level "good" indicates a conversational experience with delays that are not disturbing for the conversation and where the quality level "usable" may include delays that are clearly noticeable by the user, but not so disturbing that the user would wish to give up with the conversation.

When the mean end-to-end time between entry and display of each character in a conversation is less than one second (measured over one minute) it may be expected that the user would perceive the quality as "good".

When the mean end-to-end time between entry and display of each character in a conversation is less than two seconds (measured over one minute) it may be expected that the user would perceive the quality as "usable".

Technical requirement 3: The delay should be limited in accordance with ITU-T Recommendation F.700 [46].

6.2.4 Conversational quality

To provide the uncorrupted live conversational text communication required by **user requirement 4**, a user should feel that he is receiving the text as it is typed without any additional errors being introduced by the system.

With regard to errors, ITU-T Recommendation F.700 [46] again describes two levels of subjective quality for text conversation: "usable" and "good", where the quality level "good" indicates a conversational experience with errors in text transfer that are so seldom that they do not disturb the communication any more than mis-typing and other human errors and where the quality level "usable" may include errors that are clearly noticeable by the user, but not so disturbing that the user would wish to give up with the conversation.

Where the character corruption rate is less than 0,2 % it may be expected that the user would perceive the quality as "good".

When the character corruption rate is less than 1 % it may be expected that the user would perceive the quality as "usable".

Technical requirement 4: To prevent corruption of the text display, the character corruption rate should be limited in accordance with ITU-T Recommendation F.700 [46].

The text transmission should permit a rate of at least 30 characters per second (measured over one minute).

6.2.5 Speech communication

To meet **user requirement 5**, that speech communication of good quality should be available simultaneously with the text communication, the terminal software should be of adequate quality and the service provider should provide support for the speech coding used.

In IP telephony the speech coding permitted will be determined by the service provider.

Technical requirement 5: The speech coding of the terminal should comply with that specified by the service provider.

6.2.6 Display

User requirement 6, that terminals should be able to display the text of both parties, is solely a terminal function and may be considered to be a function of the market place rather than of standardization. Nevertheless designers manufacturers and suppliers could benefit from guidance on the types of display likely to be preferred by users and by the characteristics of such a display. Terminals should also support a range of character sets so as to permit a number of natural languages to be displayed.

Technical requirement 6: Guidance can be found in ETR 333 [7] and ITU-T Recommendation T.140 [64]. Terminals should provide support for all characters in ISO/IEC 10646-1 [43].

6.2.7 Loss in transmission

In order to meet **user requirement 7,** that missing text should be detected and that an indication should be given in the display, it is necessary to make provision to detect that text is missing and to specify a standard symbol to give an indication in the text marking the loss.

For SIP and H.323 terminals on IP networks, missing text detection may be performed in the terminal by applying RFC 4103 [42]. For terminals on other networks, new standards may be required.

Technical requirement 7: Missing text should be detected. Missing characters should be indicated using the replacement character 0xFFFD of ISO/IEC 10646-1 [43].

6.2.8 Editing

The requirement that simple editing functions should be provided that is contained in **user requirement 8** can be satisfied by providing facilities to erase the last character and to insert a new line. "Control Characters" are required which can be transmitted to indicate these functions. ITU-T Recommendation T.140 [64] defines suitable editing control characters.

Technical requirement 8: Facilities should be provided to erase the last character and to insert a new line.

6.2.9 Service accessibility

In order to satisfy **user requirement 9,** that text users be provided with full access to all basic and supplementary services (including emergency services) that are normally provided to speech users of the network to which they are connected, such services should be provided in a form compatible with the text medium.

Technical requirement 9: All services should be provided in a form compatible with the text medium.

6.2.10 Call progress information

In order to provide **user requirement 10** that alternative modes of communication should be available for all of the call progress information that is normally provided in audio form in both basic and supplementary services, i.e.:

1) a consistent vocabulary of call progress information should be used;

2) a text equivalent of any prompts or call progress information should be provided by the service provider, or if not;

3) terminals will be required which provide a conversion of such information to some visual mode;

4) some provision for the connection of a suitable assistive device should be made at terminals.

Standards documentation is needed which gives guidance on the provision of call progress information in alternative forms.

Technical requirement 10: Guidance should be provided on the provision of call progress information in alternate forms.

6.2.11 Conferencing

Where conference facilities are offered, it should be possible to use text conversation in such multi party sessions as called for in **user requirement 11**. The necessary text support should be provided and it should be possible for the text from any user to be provided to all other participants in the conference. In the case of controlled conferences, it should be possible for any delegate to invoke control commands. In a video conference the input from a text user should be displayed as a subtitle under his image on the screen. Such a facility can also be provided for all speakers to provide subtitling of speeches.

Technical requirement 11: Standards should be identified for the support of multi-party communication using text.

6.2.12 Multimedia telephony

In order to offer service in all modes within the capabilities of the terminals and networks engaged on a call so as to satisfy **user requirement 12**, it is necessary that there are standard protocols available to provide the necessary call set up information and transport allowing any combination of media.

Technical requirement 12: Standard protocols are needed for the protocols at the start of a communication session to ensure support of text and other media.

6.2.13 Terminal configurability

User requirement 13 requires that terminals supporting the DUST service capabilities should be configurable by the user to suit the communication preferences and abilities of the user. This is a function of each individual terminal and may be considered to be the responsibility of the market place rather than standardization. Nevertheless users would benefit from the provision of guidance to designers, manufacturers and suppliers of such equipment.

Technical requirement 13: Guidance should be provided on the likely preferences and abilities that need to be catered for when configuring a terminal.

6.2.14 Signing and lipreading

ITU-T H-series Supplement 1 [44] describes quality requirements of low bitrate video communication for sign language and lipreading. In order to provide a sufficient quality of display in the video mode a minimum picture rate of 20 images per second is necessary. The resolution should be 256×192 pixels or higher.

Technical requirement 14: Any service offering a video mode should provide a display at least in accordance with ITU-T H-series Supplement [44].

6.2.15 Service configurability

User requirement 15 specifies that a DUST compliant service should be configurable by the user to suit the preferences and abilities of the user. This would enable networks to provide appropriate facilities to a DUST user such as automatic call diversion to a relay service.

Standards documentation is needed which gives guidance on the likely preferences and abilities that need to be catered for when configuring a service.

Technical requirement 15: Guidance should be provided on the likely preferences and abilities that need to be catered for when configuring a service.

6.2.16 Call configurability

For a user of the DUST service to be provided with the means to alter the terminal or call configuration during a call as set out in **user requirement 16**, it is necessary to provide for additional signalling during the session. This signalling will require the use of standard protocols.

Technical requirement 16: Standards need to be identified and implemented for any signalling protocols required for call configuration during a communication session.

6.2.17 Relay services

To meet **user requirement 17**, some service is needed to convert between the text and speech mode. Such services should also provide a speech path in both directions so as to permit voice connection in either direction when desired. They should make provision for legacy text systems and should provide video capabilities to enable a deaf signer to interact with a voice or text user.

Technical requirement 17: A formal standard is needed to implement the recommendations given in TR 101 806 [10].

6.2.18 Masquerade

To provide for **user requirement 18**, that means should be provided to validate the source of text, it is necessary to make provision for third party verification of the identity of the calling party. It would be preferable to be able to validate and authenticate the identify through a trusted third party. Verification and authentification should be available from wherever they are calling.

Technical requirement 18: Facilities should be provided for the verification, validation and authentication of the identity of a caller.

6.2.19 Integrity

In order to meet **user requirement 19** that the integrity of all communication should be maintained it is necessary to extend the signalling protocols to ensure that the integrity can be checked. This may be performed using a checksum.

Technical requirement 19: A set of standards are needed to ensure that the integrity can be checked across a variety of networks taking account of malicious and non-malicious alteration of content.

6.2.20 Confidentiality

In order to meet **user requirement 20** that all communication should be treated as confidential it is necessary either to protect the transmission path from malicious eavesdropping or, if this is not feasible, to encode the content in such a way that meaning is only visible to the end party.

Technical requirement 20: A set of standards are needed to ensure that the confidentiality can be maintained across a variety of networks taking account of malicious eavesdropping.

6.3 User requirements from Legislation

There are some legislative requirements that can apply to the provision of facilities for disabled users. Examples are given below.

6.3.1 Directive 2000/78/EC

The Directive 2000/78/EC [71] lays down a general framework for combating discrimination on the grounds of religion or belief, disability, age or sexual orientation as regards employment and occupation, with a view to putting into effect in the Member States the principle of equal treatment. In order to guarantee compliance with the principle of equal treatment in relation to persons with disabilities, article 5 requires that employers shall take appropriate measures, where needed in a particular case, to enable a person with a disability to have access to, participate in, or advance in employment.

6.3.2 Directive 2002/21/EC

The Framework Directive (2002/21/EC) [72], states in article 8.4 that:

"The national regulatory authorities shall promote the interests of the citizens of the European Union by *inter alia*

(a) ensuring all citizens have access to a universal service specified in Directive 2002/22/EC [73] (Universal Service Directive); ...

(c) contributing to ensuring a high level of protection of personal data and privacy: ...

(e) addressing the needs of specific social groups, in particular disabled users."

6.3.3 Directive 2002/22/EC

The Universal Service Directive (2002/22/EC) [73], states in article 7 that:

"1) Member States shall, where appropriate, take specific measures for disabled end-users in order to ensure access to and affordability of publicly available telephone services, including access to emergency services, directory enquiry services and directories, equivalent to that enjoyed by other end-users.

2) Member States may take specific measures, in the light of national conditions, to ensure that disabled end-users can also take advantage of the choice of undertakings and service providers available to the majority of end-users."

In article 26 it requires that "all end-users of publicly available telephone services ... are able to call the emergency services free of charge, by using the single emergency call number "112" ."

6.3.4 Directive 2004/18/EC

The Public Procurement Directive (2004/18/EC) [74] in article 23 requires that "technical specifications ... shall be set out in contract documentation" and states that "Whenever possible these technical specifications should be defined so as to take into account accessibility criteria for people with disabilities or design for all users".

7 Stage 1 outline

7.1 Introduction

ITU-T Recommendation I.130 [59] defines a staged method of development of protocols as follows:

- Stage 1: Specify requirements from the user's perspective.
- Stage 2: Develop a logical model to meet those requirements.
- Stage 3: Develop a detailed specification of the protocol.

The outline contents of a stage 1 definition are given in this clause.

7.2 Definition

DUST is a service offering conversational text and speech capability to the telephony user. The end-user communicates by means of a terminal with a means for entering and display of the text component, and a telephone handset for the speech component. Outgoing text communication is usually entered via an alphanumeric keypad and incoming text communication is usually displayed on a screen. Call setup, cleardown, and the use of supplementary services should be similar to those functions of a conventional telephone set.

Where the incoming call is from a conventional telephone set the receiving user should be able to choose to receive the spoken communication in the form of text or as speech. Outgoing text from a DUST user should be received by the user of a conventional telephone set in the form of speech.

A DUST service should support conversational text and speech calls to an emergency service and to a relay service.

7.3 Description

7.3.1 General description

The terminal used for DUST may be considered as both a specialization (i.e. a text terminal) and a composition of a conventional telephone terminal (i.e. combines capabilities of text, speech and video terminals). This is illustrated using UML classes in figures 1 and 2.

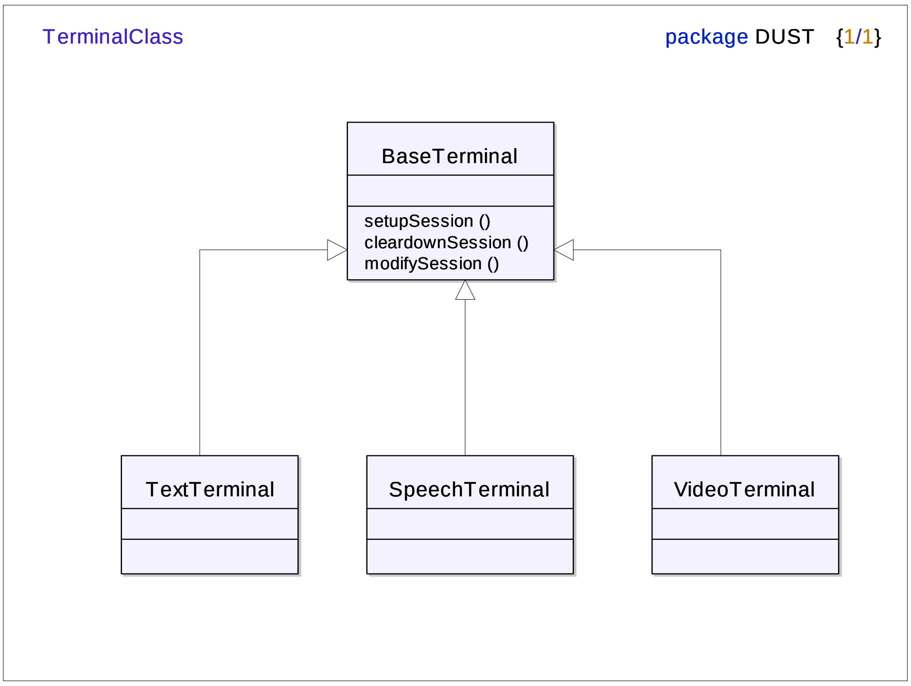


Figure 1: UML diagram showing text, speech and video terminals as specializations of a base terminal

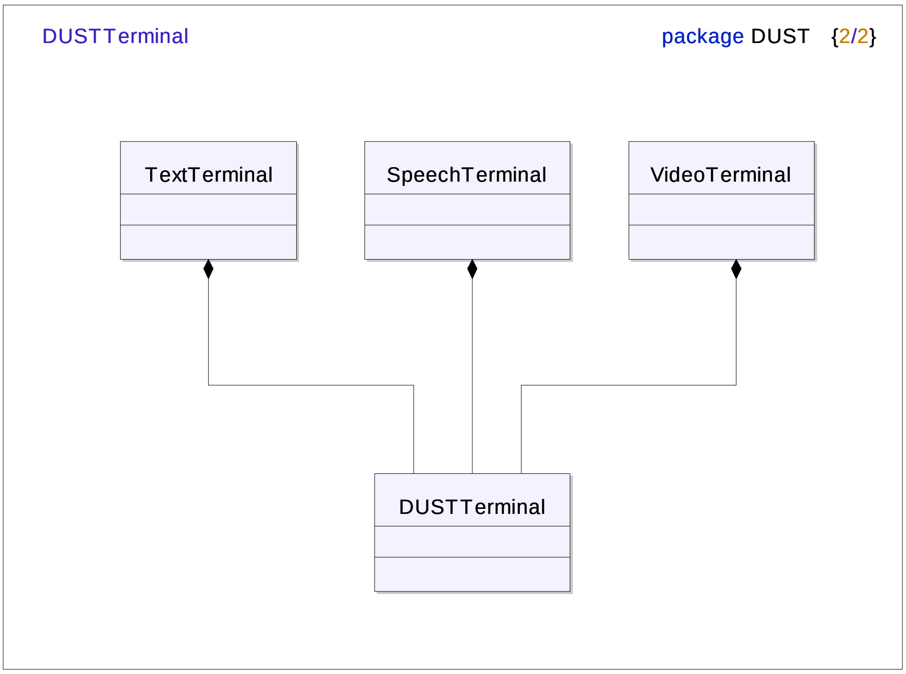


Figure 2: DUST terminal may be a composition of other terminal specializations

The transmission (bearer) service used for conversational text should give the appearance to the user of real time character by character transmission as described in clause 6.2.3.

When establishing a conversational text session (call), the initiating equipment should be able to uniquely identify to the network the type of session required, i.e. conversational text.

7.3.2 Procedures

7.3.2.1 Provision/withdrawal

The DUST service should be available to all DUST subscribers at all times.

Where a subscriber moves the physical point of attachment of the terminal the service should be available at the new point of attachment without prior notification. i.e. the DUST service is associated with the subscriber and not tied to a single network point.

7.3.2.2 Normal procedures

7.3.2.2.1 Activation/deactivation

DUST service shall be permanently activated.

7.3.2.2.2 Invocation and operation

DUST calls shall be invoked by the calling user (party A) requesting a connection of defined parameters to a second party (party B).

Party A may indicate a preferred carrier of the connection at invocation.

Party A may indicate its own name at invocation to allow the called party to identify the calling party.

Either party to the DUST call or the serving network shall be able to remove the temporary logical association formed on invocation of the service.

During call set-up transcoding between different codecs (e.g. text to speech, speech to text) may be established.

7.3.2.3 Exceptional procedures

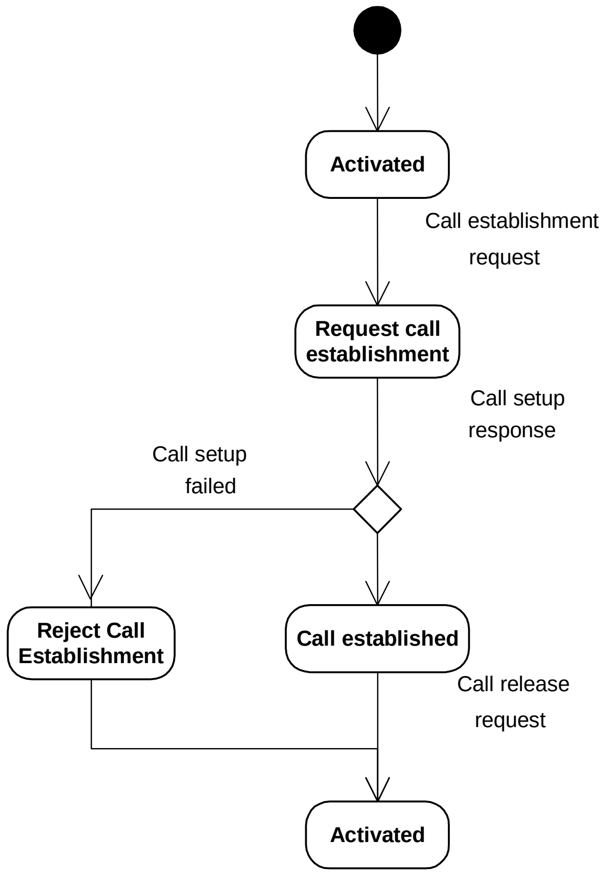
If the network cannot complete the connection to party B then party A shall be notified.

If any of the following failure conditions are detected within a domain the association shall be removed in each domain involved and any associated resource released. The domain that detects the failure shall supply a reason for failure to each other domain. Failure conditions:

- No route found to called party.
- Requested policy not supported.
- Required transport resources not available.
- Reserved media and transport resources released due to time out.
- No compatible codec supported by called party.
- Called user busy (number of simultaneous calls exceeded).
- Call release before call set-up completed.

7.4 Overall behaviour

The UML activity diagram in figure 3 shows the dynamic simple call signalling for a DUST system providing simple call service.



NOTE: The service behaviour model in figure 3 defines global states of the simple call service, hence the same state names are not re-used in the behaviour descriptions of the individual functional entities.

Figure 3: Overall behaviour of simple call service signalling

8 Stage 2 outline

The stage 2 model for DUST, the overall description of the organization of the network functions required to map the service requirements into network capabilities, is described in annex B.

When applying the model, conversational text and audio should be included in the media handled in the call.

9 Host environments

9.1 Introduction

The user requirements for DUST can be implemented in a number of ways using existing or future sets of protocols and services. This clause identifies a number of protocol host environments that may be used for implementing a DUST service.

DUST can best meet its requirements to provide global interoperability by being implemented by extending a voice telephone service or videophone service by the addition of a text capability.

Especially well suited to host this extension are the protocol environments where the media to be used in a call are negotiated by means of a call control protocol, opening transport channels for each agreed medium. Each protocol environment where DUST can be implemented is called a host environment.

Each protocol host environment may be implemented in various ways to achieve a DUST service. Three examples are given below:

EXAMPLE 1: In the simplest case two terminals may each implement the protocol stacks needed to use the host environment protocol and the DUST additions so as to be able to call and have a conversation over a basic network.

EXAMPLE 2: In more advanced cases the terminals rely on a set of service functions provided by a DUST service provider in the network environment in order to simplify the addressing, to enable connectivity through network elements, to provide quality of service and to support supplementary services etc.

EXAMPLE 3: Example 2 may be enhanced by use of a managed service to provide some guarantees of quality and reliability.

This clause contains a general discussion of host environments, while specific stage 3 aspects of a set of initially imagined host environments are found in annexes to the present document.

9.2 Requirements on host environments

A suitable host environment for DUST provides a service and protocol environment where at least voice and text calls are supported.

The following services are required to be supported by the infrastructure of the networks that support DUST:

- Conversational text and speech.
- Conversion between telephone speech and conversational text and speech (usually called a "relay service").

Furthermore, the following requirements apply:

- The host environment should be an open standardized environment.
- It should allow simultaneous voice and text.
- It should be possible to meet the requirements in ITU-T Recommendation F.703 [47] for text telephony or total conversation in the environment.
- It should be possible to fulfil the user requirements on DUST.
- Media negotiation should be done in a separate control channel.

When examples are given in the present document, they are given for a SIP host environment.

For each host environment there should be descriptions at least of the following items:

- Network environment.
- Control protocol.
- Text presentation protocol, usually ITU-T Recommendation T.140 [64].
- Text transport protocol.
- Control protocol action to declare text capabilities and open text transmission.
- Default codecs.

- Network transmission considerations.
- DUST interoperability considerations.
- PSTN interoperability.

9.3 Regulated environments

Where a DUST service is provided in a regulated environment, the following network or user services will need to be deployed in order to comply with regional or national regulations for privacy and law enforcement (this may be an incomplete list).

9.3.1 CLIP/CLIR

The DUST service should be able to support the user preference for display or restriction of identity. The user preference may be overruled only for emergency calls (112).

9.3.2 MCID

A DUST user should be able to report an incoming caller as malicious and the DUST service provider should provide means to identify and trace such calls.

9.3.3 Lawful Interception

The DUST service provider should ensure that the conversational text service is able to be lawfully intercepted and also ensure that the result of the interception can be handed over to the Law Enforcement Agencies in a manner consistent with the requirements stated in TS 101 331 [11].

9.4 Target implementation

DUST can be implemented in a number of host environments where voice telephony or multimedia conversational services are defined. The following environments are seen as possible host environments and their characteristics are briefly described in annexes to the present document.

- IP networks and RFC 3261 [30] see annex C.
- 3GPP IP Multimedia subsystem and RFC 3261 [30], see annex F.
- IP networks and ITU-T Recommendation H.323 [57], see annex D.
- ISDN networks and ITU-T Recommendation H.320 [55], see annex E.
- Data conferencing in any network, ITU-T Recommendation T.120 [61], combined with audio transmission, see annex G.
- 3GPP mobile network videophone service and 3G.324, see annex F.

9.5 Terminals supporting DUST

A user terminal for DUST will fall within a continuum of implementations from hardware without built-in intelligence, to a combined hardware and software platform with advanced intelligence. Examples are given in the following lists:

- IP terminal types with capabilities for text, voice and optionally video.
- Personal computer + SIP SW client.
- 3GPP IP Multimedia SIP terminal with text, speech and optionally video.
- Personal computer + H.323 SW client.

- SIP HW telephone with real time text capabilities.
- H.323 HW telephone with real time text capabilities.
- ISDN videophone with text capability.
- 3G/UMTS circuit switch terminal with text, voice and optionally video capability.

9.6 DUST interoperability targets

DUST services should provide interoperability with alternative implementations of DUST without loss of any functionality (such as may be caused by transit through the PSTN).

DUST services should provide interoperability with a number of real time text conversation environments with as little loss of functionality as possible.

Examples of targets for interoperability from a DUST capable terminal are given in the following list.

- SIP terminal with DUST capability.
- Interoperability in voice mode with any voice terminal reachable through public networks.
- DUST capable terminals in any DUST host environment.
- IP terminal with capability for text only:
  - GSM GPRS terminal.
- SCN terminal types with capabilities for text and voice:
  - PSTN textphone.
  - GSM terminal with text and voice capability.
- Hybrid terminal environments:
  - PSTN/ISDN terminals connected to an adapter for SIP telephony.
  - PSTN/ISDN terminals connected to an adapter for ITU-T Recommendation H.323 [57] telephony.

9.7 Generic Firewall and NAT traversal

9.7.1 Introduction

A multimedia session signalling protocol is a protocol that exchanges control messages between a pair of agents for the purposes of establishing the flow of media traffic between them. This media flow is distinct from the flow of control messages, and may take a different path through the network. Examples of such protocols are the Session Initiation Protocol (SIP) (RFC 3261 [30]), the Real Time Streaming Protocol (RTSP) (RFC 2326 [20]) and ITU-T Recommendation H.323 [57].

These protocols, by nature of their design, are difficult to operate through Network Address Translators (NAT). Because their purpose in life is to establish a flow of packets, they tend to carry IP addresses and port numbers within their messages, which are known to be problematic through NAT (RFC 3235 [29]) and Firewalls. The protocols also seek to create a media flow directly between participants, so that there is no application layer intermediary between them. This is done to reduce media latency, decrease packet loss, and reduce the operational costs of deploying the application. However, this is difficult to accomplish through NAT and Firewalls.

Numerous solutions have been proposed for allowing these protocols to operate through NAT. These include Application Layer Gateways (ALGs), the IETF Middlebox Control Protocol (RFC 3303 [31]), Simple Traversal of UDP through NAT (STUN) (RFC 3489 [34]), Traversal Using Relay NAT (TURN) (see Bibliography), and Realm Specific IP (RFC 3102 [28]) along with session description extensions needed to make them work, such as the SDP attribute for RTCP (RFC 3605 [35]).

Unfortunately, each of these techniques has benefits and problems that make each one optimal in some network topologies, but a poor choice in others. Attempts have been made to use a combination of techniques, such as the IETF's Interactive Connectivity Establishment (ICE), but have as yet not solved the problem with all scenarios.

9.7.2 Firewalls

The generic role of a Firewall is to ensure and maintain some degree of filtering and protection against unauthorized access to internal network resources. A firewall may be implemented either in hardware or software. It works by blocking traffic based on three items of information: the source address, the destination address and port number and traffic type. Firewalls also make decisions based on the direction of traffic flow. Typically, incoming traffic (from the un-trusted, public domain) is only allowed if that session was initiated from a device on the trusted, private domain.

As a consequence it is necessary to identify the ports that need to be open or closed as required by specific applications and their underlying protocols. Whilst current firewall implementations are able to dynamically open and close multiple ports as required by signalling protocols, such as SIP, they remain ineffective at securely supporting unsolicited incoming media flows.

NAT devices prevent two-way voice and multimedia communication, because the private IP addresses and ports inserted by client devices (IP phones, DUST capable terminals, video conferencing stations etc.) in the packet payload are not routable in public networks. Thus, incoming calls that are essential in any service implementing a call model are problematic to implement with existing NAT/Firewalls.

SIP-based communication, like traditional telephony, is based on receiving incoming calls from a wide range of unknown (and therefore un-trusted) sources. However, this is not in line with the Firewall filtering policies described above.

9.7.3 NAT traversal

NAT devices translate IP addresses and port numbers in private address ranges into public addresses when traffic traverses between private and public networks. This allows a limited number of public IP addresses to serve the needs of even the largest corporation.

Each device in the network has its own private IP address. Traffic (a media stream, for example) sent to a device on the public network is dynamically assigned a specific port number at the public address by the NAT. The NAT maintains a "table" that links private and public addresses and port numbers. It is important to note that these "bindings" can only be initiated by outgoing traffic.

The NAT acts in a similar way to a PABX. Users of the PABX can dial out using one of the few public telephone lines (equivalent to public IP Addresses) that are available. The line that is used (the port number) is automatically selected and invisible to the user. Receiving an incoming call is more difficult, because the internal extension numbers are unknown to the public network. Users dialling in must be routed to an attendant to be connected to the correct extension. Clearly in the case of a NAT, there is no equivalent of an attendant so unsolicited incoming calls cannot be supported.

To complicate matters further, the end-to-end SIP messaging between clients contains details of the private IP addresses and ports that the clients (User Agents) want to use for the media flows. When the clients attempt to use these private addresses to send/receive media, the connection fails because they are not routable. This issue also applies to other signalling protocols such as ITU-T Recommendation H.323 [57] and MGCP.

Any approach to solving this problem must allow secure two-way communication - including unsolicited incoming calls - and minimize dependence on upgrading NATs or even using any specific vendor's NAT device.

9.7.4 NAT traversal solutions

There are a range of proposed solutions to NAT traversal and have varied implications with regard to security (i.e. how it solves the "Firewall problem").

Examples for solving NAT traversal are:

- Universal Plug and Play (UPnP).
- Simple Traversal of UDP Through Network Address Translation devices (STUN).

- Traversal Using Relay NAT (TURN) (see Bibliography).
- Application Layer Gateway.
- Interactive Connectivity Establishment (ICE) is a Methodology for Network Address Translator (NAT) Traversal for Multimedia Session Establishment Protocols (see Bibliography).
- Tunnelling Techniques.
- Manual Configuration.

Service Providers are encouraged to implement techniques that support and overcome NAT traversal and Firewall issues.

10 Numbering, naming and addressing

10.1 ENUM

10.1.1 E.164 to URI DDDS Application (ENUM)

ENUM is an address translation system that exists primarily to facilitate the interconnection of systems that rely on telephone numbers with those that use URIs to identify resources. ENUM (RFC 3761 [36]) discusses the use of the Domain Name System (DNS) for storage of E.164 numbers [45]. More specifically, how DNS can be used for identifying available services connected to one E.164 number [45]. It specifically obsoletes RFC 2916 [27] to bring it in line with the Dynamic Delegation Discovery System (DDDS) Application specification found in the document series specified in RFC 3401 [33].

ENUM (RFC 3761 [36]) is an extension of DNS and defines a method to translate E.164 numbers (see Bibliography), such as +33492944200, into URIs and then use this to point to resource names (URIs, RFC 2396 [80]), such as pres:user@host.com.

RFC 4002 [41] registers the Enumservices "web" and "ft" by using the URI schemes "http:", "https:" and "ftp:"

RFC 3762 [37] registers a Telephone Number Mapping (ENUM) service for H.323 "E2U+H323" according to specifications and to be used in the "service" sub-field of the "enumservice" as defined in RFC 3761 [36].

RFC 3764 [38] registers an Electronic Number (ENUM) service for the Session Initiation Protocol (SIP), pursuant to the guidelines in RFC 3761 [36]. Specifically, this document focuses on provisioning SIP addresses-of-record URIs in the "enumservice" field. Various other types of URIs can be present in SIP requests. A URI that is associated with a particular SIP user agent (for example, a SIP phone) is commonly known as a SIP contact address.

10.1.2 Enumservice Registration for Presence Services

Presence is a service defined in RFC 2778 [21] that allows users of a communications service to monitor one another's availability and disposition in order to make decisions about communicating. Presence information is highly dynamic and generally characterizes whether a user is online or offline, busy or idle, away from communications devices or nearby, and the like. Presence service is further described in annex A.

The IETF has defined a generic URI used to identify a presence service for a particular resource: the "pres" URI scheme (defined in CPP, RFC 3859 [39]). This document describes an enumservice for advertising presence information associated with an E.164 number. RFC 3953 [40] specifies an enumservice field that allows presence information to be provided for an E.164 number. This may include presence states associated with telephones, or presence of non-telephony communications services advertised by ENUM.

10.2 Distributed Universal Numbering Discovery (DUNDi)

Distributed Universal Number Discovery (DUNDi) (see Bibliography) protocol is an alternative to using ENUM, that has been proposed within the IETF for resolving phone numbers into Internet resources for contacting those phone numbers, through a peer to peer, trusted system.

DUNDi has been designed to facilitate the sharing of resources that can be used to terminate E.164 numbers by using a peer to peer system, requiring no centralized controlling authority, no single point of failure, and no enforced hierarchy. In this way, systems sharing a dialplan across a private enterprise domain or internationally in the public network can be assembled in an ad-hoc manor, while retaining confidence in the accuracy of the routes that are supplied and the security of both the queries and the answers within the trust group.

10.3 Universal Communications Identifier (UCI)

The identification requirements of users can be enhanced by the Universal Communications Identifier (UCI) architecture specified in EG 203 072 [4], which identifies a relationship between the UCI user and his Personal User Agent (PUA). The purpose of the PUA is multi-fold and can store, or have access to, current state and personal preferences information in relation to all communications services.

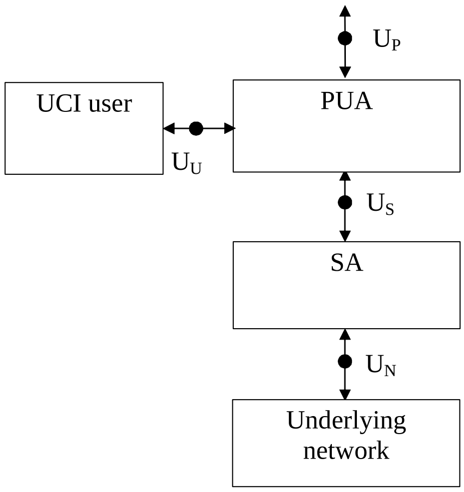


Figure 4: UCI reference points

In clause 9 of EG 203 072 [44] the use of preferred service indicators that may be used as a filtering option on call setup is introduced. A UCI user that has access to the "Preferred services indicators" of the UCI user that they wish to contact can choose an appropriate communication service to try to contact that user. By making this choice, the UCI user will improve the chance that their communication request will be accepted and it will also reduce the need for PUA to PUA negotiation or the need to use media conversion services (e.g. text to voice services).

As well as the use of "Preferred services indicators" people who have special communications needs may wish to make use of some specialized information fields. The use of such fields might be beneficial for people with disabilities who might wish to indicate either specific modalities of communications that will or will not be acceptable (e.g. voice is acceptable and text is unacceptable).

This information may be different from "Preferred service indicators" as one modality of communication may map to several communications services or specific disabilities.

The coding and sharing of disability information is a potentially very sensitive issue and will be seen as unacceptable by many people with disabilities. However some people with a disability may judge that sharing such information may bring benefits much greater than any threat from making such information public.

11 Presentation characteristics

In order to fulfil user requirements for easily readable text, and minor editing facilities, the presentation level of the real time text conversation requires to be defined. It is recommended that ITU-T Recommendation T.140 [64] is adopted for this purpose for all host environments. ITU-T Recommendation T.140 [64] does not mandate a presentation layout, but it shows some examples of suitable layouts. It is recommended that the two column view with some level of time synchronization in the display is adopted in terminal applications (see figure 5).

NOTE: ITU-T Recommendation T.140 [64] is a straightforward application of the international character set ISO/IEC 10646-1 [43] on this area. It also identifies some control functions and extension mechanisms. ITU-T Recommendation T.140 [64] describes the coding and presentation aspects, and leaves the transport of the text to be specified by other standards related to the host environment.

However, it specifies that the format of the characters in transmission shall be the UTF-8 transform, described in ISO/IEC 10646-1 [43]. UTF-8 is a coding form that transforms the characters to sequences of 8-bit bytes, where it is easy to find the beginning of a character. This can be important when recovering from loss. For figures and letters used in English, the UTF-8 code results in one byte per character, while for letters from other alphabets, there are more bytes per character.

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 5: A possible way to display a conversation with one window for each participant

Communication always includes risks of errors and occasional drop-outs. The transport mechanisms should prevent such disturbances from distorting the information as far as it is realistic. ITU-T Recommendation T.140 [64] specifies a character to be inserted in the character stream in places where the transport functions indicate that something may be missing because of a communication error. The specified character for this indication is a question mark in a rhombus. If the target system cannot display this character, it is acceptable to replace it with an apostrophe (').

The preferred mode of transmission for T.140 is to transmit each character as it is entered. If there are network capacity motivated reasons to collect characters for transmission, it may be done so, for up to a maximum of 500 ms before transmission. This limit is set in order to maintain a reasonable real time feeling in the call, although the figure of 300 ms referred to elsewhere in the present document can be considered as a preferred figure.

12 Service capability requirements

In order to provide a DUST service, the components shown in table 1 are required.

Table 1: DUST components

Component	Short description	Long description
SERVICE PROVIDER	A DUST service provider	In order for a user to use a DUST terminal it is necessary that a service provider offers a DUST service. The service may require pre-registration of the user.
SIGNAL (note 1)	Signalling of the desire of the user to establish DUST service	The requirement to use a DUST service is explicitly signalled by the end-user and identification of the DUST service needs to be maintained throughout the signalling phase. Signalling covers the following phases of a DUST call: • Setup; • [Modification]; • Cleardown.
DESCRIBE	Description of the media	The DUST service should be able to fully describe the boundary conditions for transmission of the conversational text. This should include the following parameters: • Media type; • Symmetry; • End-to-end delay; • Packetization format; • Jitter (variation in end-to-end delay).
TRANSPORT (note 2)	Definition of the bearer	It is necessary that the DUST bearer service meets the requirements identified in the media description.
NOTE 1: In order to allow the signalling to complete it may be necessary to maintain a record of the state of the call at points other than the terminals. NOTE 2: The bearer may be either packet mode or circuit mode.		

12.1 Emergency services for DUST users

As described in clause A.9.1, the Universal Service Directive [73] requires that all users are able to call the emergency service free of charge by using the single emergency call number "112". When new media, new terminal types and new networks emerge and become everyday communication means, it is important that emergency services are reachable using them. This is also applicable for DUST.

12.1.1 PSTN based Emergency service centre

When the emergency service centre is based on PSTN technology the necessary access for DUST users may require to be provided through gateways. If gateways are used they should be able to handle calls which include the text component.

12.1.2 IP based emergency service centre

When the emergency service centre is based on IP technology the necessary access for DUST users requires native support for DUST within the emergency service centre. If native support is not provided, gateways may be used.

All other aspects of emergency calls in DUST host environments should be provided as in general emergency services. These further requirements are set out in SR 002 180 [8] and SR 002 299[9].

For calls with PSTN textphones, gateways should be arranged on the normal emergency number (112), and brought into the IP technology, maintaining the capability to alternate between voice and text which may have been supported for legacy PSTN textphones.

12.2 Relay services

12.2.1 Types and purposes of relay services

As described in clause A.9.2, a relay service enables users of different modes of communication to interact by providing conversion between the modes.

The most common relay service types are text relay service, sign language relay service and speech relay service.

- Text relay services convert between text and voice.
- Sign language relay services convert between sign language and voice.
- Spoken to spoken relay services enable speech impaired telephone users and other users to interact by providing skilled assistance between them.

Relay services are usually manned services using specially trained operators, although some conversions can be provided by automatic means.

All three types of relay service can be enhanced by the use of DUST capable terminals and it is important that any future relay service offers access over IP to permit their use. Any service provider claiming to offer a DUST compliant service should offer full relay facilities.

In a text relay service, the text component of the call usually carries the whole conversation, but in some cases it may be only used for one direction of the conversation, whilst the other is conducted directly in voice.

In a sign language relay service, the text component of the call may be used occasionally to convey some fact that need exactness such as telephone numbers, note-taking, precise spelling or for clarifications that may be better made in text.

In a speech relay service, the text component can be used for occasional clarifications.

During a relayed call, the relay service can convert either the complete conversation, or just one way, depending on the capabilities and preferences of the participants.

TR 101 806 [10] gives additional guidance on all kinds of relay services.

12.2.2 Invocation of relay services

In most existing cases the invocation of relay services is a two step process. First a call is made to the relay service, and then the service connects the call to the desired destination.

This method has some drawbacks.

- It is complicated to describe, e.g. on a business card.
- It does not easily allow the use of electronic phonebooks for the destination.
- It takes extra time.

In the UK an overlay text network is provided whereby a prefix is dialled before the number of the desired destination The relay service is then invoked automatically if the call is not answered by a textphone. Such a service is described in clause 4.2 of TR 101 806 [10].

In a DUST environment setting up a call over IP, a one step calling method can more easily be arranged. RFC 3351 [32] describes some valid scenarios for relay service invocation, using SIP features and aiming at making the invocation process smooth.

Table 2 shows some possible methods to establish one step calling though a relay service.

Table 2: One step calling possibilities

Method	Description	Comment
Service indicator	Add a service indicator to the destination address	Can be made general so that the relay service is only invoked if the call participants prefer different modes
Preference and capability comparison	The users have expressed their preferences by registration or during the call. The preferences are compared with each other and with terminal capabilities, and a relay service is invoked to cover gaps or mismatches.	Requires the host environment or the answering terminal to support preference and capability registration and service invocation based on comparisons.
Separate personal address for relayed calls	By providing a separate address for calls in the mode that the user wants relayed, the call can be routed through a relay service which gets the information necessary to complete the call.	Usually requires an extra subscription for the user
Three party call invocation by one user	Any user can invoke the relay service as a third party in a call, and establish the media needed for relaying.	Does not work well in the PSTN, where the voice channel is used for text modem signals. Works better in multimedia host environments
Call forwarding by one user or terminal	The call can be forwarded to the relay service manually or by automatic comparison of preferences and capabilities. The relay service reconnects another call leg to the user.	Caller preferences may not be available for the terminals.

13 DUST and legacy systems

13.1 Alternative approaches

Any move to a new design for all IP based text communication system requires consideration of what should be done with existing legacy text communication systems and terminals. There are basically three possible ways to manage the transition:

1) Change all current terminals when the new system is installed.

2) Use a special conversion box in each terminal.

3) Provide network gateways.

Of these three options only the last is really practical. It allows new and old terminals to work together, facilitates communication between different national systems and avoids the cost and management difficulties of a single switch over date implied by both options 1 and 2.

13.2 DUST/legacy interoperability

13.2.1 Gateway support between legacy and DUST users

Apart from some proprietary devices, legacy text telephones make use of modems compatible with ITU-T Recommendation V.18 [65] using one of its submodes. Most of them can be used alternating with voice in the call. There is also a standardized mode of ITU-T Recommendation V.18 [65] (described in ITU-T Recommendation V.61 [68]) which permits the use of voice and text simultaneously. Legacy devices are generally connected within a PSTN network which is normally capable of providing a textphone service without any modification or special provisioning. Text telephony and total conversation can also be provided in other environments as described in annex A.

One protocol for a DUST compliant service is SIP with text according to RFC 4103 [42], and voice and optionally video as in general SIP calls. A DUST terminal for SIP will be connected within an IP network.

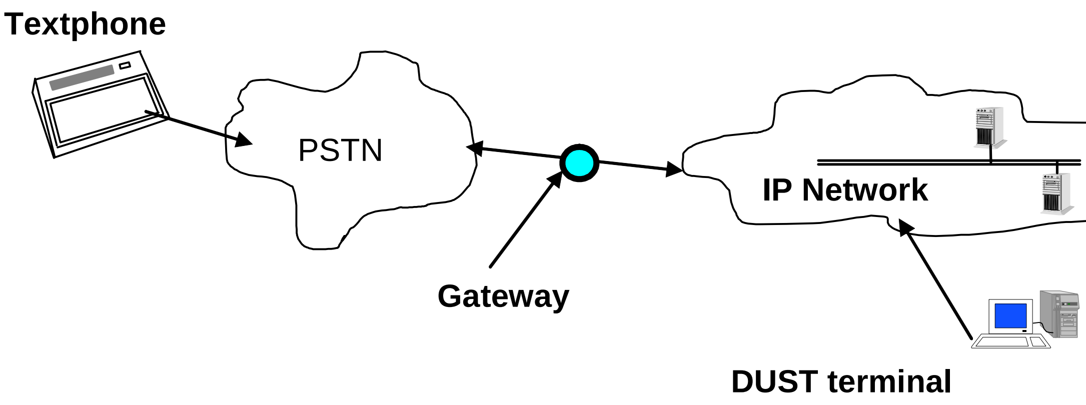


Figure 6: The use of gateways

For communication between these different worlds, a gateway is needed as shown in figure 6.

Any operator or organization providing a DUST capable service in a country where PSTN textphones are in common use would therefore need to provide access to such gateways which should at least allow alternating text and voice communication using the procedures described in ITU-T Recommendation H.248.2 [53]. These procedures can be applied to any text gateway even if the H.248 architecture itself is not used.

Using the proper gateways, calls will be possible with users and services still using legacy text conversation services, although there may be some functionality degradation due to the limitations of the legacy systems.

DUST users will be accustomed to native DUST calls supporting:

- Simultaneous text and voice - with no cumbersome mode switching actions.
- Simultaneous transmission in both directions, with no need for formal turn-taking.
- Transmission at full typing speed.
- Support for text in any language.
- The possibility to use text in video calls - or video in text calls.
- Multi-party calls.
- Call transfer.

When DUST users are connected to PSTN text telephones through gateways, they will experience lower functionality in such calls.

- Only alternating text and voice will be available. It will not be possible to talk while the other end is typing.
- It will be necessary to take strict turns in typing. Simultaneous typing may corrupt the screen output at the PSTN side, or cause missing or delayed characters.
- For some PSTN textphones systems, the speed of transmission is slower than the input to thee gateway. Buffering is therefore needed in the gateway so that text is delayed but not dropped.
- Character sets are limited to one national language per terminal. National characters from other languages will be dropped or converted.
- There will be no video support.
- Only point to point calls will be possible.
- Call transfer may not be possible.

A gateway can be made general, so that it connects in voice mode and observes the connections for text activity. If any text activity is detected, text connection and transmission can start. On the PSTN side it is essential that text connection is not tried unless there is an indication that the users want to use text. Most PSTN textphone systems can only handle alternating between voice and text, and it would disturb a voice user with modem tones in the ear if the gateway tried to connect in text mode when it was not intended.

Invoking text gateway functions requires specific planning. DUST is designed to be integrated with any SIP based telephony system. Such systems are most commonly provided with a voice gateway to the PSTN. Therefore when introducing DUST, there will be many VoIP gateways already installed that do not offer support for text.

There may therefore be a need to establish mechanisms for routing calls that may contain text through text capable gateways.

13.2.2 Procedures for text aware gateways

A text aware PSTN/IP gateway should support the auto-mode monitoring procedure as specified in ITU-T Recommendation V.18 [65]. When receiving an incoming call from the PSTN, the gateway should remain in speech mode until such time that text is detected. The gateway should attempt to establish a connection with an IP end-point specifying both text and audio media, the IP end-point should then reply with its capabilities (e.g. text only or speech and text capable)

Attempting to establish a text and audio connection with the IP end-point when text has not yet been detected on the incoming PSTN call, overcomes potential problems with carrier-less PSTN text telephones. This procedure also overcomes an additional delay of 3 s to 4 s that would result in waiting to detect modem carrier before proceeding with the call set-up to the IP end-point

With any procedure there will be compromises that have to be made, in this instance it may result that the originating text phone user believes that they have a text connection when they have not.

13.2.2.1 Scenario 1: Carrier based PSTN Text Telephone

A PSTN/IP Gateway compliant with ITU-T Recommendation V.18 [65] auto-mode will terminate calls originating from carrier based PSTN Text Telephones and should detect the presence of a text telephone. When routing the call to the IP end-point, should the request for audio and text media result in a text only supported response, then the gateway should establish a text media connection and operate in text mode. However, should the text and audio media request be unsuccessful with audio media accepted only, then the gateway should connect the call and pass the text media in the audio stream. This will enable an end-user to take the initiative to either enable text mode on their end-point or transfer the call to a text capable IP device. The IP end-point should then send a reinvite message to the gateway to include text media.

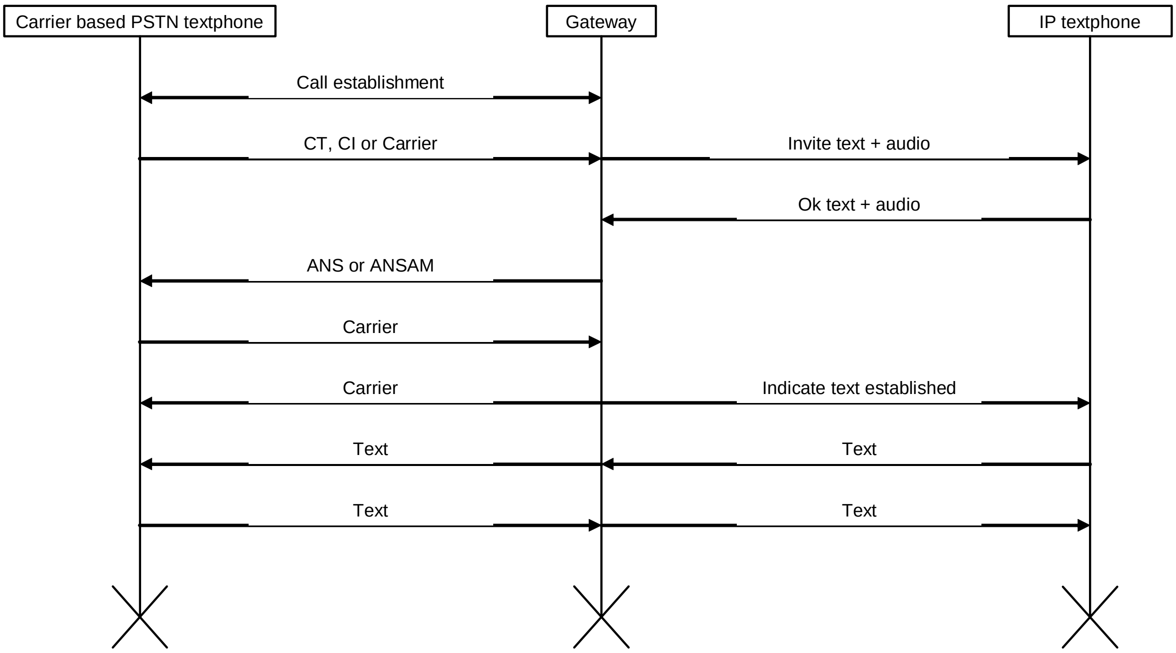


Figure 7: Scenario 1, call from carrier based textphone

13.2.2.2 Scenario 2: Carrier-less PSTN Text Telephone

A PSTN/IP Gateway compliant with ITU-T Recommendation V.18 [65] auto-mode will terminate calls originating from carrier-less PSTN Text Telephones but will not detect the presence of a text telephone until such time that the originating user sends alphanumeric characters. In this instance the gateway will remain in speech mode until such time as text (alphanumeric characters) has been detected from either side.

When routing the call to the IP end-point, should the request for audio and text media be accepted by the end-point then the gateway will be able to convey the text component. If text is first received from the IP end-point, the gateway should revert to V.18 automode answering mode in the PSTN termination, and actively try to establish text connection and resolve the textphone type, so that communication can continue in text mode.

If text is first received from the PSTN textphone, the type is resolved, and communication can be continued in the proper mode.

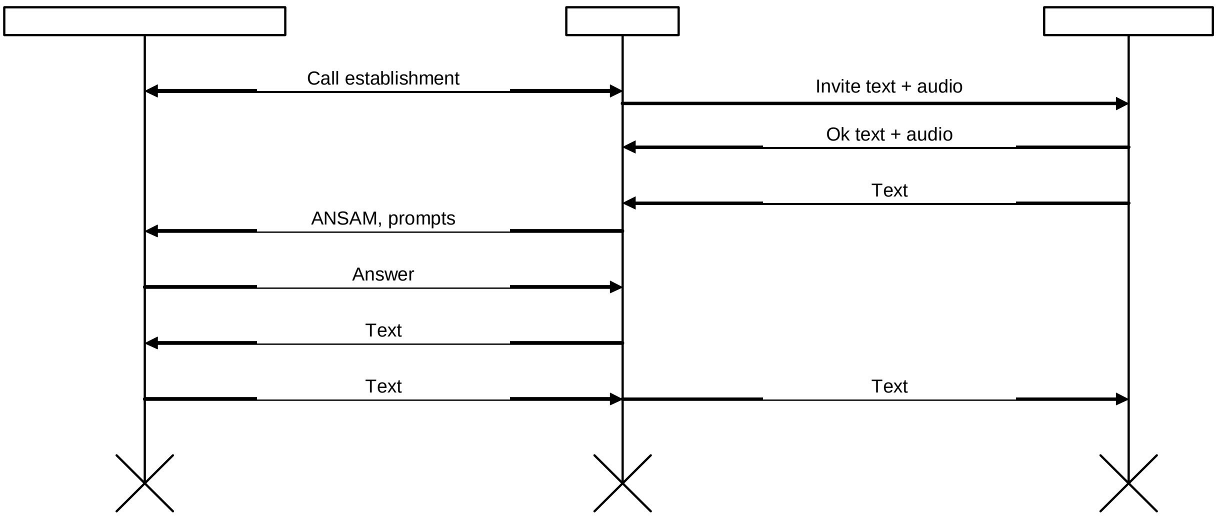


Figure 8: Scenario 2, calling from carrier less textphone

Care should be taken to keep the rate of false text detection low at the gateway, as there are no reliable mechanisms to detect carrier-less text phones until after a large part of the first valid character has been received. In order to achieve reliability in such instances it is recommended that the gateway buffer the first portion of the content of communication.

13.2.2.3 Scenario 3: IP end-point originating call to PSTN

A call invoking the DUST service from an IP end-point declaring text preference or only text capability and being routed to a PSTN terminal device should only be completed when the PSTN terminal capabilities have been established as supporting the DUST service. The PSTN/IP gateway compliant with ITU-T Recommendation V.18 [65] should start in V.18 automode originating mode, transmitting modem tones until one of the following occurs:

1) The PSTN terminal is a carrier-based device and transmits answer tone, and the textphone type can be determined by the V.18 procedure.

2) The PSTN terminal is a carrier-less device, the end user starts sending characters such that these may be detected by the PSTN/IP gateway.

3) The gateway cancels the connection effort after a timeout without detection of textphone tones.

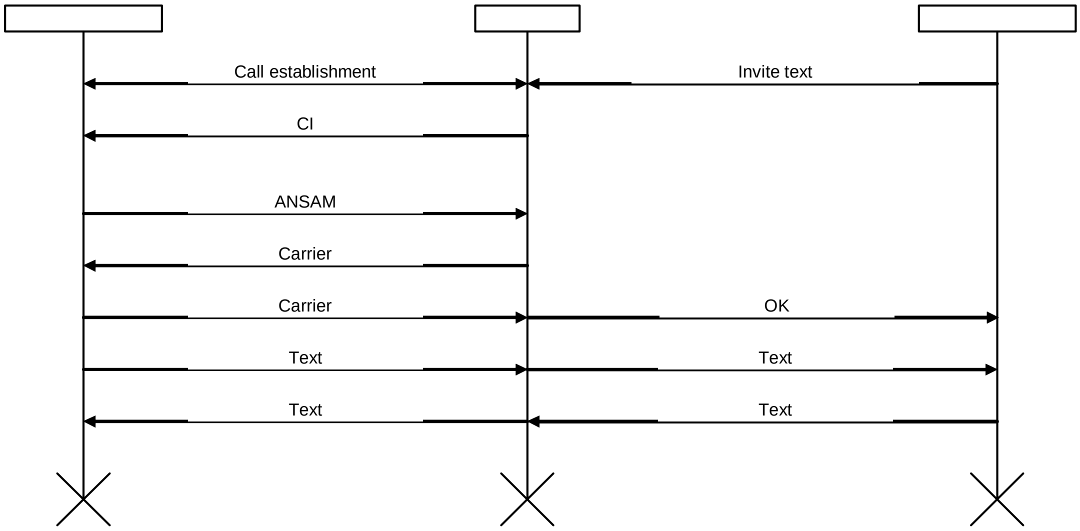


Figure 9: Example 2 - Call from IP to carrier based textphone, text only or preferred

In the instance where the IP end-point has declared both text and audio capabilities and no specific text preference, the PSTN/IP gateway compliant with ITU-T Recommendation V.18 [65] should complete the call with the IP end-point, and start V.18 automode monitoring mode towards the PSTN termination, not transmitting modem tones until one of the following occurs:

4) The PSTN terminal is a carrier-based device and transmits answer tone, leading to the gateway resolving the text telephone mode according to V.18 procedures and then entering that mode (see figure 10).

5) The PSTN terminal is a carrier-less device, the end user starts sending characters such that these may be detected by the PSTN/IP gateway, leading to the gateway detecting the text telephone mode and entering that mode (see figure 11).

6) The IP terminal transmits text characters, leading to the gateway entering V.18 automode originating mode to stimulate the PSTN terminal to reveal its type and then connecting.

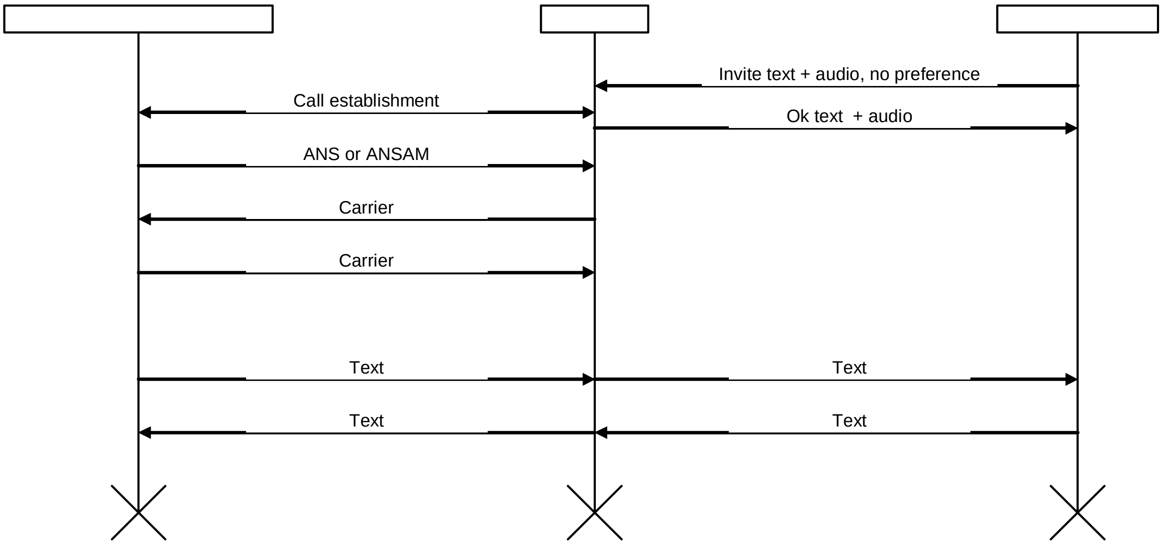


Figure 10: Example 4 - Call from IP to carrier based textphone, no text preference

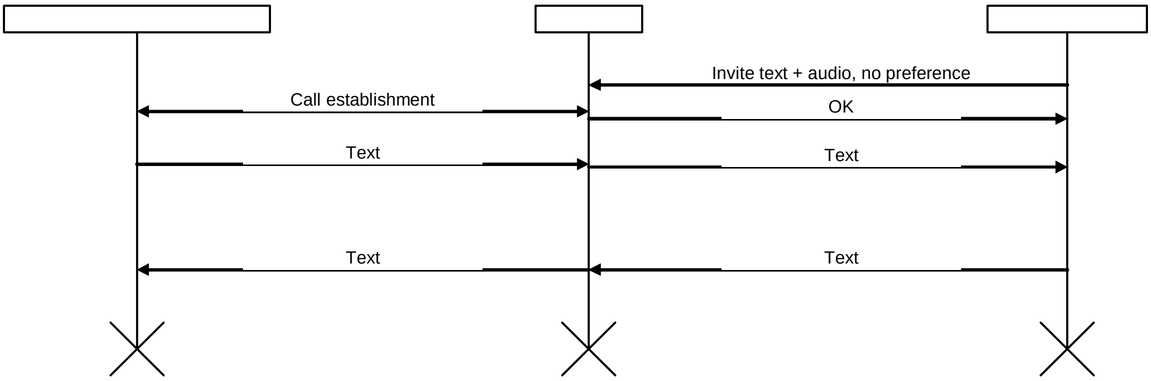


Figure 11: Example 5 - Call from IP without text preference to carrier less textphone

By this procedure the call can continue in voice mode for the whole call if so required.

13.2.3 Procedures for text specific gateways

A gateway can be made text specific.

In a text specific gateway, the gateway begins each call with an effort to connect in text mode.

Towards the PSTN side, this implies using the automode originating procedures or the automode answering procedures of ITU-T Recommendation V.18 [65] depending of the call situation.

Towards the IP side, it implies declaring text capability and opening the text channel.

13.2.4 Common procedures for conveying text in calls through a gateway.

When a call is established through the gateway, text is conveyed between the participants. There are no flow control mechanisms for the receiving terminals. Characters should be sent as soon as the conditions on the transport channel allow. A buffer should be provided for storing text while waiting for the transmit path to be enabled or for preventing text overflow when the PSTN channel output is slower than the text input. The human users are expected to have interactions during a call, so that a gateway buffer of 2k characters is sufficient. Since ITU-T Recommendation T.140 [64] is the dominating protocol for transmission of real time text conversation, it should be used as the common coding for all text exchange in the gateway.

13.2.5 Alternating text and voice through a gateway

Text capable gateways should support alternating text and voice.

On the IP side, separate channels should be established for text and voice. These are capable of simultaneous transmission of voice and text.

On the PSTN side, the gateways should support alternating text and voice.

When a connection is made with a carrierless textphone, text transmission should have priority over voice transmission on the PSTN side. If the IP user types and talks at the same time, only the text will be received by the PSTN user during that time.

When a connection is made with a carrier based textphone as shown in figure 12, the connected PSTN textphone controls the alternation between text and voice. When the PSTN textphone user breaks the carrier to use voice, the gateway detects the loss of carrier, drops its carrier, and goes to voice mode with monitoring of the connection in ITU-T Recommendation V.18 [65] automode monitor mode. When the PSTN terminal returns to text mode by turning carrier on again, this is detected in the gateway, which also turns on its carrier and returns to text mode. Return to text mode should also be done when the gateway receives valid text for transmission from the IP side.

The Interrupt command of ITU-T Recommendation T.140 [64] may also be used by the IP terminal to request a change to voice mode.

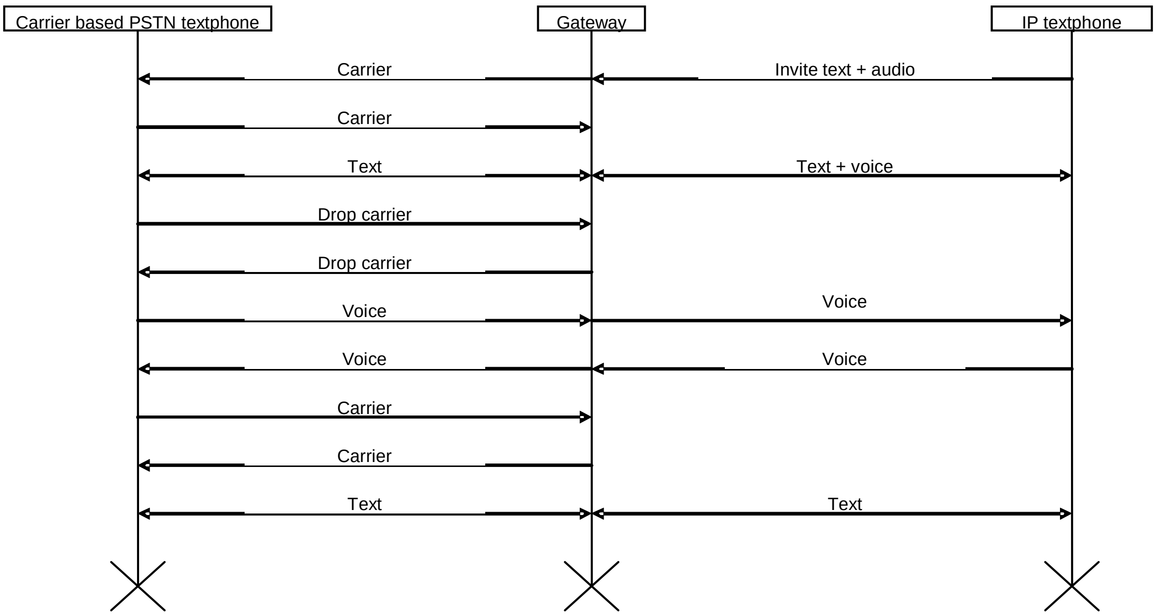


Figure 12: Alternating text and voice with carrier based textphone

13.2.6 Simultaneous text and voice through the gateway

If a protocol for simultaneous text and voice is supported in the PSTN termination by the gateway and such mode achieved, the voice and text channels are conveyed independently through the gateway. The reasoning about the need for a gateway buffer still applies because the two sides may have different transmission speeds.

13.2.7 Gateway invocation

Invoking text gateway functions requires specific planning. During migration, there will still be many VoIP gateways which do not properly support text. Calls from the SIP side may be directed to specific text capable gateways by routing based on e.g. call type indicator or preference registration. Calls from the PSTN side may use a special text service for a subscriber using text, thus permitting the call to be routed through a text capable gateway. Other approaches are possible.

Gateways can be deployed as part of an enterprise network solution, or provided by a service operator. They can also be residential gateways, allowing connection of PSTN devices to IP networks.

Figure 13 shows an example of a gateway function located in a business network providing support for both DUST and PSTN textphones for the office employees, having DUST terminals for their communication.

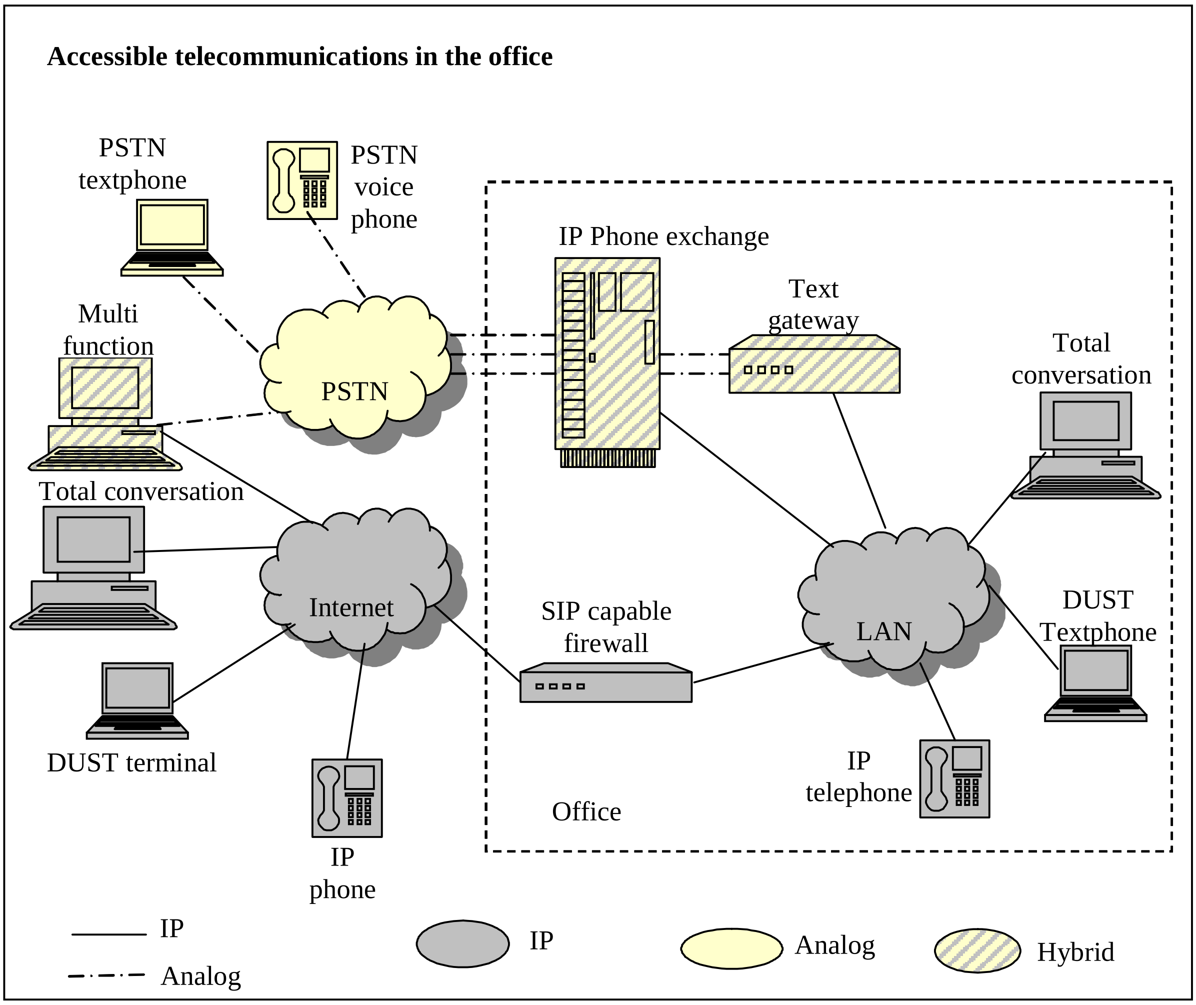


Figure 13: A business network solution

13.2.8 Invoking gateways to support DUST calls

Textphone users need simple methods to invoke text capable gateways in their calls. At best, any call between a text telephone and a text capable IP terminal should pass a text capable gateway without requiring any extra actions from the text telephone users. This may be difficult to achieve initially, when many gateways are already deployed which do not have specific text support. Therefore other approaches to getting text capable gateways into the call path must also be specified. National or European policy actions may be needed to encourage widespread implementation of text capable gateways.

Table 3 shows some possible text gateway invocation methods. There may be more methods with their variants.

Table 3: Possible invocation methods

Call origin	Method	Function	Comment
PSTN	All calls through text capable gateway	Gateways start the call in voice mode and monitor for text activity.	Easily achieved for enterprise networks. Can be deployed per operator and region in public networks.
PSTN	The IP user is a text subscriber	The E.164 number of the IP user is connected to a text subscription and causes routing to a text capable gateway.	The gateway can still be a general gateway allowing voice calls.
PSTN	Prefix on E.164 number	The prefix causes routing to a text capable gateway.	The gateway may be a text-only gateway. Voice calls to the same user may be made without a prefix.
PSTN	Two step dialling	The PSTN user dials a number to the gateway. The gateway performs a dialogue with the user to resolve the IP address.	Not inconvenient in its basic form.
IP	A call using tel uri is routed to text capable gateway	The IP user calls using a tel-uri type of address containing the telephone number of the PSTN text telephone. The address analysis in the SIP proxy invokes a text capable gateway for this IP user or for sessions where text medium capability is declared.	Situations for nomadic users are still to be resolved.
IP	Two step dialling	The IP user calls the gateway. The gateway performs a dialogue with the user to resolve the IP address.	Not convenient in its basic form.
IP	Explicit addressing of the text capable gateway as the domain for the call	The IP user addresses the PSTN terminal by PSTN-number@gateway-domain or some other form with the same result.	Requires extra knowledge from the IP user.

13.3 Transmission problems for legacy text telephony in new environments

IP transmission is becoming more common as a means to carry PSTN calls. The methods should not introduce any changes for the PSTN users, but it is evident that when this method was introduced, text telephony was not taken into full consideration. The quality of text telephony may deteriorate because of packet loss, jitter and voice coding distorts the modulated signals between the textphones. This is described in clause A.4.

There are two similar situations which exhibit this problem.

- IP transit between PSTN segments.
- Telephony provided through a residential gateway or an analogue adapter.

These problems should be taken into account in network planning, and in the planning of equipment to support PSTN devices.

There are recently approved standards for improving this situation. There are also further standards in development.

These standards should be applied to overcome the observed problems.

The standards are:

- ITU-T Recommendation H.248.2 [53], for gateways between PSTN text telephony and IP text conversation, to be used for transit traffic or for traffic with IP endpoints.
- ITU-T Recommendation V.152 [70], for transit gateways, invoking reliable audio transmission of textphone tones when textphone signals are detected.

The drafts in creation are:

- draft-ietf-sinnreich-sipdev-req-05.txt. A general SIP device requirement specification that also requires analogue adapters to have support for text telephony and text coded transport in SIP.
- ITU-T Recommendation V.151 (in preparation). Gateway procedures for text relay for transit traffic. This work has the ambition to create a Recommendation for a transit gateway procedure that transmits text packets, and enables text telephone protocol conversion between the different PSTN textphone protocols.

14 Requirements for DUST verified against technical facilities

The requirements from clause 6 can be met in a number of ways in practical DUST implementations. Table 4 identifies for each requirement an analysis of the viability of meeting the requirement in existing systems or protocols and as a consequence an analysis of where further standardization work may be required.

NOTE: A service provider may choose to implement only a subset of the DUST requirements but may still be in a position to offer a commercially viable service, e.g. a service that has no guaranteed quality of service.

Table 4: Verification of DUST requirements

<table>
<tr><th>Requirement</th><th>User statement</th><th>Technical statement</th><th>Point of technical application</th></tr>
<tr><td rowspan="2">1</td><td>Text communication should be available on all networks on a universal basis</td><td>Global Standards that are compatible with international transmission networks are needed to ensure interconnection and interoperability of conversational text communication terminals</td><td>Terminal
Network</td></tr>
<tr><td colspan="3">NGN has the ambition to enable provision of its services globally and throughout various network technologies. By making DUST part of NGN the best opportunities are established for global interoperability of DUST services. Pre-NGN, SIP provides an environment for DUST that enable global interoperability within the Internet, and it is extendable to other networks by gateways. See clauses 9.6, 13 and annex C.</td></tr>
<tr><td rowspan="2">2</td><td>Text communication should be as easily set up as a normal speech call.</td><td>DUST services should be addressable within the international telecommunications numbering space.
The text component of a call should be enabled automatically or by a very simple user action</td><td>Network
Terminal</td></tr>
<tr><td colspan="3">In an environment where SIP is deployed as the call setup mechanism, a DUST service call setup is made as easily as any SIP call and from a user's perspective will be indistinguishable from a speech telephony call. SIP has a number of addressing possibilities, with E.164 [45] numbers being one of them. The ENUM service can be used for resolution of E.164 [45] number addressing, into SIP URI addressing.
Negotiating the text medium is part of the media negotiation of SIP. Declaring text capability should normally be enabled, so that the text channel is made available in all calls where both terminals are text capable.</td></tr>
<tr><td rowspan="2">3</td><td>Live two way conversational text communication should be available to all users with delays that do not impair normal interactive conversational flow.</td><td>The delay should be limited in accordance with ITU-T Recommendation F.700 [46].</td><td>Terminal
Network
Management</td></tr>
<tr><td colspan="3">This requirement is documented in ITU-T Recommendations F.700 [46] and F.703 [47]. Implementations may use ITU-T Recommendation T.140 [64] and RFC 4103 [42] as these were written to meet F.700/F.703 requirements. Clause 11 and annex C.2 of the present document describe how the requirements are met.</td></tr>
<tr><td rowspan="2">4</td><td>Error free live conversational text communication should be available to all users.</td><td>To prevent corruption of the text display, the character corruption rate should be limited in accordance with ITU-T Recommendation F.700 [46].
The text transmission should permit a rate of at least 30 characters per second (measured over one minute).</td><td>Management
Transport
Terminal</td></tr>
<tr><td colspan="3">This requirement is documented in ITU-T Recommendations F.700 [46] and F.703 [47]. Implementation in RFC 4103 [42] meets the requirements specified in F.700/F.703 as RFC 4103 [42] specifies redundant transmission as a general tool for assuring proper quality. Annex C describe further how the requirements are met.</td></tr>
<tr><td rowspan="2">5</td><td>Speech communication of good quality should be available simultaneously with the text communication.</td><td>The speech coding of the terminal should comply with that specified by the service provider.</td><td>Terminal
Network
Gateway</td></tr>
<tr><td colspan="3">This requirement is provided by meeting the requirements for speech quality set out in the relevant ETSI documents.</td></tr>
<tr><td rowspan="2">6</td><td>Terminals should be able to display the text of both parties in the character set in which they are typed.</td><td>Guidance can be found in ETR 333 [7] and ITU-T Recommendation T.140 [64]. Terminals should provide support for all characters in ISO/IEC 10646-1 [43].</td><td>Terminal</td></tr>
<tr><td colspan="3">By using ITU-T Recommendation T.140 [64] for coding and presentation of the text dialogue it is possible to provide international support in a consistent way. ITU-T Recommendation T.140 [64] is based on the International character set ISO/IEC 10646-1 [43] as described in clause 11.</td></tr>
</table>

<table>
<tr><th>Requirement</th><th>User statement</th><th>Technical statement</th><th>Point of technical application</th></tr>
<tr><td rowspan="2">7</td><td>Missing text should be detected and an indication should be given in the display</td><td>Missing text should be detected. Missing characters should be indicated using the replacement character 0xFFFD of ISO/IEC 10646-1 [43]</td><td>Terminal
Transmission protocol
Gateway</td></tr>
<tr><td colspan="3">The presentation protocol ITU-T Recommendation T.140 [64] and transport protocol RFC 4103 [42] define methods for detection and marking residual loss after efforts to recover from transmissions errors. This is described in clause 11 and annex C.2</td></tr>
<tr><td rowspan="2">8</td><td>Simple editing functions should be provided</td><td>Facilities should be provided to erase the last character and to insert a new line</td><td>Terminal
Gateway</td></tr>
<tr><td colspan="3">This requirement may be met if implementing the presentation protocol ITU-T Recommendation T.140 [64] described in clause 11, which defines some simple editing found to be desirable by users of real time conversation.</td></tr>
<tr><td rowspan="2">9</td><td>All services should operate with text in addition to other media</td><td>All services should be provided in a form compatible with the text medium.</td><td>Terminal
Gateway
Network</td></tr>
<tr><td colspan="3">For basic and supplementary services this requirement means that DUST must operate with the text medium and without reliance on audio. The benefit of a DUST host environment deploying SIP, is that SIP is media independent. Therefore the opportunities are good for fulfilling this requirement, but conscious action is required for each service to make sure that they operate with text as well as audio.
The requirement to use text in service invocation and progress is best fulfilled by assigning standardized text information for all service progress indication messages. In the example where SIP is used, service progress is reported with reason codes that can be translated to text messages. Further work is needed here to make these messages formulated properly for users of various languages.
One important conclusion of this requirement is that Emergency services must be provided in text and voice where users invoke the DUST service. This requirement must be carried on in the European project EMTEL and other initiatives for implementing emergency services. Some conclusions of this requirement are described in clause 12.1.</td></tr>
<tr><td rowspan="2">10</td><td>Alternative modes of communication should be available for all of the call progress information that is normally provided in audio form in both basic and supplementary services to make it accessible to text users</td><td>Guidance should be provided on the provision of call progress information in alternate forms.</td><td>Terminal
Network</td></tr>
<tr><td colspan="3">In the example where SIP is deployed, call progress is reported with reason codes that can be translated to explaining text messages. Further work is needed here to make these messages formulated properly for users of various languages.</td></tr>
<tr><td rowspan="2">11</td><td>Conferencing services should support DUST capable terminals</td><td>Standards should be identified for the support of multi-party communication using text</td><td>UNI
NNI</td></tr>
<tr><td colspan="3">Display of the text medium from a conference participant should be done similarly to how video is handled, not mixed, but displayed under a label indicating the source.
Mechanisms for provision of conference services including real time text is described in ITU-T Recommendation H.248.19 [75] and in the XCON group of IETF.</td></tr>
<tr><td rowspan="2">12</td><td>A DUST compliant service should offer service in all modes within the capabilities of the terminals and networks engaged on a call.</td><td>Standard protocols are needed for the protocols at the start of a communication session to ensure support of text and other media</td><td>UNI
NNI</td></tr>
<tr><td colspan="3">The text component can be added to a general call environment that provides audio and video as well. This is shown in clause 9. In the host environments deploying SIP each medium is separately declared and established so there are good opportunities to get all common media connected in a multimedia call.</td></tr>
<tr><td rowspan="2">13</td><td>Terminals supporting the DUST service capabilities should be configurable by the user to suit the preferences and abilities of the user.</td><td>Guidance should be provided on the likely preferences and abilities that need to be catered for when configuring a terminal</td><td>Terminal</td></tr>
<tr><td colspan="3">Obvious terminal configuration features are text colour, background colour, text size, text window size. Advice on such factors are found in EG 202 116 [3]</td></tr>
<tr><td rowspan="2">14</td><td>Any service offering a video mode should provide a video display with a quality that is sufficient for signing and lipreading.</td><td>Any service offering a video mode should provide a display at least in accordance with ITU-T H-series Supplement 1 [44].</td><td>Terminal
Network</td></tr>
<tr><td colspan="3">ITU-T H-series Supplement 1 [44] provides a test sequence that can be used by implementers to verify that video transmission implementations comply with the requirements.</td></tr>
</table>

Requirement	User statement	Technical statement	Point of technical application
15	A DUST compliant service should be configurable by the user to suit the preferences and abilities of the user	Guidance should be provided on the likely preferences and abilities that need to be catered for when configuring a service	Terminal Service registrar UNI
	RFC 3351 [32] gives guidance on service invocation based on user preferences and terminal capabilities. Its use for relay service invocation is mentioned in clause 12.2.		
16	A user of the DUST service should be provided with the means to alter the terminal or call configuration during a call.	Standards need to be identified and implemented for those signalling protocols required during a communication session.	Terminal UNI
	Media may be added and subtracted during the call, by means of the call control protocol specified in annex B.		
17	A capability should be provided within the network to enable communication between users of terminals that do not share common modes of communication	A formal standard is needed to implement the recommendations given in TR 101 806 [10]	NNI Relay service provision
	Relay service function and invocation is described in clause 12.2		
18	Means should be provided to minimize the possibility of masquerade	Facilities should be provided for the verification, validation and authentication of the identity of a caller	UNI NNI E2E Terminal
	When using SIP as the host environment for DUST the security measures associated with SIP are able to provide protection from masquerade attacks in combination with appropriate identity and attribute management..		
19	The integrity of all communication should be maintained	A set of standards are needed to ensure that the integrity can be checked across a variety of networks taking account of malicious and non-malicious alteration of content	UNI NNI E2E Terminal
	When using SIP as the host environment for DUST the security measures associated with SIP are able to provide some integrity protection in combination with appropriate management of attributes which may include cryptographic keys..		
20	All communication should be treated as confidential	A set of standards are needed to ensure that the confidentiality can be maintained across a variety of networks taking account of malicious eavesdropping	UNI NNI E2E Terminal
	When using SIP as the host environment for DUST the confidentiality of the text medium can be protected e.g. with SRTP as described in RFC 4103 [42].		

15 Outstanding issues

The present document identifies some actions that need to be carried out to enable a full implementation of a DUST service.

- Clause 13.3 identifies work in being carried out in the IETF and the ITU-T which is not yet complete.
- Project EMTEL will need to describe the necessary requirement for implementing text in emergency services as identified in table 4, clause 8.
- As noted in table 4, clause 9 work is needed in TC HF to generate guidance on the provision of call progress information in alternate forms. Further work will be needed to describe actual messages in the range of European languages. Subsequent work will be needed in NGN to define suitable implementations.
- As noted in table 4, clause 14 work is needed in TC HF to give guidance on likely user preferences that need to be catered for when configuring a DUST service.
- In clause A.2.10, the STF was unable to determine the types of textphone system used in certain of the new accession countries.
- Clause A.5.1 notes that conversion from CTM to V.18 is described in TS 126 226 [15]. Further work will be needed in 3GPP to define the V.18 performance requirements in a similar manner to which Baudot requirements are specified in TS 126 231 [17] in order to ensure interoperability between mobile text telephony and legacy PSTN terminals.
- NGN will need to be agree that the stage 1 and stage 2 definitions given in the present document should be adopted by the NGN and expanded to allow design at stage 3 for deployment in the NGN to be undertaken.
- Service Providers are encouraged to implement techniques that support and overcome NAT traversal and Firewall issues.

16 Summary and recommendations

The present document demonstrates that there are user requirements for real time text conversation to be included as a mainstream feature in conversational services. The need is for global interoperability on the same level as for voice telephony.

The present document describes a design for all solution providing real time conversational services that support text, speech and optionally video on current and future networks by applying standards for real time text conversation to telephony and video telephony, thus creating the DUST service.

Most telecom users will benefit from the DUST service, while it is essential for people who have limited or no use of voice telephony.

The present document also reports a study of the many different textphone systems currently in use and gives details of how they can migrate to the new service.

The following recommendations can be expressed for service implementors:

- Implement DUST services from the beginning in new service deployments.
- Provide means for the migration of textphone systems to the DUST service.

Annex A (informative): Text conversation - The existing situation

A.1 Text telephony

In the PSTN, seven different, openly specified systems for text telephony exist, and are used in different regions. The text feature is thus not harmonized or globally supported even though international standards exist. Some proprietary modes are also used.

The open specifications are Baudot, Dual Tone Multi Frequency (DTMF), the European Deaf Telephone (EDT), ITU-T Recommendation V.21 [66], Bell 103, Minitel and ITU-T Recommendation V.18 [65]. They all use different modem technologies and character coding for the transmission of text. They are all briefly described in the annexes of ITU-T Recommendation V.18 [65], and below.

In the GSM and 3GPP environments, the Cellular Text Modem (CTM) is used. CTM is specified in TS 122 226 [13], TS 123 226 [14] and TS 126 226 [15].

A.2 Current PSTN textphone systems

A.2.1 Overview

There are still a number of different textphone systems in use in Europe today although the situation has improved somewhat since the COST 219 (see Bibliography) and COST 220 (see Bibliography) reports in 1991 due to the implementation of ITU-T Recommendation V.18 [65] modems in some terminals and relay services. Some of the systems described in the COST reports have now fallen into disuse. This annex gives an overview of the openly specified systems in use in the PSTN, both in Europe and globally. There are also a few proprietary systems in use.

These current systems cannot in general provide for user Requirements in clause 6.1 numbered 1, 3, 6, 8, 9, 10, 11, 15 and 17.

A.2.2 DTMF

DTMF based text telephony is used in the Netherlands. It uses combinations of the touch-tones as characters using codes as defined in the CEPT standard T/CS 34-15 [2] and in ITU-T Recommendation V.18 [65], annex B. The method supports one way at a time alternating transmission (simplex mode), and reaches about 4 characters per second.

A.2.3 EDT

The European Deaf Telephone (EDT) is used at least in Austria, Italy, Germany, Malta, Spain and Switzerland. It is based on the ITU-T Recommendation V.21 [66] modem transmission used in simplex mode at 110 bits/s. It reaches around 10 characters per second. EDT communication is specified in ITU-T Recommendation V.18 [65], annex C.

A.2.4 V.21

The ITU-T Recommendation V.21 [66] text telephone system is used in Norway, Sweden, Finland, Denmark, Ireland, the UK and the Czech Republic.

It is based on using ITU-T Recommendation V.21 [66] modems at 300 bits/s with two way simultaneous transmission (duplex mode). Most V.21 textphones merge the transmitted and received text in one user interface area causing a need for turn-taking indicators in the text. Seven Bit ITU-T Recommendation T.50 [60] characters are used and around 30 characters per second can be transmitted. V.21 based text telephony communication is specified in ITU-T Recommendation V.18 [65], annex F.

A.2.5 Minitel

France and Belgium use a version of the videotex terminal Minitel which follows the CEPT 2 standard with additions for text telephony. Minitel based text telephony communication is based on the ITU-T Recommendation V.23 [67] modem, used at 1 200 bits/s in the forward direction and 75 bits/s in the backward direction, resulting in a possible performance of 120 characters/second and 7 characters/second respectively.

Although on the modem level, the transmission allows two way simultaneous transmission, the presentation level permits only simplex mode. The basics of Minitel text telephone communication are specified in ITU-T Recommendation V.18 [65], annex E, while many details on the presentation level are documented only in the Minitel terminal specifications.

A.2.6 EIA-825 ("Baudot")

EIA-825 is the modem standard for a text telephone type in use in USA, Canada, Australia and New Zealand. It is defined in ITU-T Recommendation V.18 [65], annex A under the name "5-bit". The modem is an FSK modem, used at 45,45 bits/s in USA and Canada, and at 50 bits/s in Australia and New Zealand. The character code is a 5-bit code, requiring a figure shift and a letter shift to present a suitable total character set. Transmission is simplex at a rate up to 6 characters per second. The letter character set is limited to upper case.

A.2.7 Bell 103 ("ASCII")

In the USA, there are computer users of text telephony using regular modems. They use the ASCII character set and initially the regional standard modem Bell 103. This is why this form of text telephony is commonly called "ASCII". The characteristics are quite similar to V.21.

A.2.8 V.18

In order to offer a tool for harmonization of the fragmented world of text telephony in PSTN, automoding procedures have been defined and standardized in ITU-T Recommendation V.18. [65].

Native modulation for V.18 is V.21, which enables alternating text and voice. The use of simultaneous voice and text as specified in V.61 [68], is a desirable improvement. At present only one product is known to use it.

V.18 and T.140 [64], the text presentation coding, are intended for use in new PSTN text telephones. They are also intended for use in gateways, providing a bridge from the fragmented situation in the PSTN into new services.

V.18 is specified to include automatic detection and support of Baudot, DTMF, EDT, Bell 103, Minitel and V.21, and V.18 itself.

V.18 is implemented in various products. In UK, there is an overlay network called TextDirect which is based on V.18. It is invoked from the calling party by dialling a prefix. There are terminals and network components on the market with V.18 implementation and where they are used this gives the big advantage that the user does not need to bother about what terminal type the other party in the call has.

Even though there has been a lot of discussion about the need for harmonization of text telephony, the market has not taken up the V.18 solution to the expected extent.

A.2.9 Other PSTN text telephone systems

There are other text telephone systems than the ones described above, among them are non-standard higher speed derivatives of the dominating US textphone type Baudot. They have become popular in USA because of the limitations in functionality of the Baudot textphones.

Users of such systems will either experience problems (when the handshaking for the proprietary method happens to succeed, but the transmission fails), or will experience a lower service level in terms of slower text transmission than they are used to (when the handshaking for the proprietary method fails, and transmission is reduced to be in accordance with the underlying standard).

Because these products are made to proprietary specifications, the standards cannot take them into consideration. Compatibility based on standards can be arranged only with the textphone types which follow an open standard.

A.2.10 Summary of existing PSTN textphone systems

Table A.1 gives a summary of the current PSTN textphone systems in use in Europe.

Table A.1: Current PSTN textphone systems

EU countries								
Country	**Type of textphone system**						**PSTN Text relay**	**PSTN text emergency**
	DTMF	**EDT**	**V.21**	**Minitel**	**Baudot 45**	**V.18**	**o = public, p = private**	
Austria		X						
Belgium				X			p	
Cyprus								
Czech Republic			X					
Denmark			X				o	X
Estonia		X						
Germany		X						
Greece		X						
Finland			X				o	X
France				X				
Hungary								
Ireland			X				o	X
Italy		X						
Latvia								
Lithuania			X					
Luxembourg								
Malta		X					o	
Poland		X						
Portugal					X		o	
Slovakia								
Slovenia								
Spain		X				X		
Sweden			X			X	o	X
The Netherlands	X					X	o	
United Kingdom			X		X	X	o	X

Other countries								
Country	**Type of textphone system**						**PSTN Text relay**	**PSTN text emergency**
	DTMF	**EDT**	**V.21**	**Minitel**	**Baudot 45**	**V.18**	**o=public, p=private**	
Norway			X				o	X
Switzerland		X					o	
Iceland			X				o	X
Romania	None used							
Turkey								
America					X		X	X
Australia					Baudot 50		X	X

A.3 Establishing a call

Current text telephony calls between two terminals can be set up over switched networks such as the PSTN or the ISDN.

A.3.1 Terminal to terminal over a PSTN connection

Current PSTN text telephone systems use the PSTN as their host environment. Call connection is made by using a voice telephone, or with the same operations as a voice telephone.

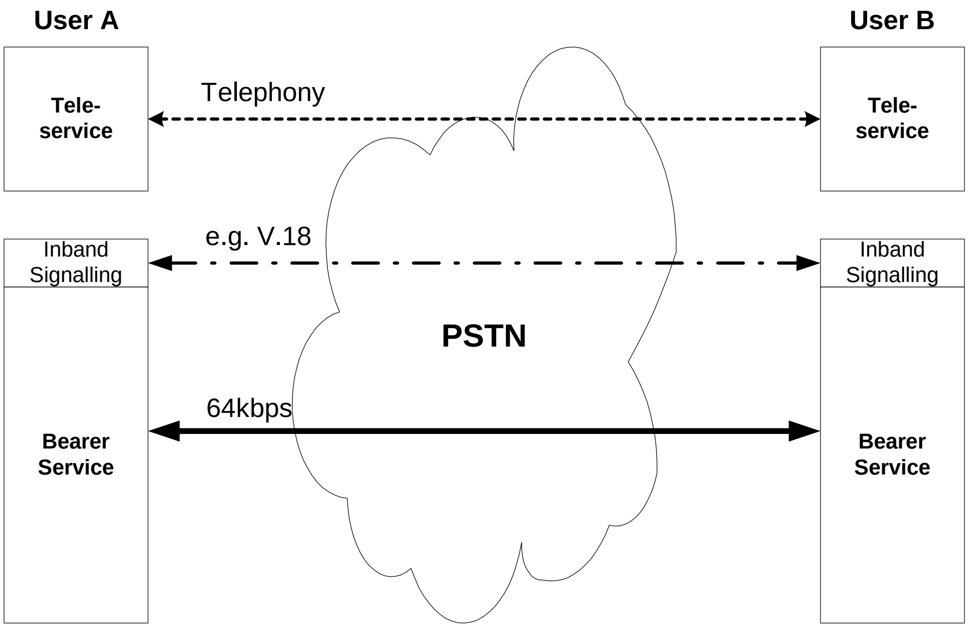


Figure A.1: Current Text Telephone implementation

Figure A.1 illustrates the separation of the bearer and establishment of the Telephony teleservice within the Analogue PSTN environment. Figure A.2 illustrates a basic message sequence for the setting up of the bearer connection, followed by in-band signalling of the text telephone device to establish communication between the end-terminals.

The text function starts when the user requests it from the terminal. It is important to note that there are two classes of PSTN text telephone transmission methods, the "carrier-based" and the "carrier-less" terminal types.

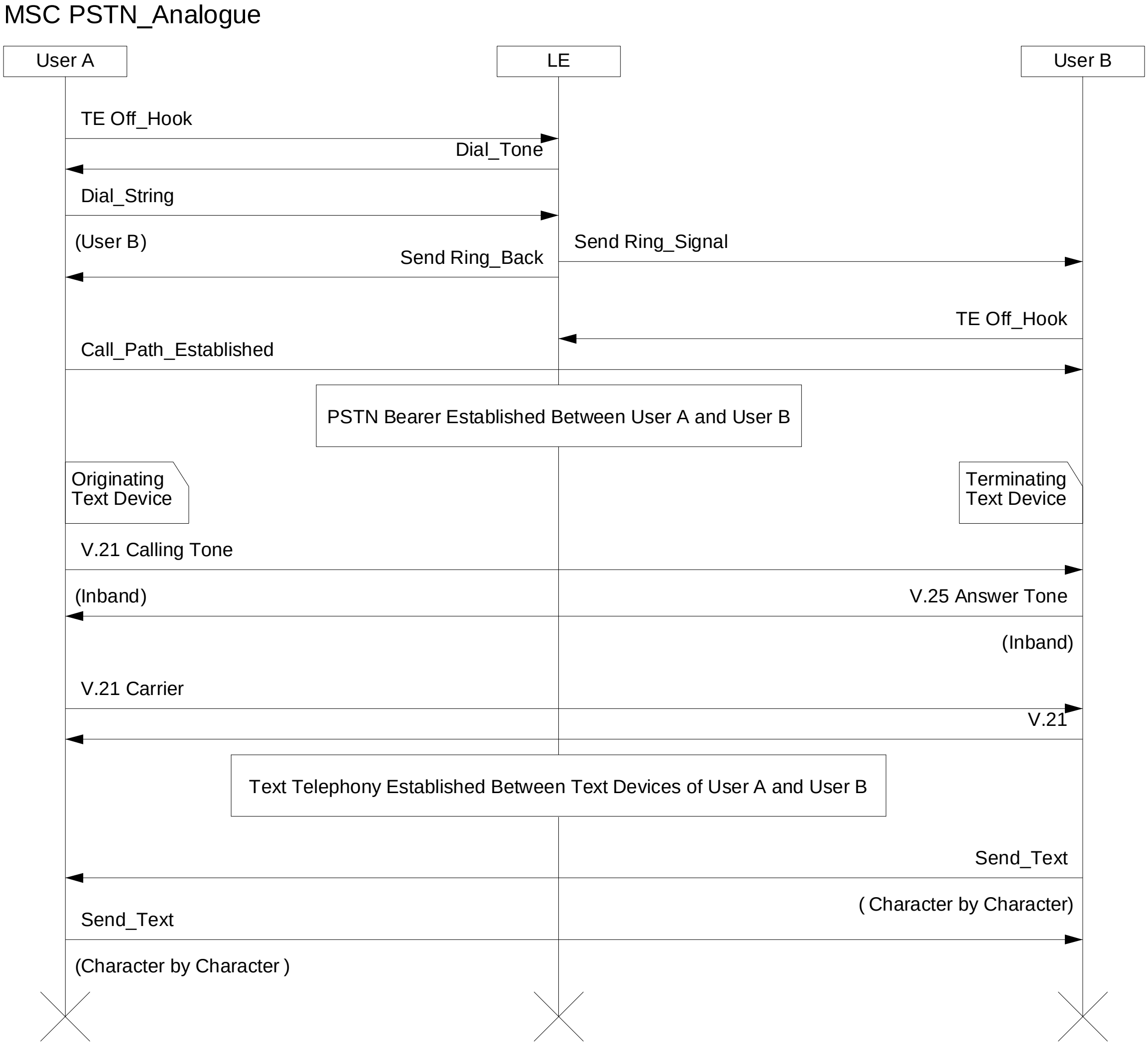


Figure A.2: PSTN Carrier Text Telephone Call Establishment

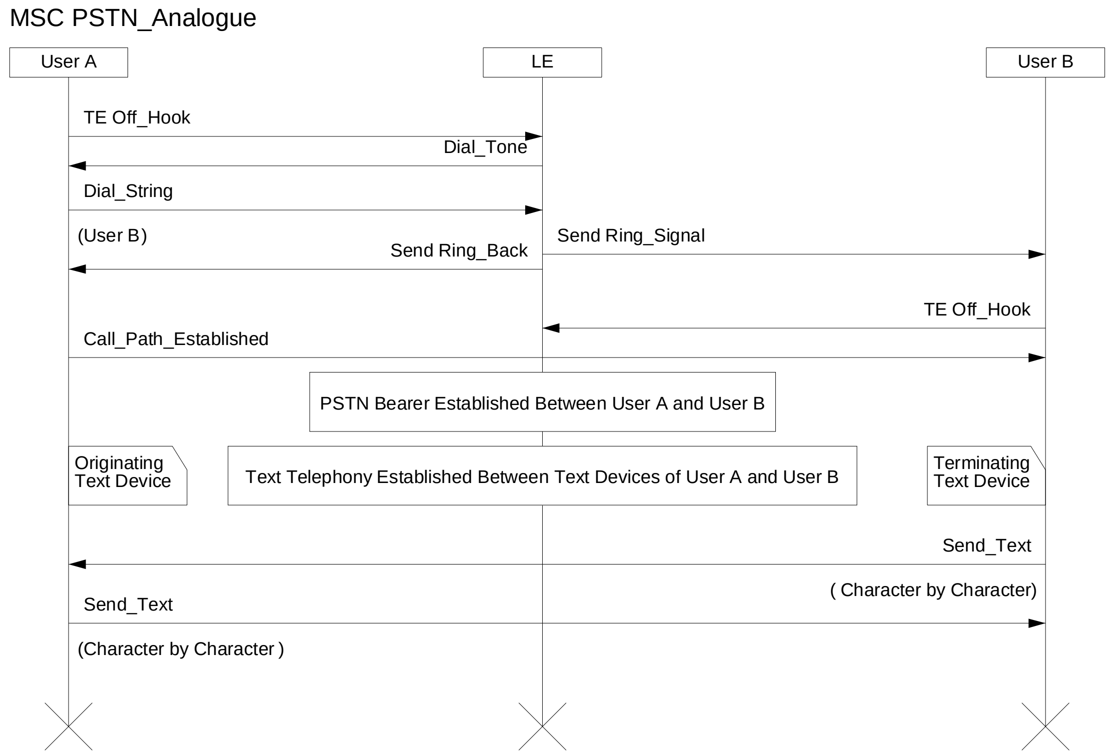


Figure A.3: PSTN Carrier-less Text Telephone Call Establishment

The carrier-based terminals utilize ITU-T Recommendation V.21 [66], ITU-T Recommendation V.18 [65], Minitel and Bell 103 modulation. Terminals utilizing EDT, DTMF, Baudot and CTM are Carrier-less.

In figure A.2 it can be observed that carrier-based terminal types initiate the in-band communication by sending a carrier tone and expecting another carrier tone back. Once the originating terminal has detected the carrier sent by the terminating text device, text telephony communication can then proceed. Text is sent character by character as it is typed, and modulates the carrier according to the modem modulation used. If the users occasionally want to talk and listen, the carrier is temporarily turned off, and the voice telephone is used for speech.

In the case of carrier-less type terminals, there is no exchange of carrier tones to identify the terminals as being text telephony devices until the user has pressed alphanumeric keys, as shown in the message sequence of figure A.3. The user's terminal will transmit the typed alphanumeric characters according to the modulation specified for that type of terminal. When typing ceases, transmission ceases. For EDT, Baudot and CTM, short gaps between characters are filled with continuous tones.

Speech is possible at any time when the terminals are not transmitting.

ITU-T Recommendation V.18 [65] is special in that it is compatible with all the other standard text telephone methods. Therefore, the handshaking determining the mode to work in contains more signals than just a carrier, and the resulting communication can end up in any carrier-based or carrier-less mode that suits the other party in the call. V.18 terminals calling other V.18 terminals use carrier-based ITU-T Recommendation V.21 [66] modulation as the default modulation method.

A.3.2 Terminal to terminal over an ISDN connection

The telephony and 3,1 kHz audio teleservices are used for text telephony utilizing the same modem based communication as in the PSTN.

Another mode of text conversation is also defined in ITU-T Recommendation H.320 [55] where n x 64 kbit/s unrestricted bearer services are used.

In the ISDN the call set up process is different from that of the PSTN. As shown in figure A.4, unlike the call set-up process within the analogue PSTN, the end-user Terminal Equipment (TE) will first request the type of connection required. The set-up information will include a number of teleservice and bearer service attributes specific for the type of call being placed.

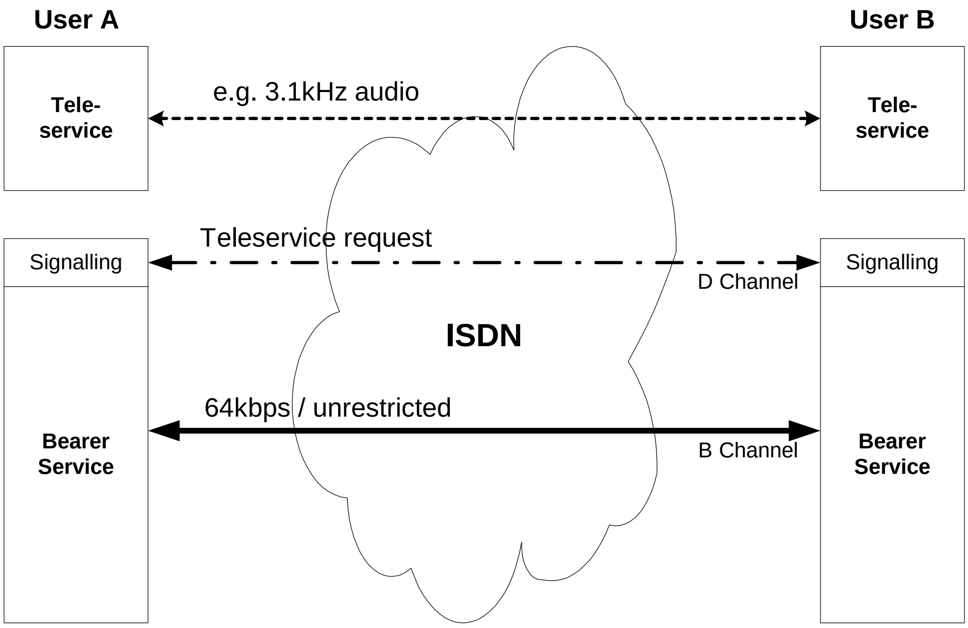


Figure A.4: ISDN teleservice activation

All signalling information is passed within the 16 kbps D Channel, including such information as the dialled digits (E.164 [45] number) of the end user (User B). Only when the ISDN TE of User B has acknowledged that it has the same teleservice capabilities will the network allocate the 64 kbps B-Channel for end-to-end communications.

A.3.3 Terminal to terminal over a GSM connection

GSM is another form of switched network over which a peer to peer connection may be set up. Figure A.5 provides an illustration of the way in which a GSM terminal sets-up a call and this may be used for generic comparison to the call set-up in the PSTN (see figure A.1).

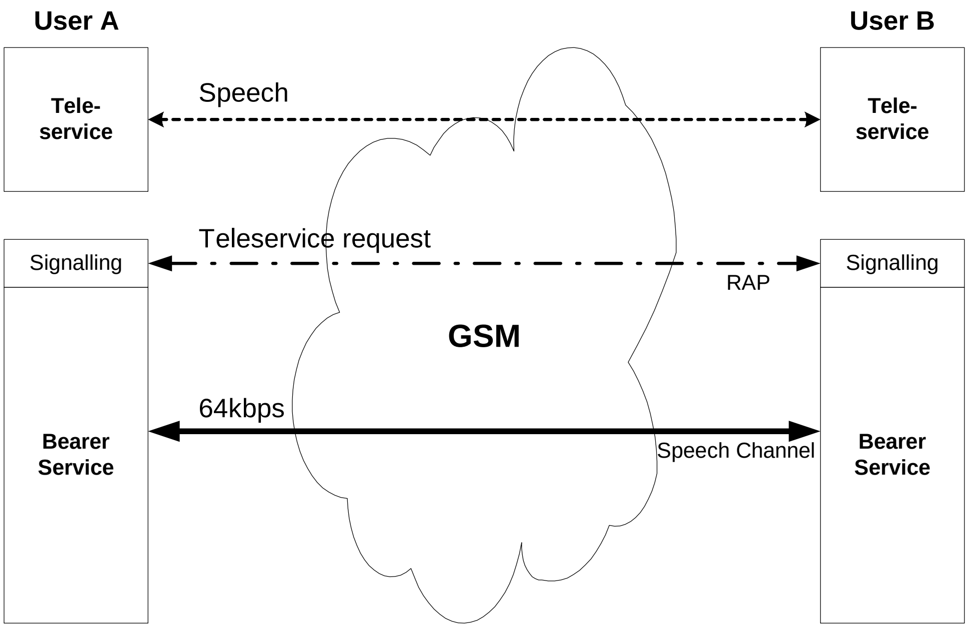


Figure A.5: GSM teleservice activation

The GSM network contains speech encoders which can corrupt the text transmission at the network gateways as described in clause 13.3. This problem can be avoided by using the robust CTM modem (TS 126 226 [15]) which was specifically defined for this purpose.

A.3.4 Terminal to terminal over packet networks

A.3.4.1 General packet network

In a packetized network, a connection may be set up between items of ITU-T Recommendation H.323 [57] equipments using the call signalling protocols described in ITU-T Recommendation H.225.0 [51]. This uses an ITU-T Recommendation H.323 [57] gateway to which may be connected multimedia terminals such as those conforming to ITU-T Recommendations H.320 [55], H.324 [58] or H.310 [54] / H.321 [56].

A.3.4.2 IP Network

In an IP network, two terminals can be effectively connected together using packet switching and routing determined by Internet Protocol (IP). IP establishes the nature and length of the packets carried in a network and provides address information used by the switches and routers that direct each individual packet to their intended destination.

TCP/IP is a protocol suite that runs on top of IP and adds important functionality. It is connection-oriented in nature which does not set up a predetermined path for the call but sets up a virtual connection between two terminals. TCP structures the datastream from the source and at the receive end, checks for errors and resequences the packets to ensure that the information can be reconstituted in its original form.

A.3.4.3 Managed IP Network

Ordinary packet switched IP networks routes data packets on a "Best effort" basis. They were not designed to support real-time traffic such as voice. Public service providers offering IP telephony use a managed network that can be configured to give a better quality of service to support real-time voice traffic although some services exist which only offer "Best effort" over the Internet.

In a managed IP network calls can be set-up between terminals as shown in figure A.6 and this may be used for generic comparison to the call set-up processes in the PSTN (figure A.1), ISDN (figure A.4) and GSM (figure A.5).

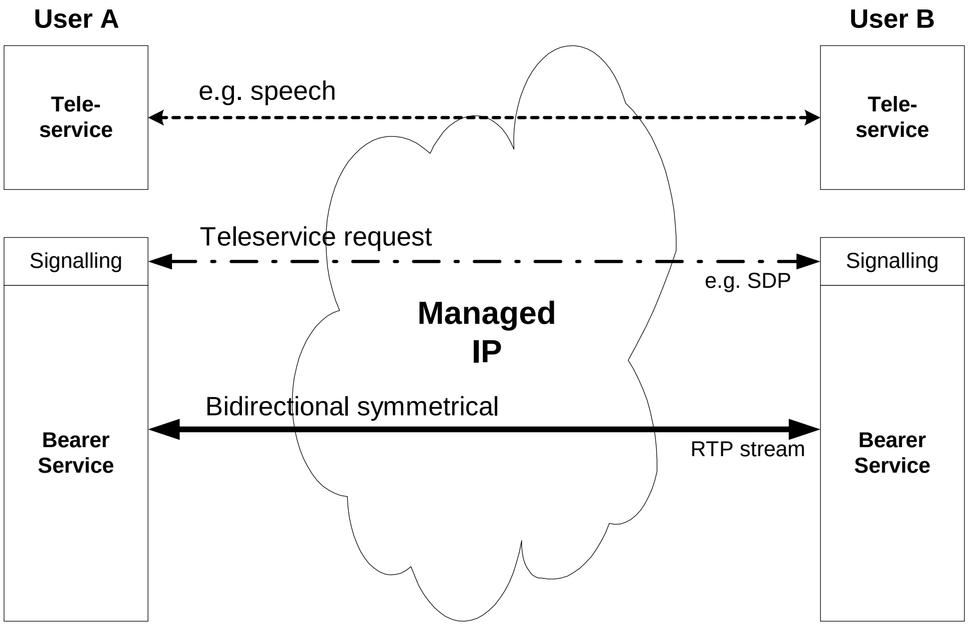


Figure A.6: Managed IP teleservice activation

A.3.5 Indirect access

A textphone call may be set up by indirect access. Such a service is provided in the UK by BT under the name of TextDirect™. This is a service that is accessed on a per call basis by prefixing the full national number with an access code which routes calls through the TextDirect overlay network.

The service provides communication between both compatible and incompatible text phones and gives all call progress announcements in text form. The Calling Line Identity (CLI) number of the calling line, or the reason for its absence is passed through the service.

Call status messages are delivered in a form appropriate to the customer at that moment. A textphone user will receive messages in text form compatible with their textphone and a voice user will receive messages as tones or voice.

The service uses ITU-T Recommendation V.18 [65] compliant modems and provides automatic access to a relay service for voice-to-textphone and textphone-to-voice calls. The service also supports both textphone-to-textphone and voice-to-voice calls. Users of textphones have to dial 18001 as a prefix to the number required, voice users wishing to call a textphone must use 18002 as a prefix. Public emergency call service can be accessed by dialling 18000.

For voice users the TextDirect access code is 18002 and is used when they wish to make a call which might be answered by a textphone use. When the call is answered the service will detect if a textphone is present and a relay operator will then be connected in the call.

A fuller description of the service can be found in the BT Suppliers Information Note SIN 359 [1].

A.4 System problems

There are a number of problems in existing system that have generally been caused by the lack of knowledge about current PSTN text telephone systems among mainstream telecom developer and also a lack of incentive to solve the problems.

When new methods of transmission are developed which use various methods of speech processing, the requirements of text telephony are often not initially considered. This creates the risk that text telephone calls will not get the expected quality of service because the processing causes some corruption of the text.

Problems have occurred when trying to use textphones on mobile services because some network gateways have been equipped with incompatible modems. A similar observed risk is caused by the move to IP transmission for PSTN calls. Even if the end users still have textphone equipment which is connected to the PSTN, their calls can be routed through IP transit connections where it can suffer both destructive compression coding and packet loss. Packet loss can easily cause gaps in the transmission that are impossible to cover and will cause loss of data. A packet loss rate of just a few percent can cause a many times higher rate of character loss.

Some broadband suppliers provide telephone services through a simulated analogue network connection using a suitable terminal adapter. There is a clear risk that any audio coded text may become seriously corrupted by compression coding and packet loss after transmission through a packetized IP network.

It is not well understood by network operators, terminal providers or users that textphone traffic can be seriously affected by low transmission quality. Although there is a standard for gateways that are capable of handling voice, text and fax traffic between PSTN and IP networks (ITU-T Recommendation H.248.2 [53]) it has largely been ignored, and many gateways have been installed in the past which do not provide this necessary support for textphone calls.

Work is currently being done on other standards in the ITU-T Recommendation V.150 [69] series which would enable modem, text, fax and voice to be transmitted successfully between pairs of gateways but the problem will remain unless the existing gateways are updated. So far, ITU-T Recommendation V.152 exists [70].

There is one property of legacy textphones that is causing problems in the design of gateways between PSTN and IP networks. Some of the legacy textphones start sending valid text only a few milliseconds after modem signals have begun without leaving any time for the necessary discrimination between text and voice signals. This problem can only be circumvented by proper design of the receiver/decoder for text. There will need to be a balance between the risk of falsely detecting text when there is none, and delaying the decision so that the first character is missed.

A.5 Mobile systems for interactive text conversation

In mobile telephone systems, there are a small number of standards for text telephony and a few proprietary systems in use.

A.5.1 Global Text Telephony in the GSM and 3GPP network

In GSM and Third Generation Partnership Project (3GPP) mobile networks there is a method for carrying text telephony over the circuit switched voice channel of the GSM and 3GPP networks using a Cellular Text telephone Modem (CTM) which transmits text coded according to ITU-T Recommendation T.140 [64].

CTM (TS 126 226 [15]), is a standard for a very robust modem which is an FSK modem with a large amount of redundancy in transmission (TS 126 231 [17]) that enables text conversation over a mobile circuit switched voice channel with about 8 characters per second. CTM can carry text over the voice channel even through low bit-rate voice coding and non-reliable transmission conditions. Conversion to fixed network text telephony as ITU-T Recommendation V.18 [65] or any of its legacy sub modes is described in TS 126 226 [15]. Further work may be needed to define the performance requirements when V.18 is used in the landline CTM conversion function.

CTM itself is internationally usable, and allows two way simultaneous transmission within a GSM or 3GPP mobile environment. The initial network implementations have only been made for the USA using TS 126 230 [16] where it is implemented with gateways to the US textphone system EIA-825 (Baudot). The solution allows simple alternation between text and voice mode.

Conversion between CTM and PSTN based text telephony can be done with a media conversion function which permits interworking between the mobile and fixed networks (TS 123 226 [14]). It is also possible to arrange the connection of a PSTN textphone to a mobile station using a similar conversion function. Other terminal configurations are possible, for example with the CTM encoded text being derived directly from a GSM enabled PDA.

A.5.2 Text telephony in the GSM data channel

Some GSM operators provide proprietary methods for real time text conversation over the circuit switched data channel of GSM. In most cases it has been implemented to the user as software in a communicator type of device. This method does not provide any voice support. PSTN textphone interoperability is provided through a gateway included in the service.

A.5.3 Text telephony using the GPRS IP access channel

In GSM networks there is an IP network access service called GPRS. Some manufacturers have established text telephony services based on this access. They provide gateways to PSTN text telephony. The GPRS text telephone services provide only text support and no voice.

A.5.4 Global Text Telephony in 3G IP multimedia services

The real time text component in IP multimedia calls in 3GPP systems is specified in TS 126 235 [18] which specifies the default codecs for multimedia applications. This sets the same specification as for SIP calls in other IP multimedia environments, using ITU-T Recommendation T.140 [64] for the presentation and RFC 4103 [42] with redundancy for the transport. This standard TS 126 235 [18] is valid from version 5 of the 3GPP system. It can be used to make both text telephony and multimedia conversations including text, video and voice.

A.6 IP based text

A.6.1 Chat systems

Text chat systems handle text messages consisting of blocks of text, usually forming a sentence or a larger amount of text. The "chat" is conducted on a turn taking basis. Therefore they do not meet the user requirement 3, requiring live two way transmission of text as it is entered. A description of some text chat systems are included here, because they have some similarities to real time text conversation, and can with some efforts and sacrifice be used as a surrogate for a real time text conversation feature.

A.6.1.1 IRC - Internet Relay Chat, WebChat

Internet Relay Chat (IRC), provides a text-based mechanism for communication with multiple participants. IRC is an interactive forum set up in virtual rooms that you can move between, and where others can virtually "hang out". Chat rooms can be used to discuss common ideas or topics, or as part of a collaborative process. The connection method used will be specific to each IRC site. The IETF has specified a number of elements of Internet Relay Chat systems including the Architecture (RFC 2810 [23]), Channel Management (RFC 2811 [24]), Client Protocol (RFC 2812 [25]) and Server Protocol (RFC 2813 [26]).

Web chat is similar to IRC, but it is achieved via a web browser, and it is not a text only forum.

Many webchat sites require the user to register before being able to participate in the activity. In some instances additional software may be needed dependent upon your particular software and PC configuration.

IRC and WebChat are not intended and generally not considered as meeting the requirement for real-time conversational text, and do not meet User Requirement 3 in clause 6. Neither IRC nor WebChat can be directly combined with real-time voice conversation and therefore does not meet User Requirement 11.

A.6.1.2 Instant Messaging

Instant Messaging (IM) differs from email primarily in that its primary focus is immediate end-user delivery. In order for instant messages to be passed from one user to another, both parties are required to be "online" and ready to communicate. Consequently some indication that a user is present within the system is required, this presence information has been implemented on internet-connected systems for many years. However, the computing infrastructure has become increasingly distributed and although a given user may be consistently available, there has been no standard way to make this information known to their peers. The IETF Instant Messaging and Presence Protocol (IMPP) working group are designing a system to address this need.

When the user launches their instant messaging application, it connects to the instant messaging server, logs the user on, and displays the screen names of people the user has added to their buddy list. A server checks the user's buddy list against the presence server information and then visually confirms those "buddies" that are available for messaging on the user's IM application. At the same time presence information is exchanged and updates the buddy lists of the user's buddies to indicate that the user is currently available for messaging.

With some instant messaging applications, it is possible to store a message on the instant messaging server temporarily if a buddy is unavailable, so that when the buddy does become available, the message will be sent. This facility is neither a message store nor an inbox in the context of store and forward services.

The IETF IMPP working group is developing the architecture for simple instant messaging and presence awareness/notification. It will specify how authentication, message integrity, encryption and access control are integrated. It is desirable, but not required, for the working group to develop a solution that works well for awareness of and communication with entities other than human users.

The IETF's Instant Messaging and Presence Protocol (impp) is defined in RFC 2778 [21] and RFC 2779 [22] and can be further defined in terms of a framework in the following elements:

1) "Common Presence and Instant Messaging: Message Format"
This memo defines the mime type "message/cpim", a message format for protocols that conform to the Common Profile for Instant Messaging (CPIM) specification.

2) "Presence Information Data Format (PIDF)"
This memo specifies the Common Profile for Presence (CPP) Presence Information Data Format (PIDF) as a common presence data format for CPP-compliant Presence protocols, and also defines a new media type "application/pidf+xml" to represent the XML MIME entity for PIDF.

3) "Address Resolution for Instant Messaging and Presence"
Presence and instant messaging are defined in RFC 2778 [21]. The Common Profiles for Presence and Instant Messaging define two URI schemes: "im" for INSTANT INBOXes and "pres" for PRESENTITIES. The address resolution document provides guidance for locating the resources associated with URIs that employ these schemes.

4) "Common Profile for Presence (CPP)"
Whilst Presence is defined in RFC 2778 [21], there are currently numerous presence protocols in use (largely as components of commercial instant messaging services), and little interoperability between services based on these protocols has been achieved. The CPP specification defines common semantics and data formats for presence to facilitate the creation of gateways between presence services.

5) "Common Profile for Instant Messaging (CPIM)"
Whilst Instant Messaging (IM) is defined in RFC 2778 [21], there currently exists numerous instant messaging protocols in use, with little interoperability between services based on these protocols having been achieved. The CPIM specification defines common semantics and data formats for instant messaging to facilitate the creation of gateways between instant messaging services.

The IETF SIP for Instant Messaging and Presence Leveraging Extensions (SIMPLE) Working Group focuses on the application of the Session Initiation Protocol (SIP, RFC 3261 [30]) to the suite of services collectively known as instant messaging and presence (IMP). The IETF has committed to producing an interoperable standard for these services compliant to the requirements for IM outlined in RFC 2779 [22] (including the security and privacy requirements there) and in the Common Presence and Instant Messaging (CPIM) specification, developed within the IMPP working group. As the most common services for which SIP is used share quite a bit in common with IMP, the adaptation of SIP to IMP seems a natural choice given the widespread support for (and relative maturity of) the SIP standard.
The primary work of this group will be to generate:

- A proposed standard SIP extension documenting the transport of Instant Messages in SIP, compliant to the requirements for IM outlined in RFC 2779 [22], CPIM and in BCP 41 (so that the transport implications of the extension with respect to network congestion are considered in the design).

- A proposed standard SIP event package and any related protocol mechanisms used to support presence, compliant to the requirements for presence outlined in RFC 2779 [22] and CPIM.

- An architecture for the implementation of a traditional buddy list-based instant messaging and presence application with SIP.

Included might be new mechanisms for message confirmation delivery, indications for when a party is in the process of typing a message, secure buddy list manipulation operations, and the extension of the CPIM presence format to describe typical IM states. Each of these mechanisms will be consistent with a SIP-based architecture.

The working group will work within the framework for presence and IM described in RFC 2778 [21]. It will define the extensions to SIP, but will not modify the baseline SIP behaviour or define a new version of SIP for IM and presence.

Instant Messaging does not include any real-time conversational text mode that would meet User Requirement 3. The opportunity to introduce a real time mode was discussed in the SIMPLE group of IETF during the summer of 2004 where it was decided to request that such a mode should be avoided and that real time text conversation with RTP as defined in RFC 4103 [42] should be used instead.

A.6.2 SIP standard based text conversation

There are so far only a few users of text conversation based on the standards for IP based total conversation that are currently used for real time interactive text conversation in some software-based products. The environment is SIP with text according to ITU-T Recommendation T.140 [64] transported as specified in RFC 4103 [42].

A.6.3 H.323 standard based text conversation

Some software-based products use the standards for IP based total conversation for real time interactive text conversation. The environment is ITU-T Recommendation H.323 [57] with text according to ITU-T Recommendation H.323 [57], annex G with ITU-T Recommendation T.140 [64] transported as specified in RFC 4103 [42].

A.6.4 Proprietary text conversation

A number of proprietary IP based mechanisms for text telephony exist, providing interoperability with legacy sub-modes of ITU-T Recommendation V.18 [65] in PSTN. They are mainly used as office textphone systems, in most cases only providing the text mode and offer no combination with speech.

A.7 Speech and text

The only widespread method for combining voice and text during a call in PSTN text telephone services is by alternating between the speech telephone mode and the text telephone mode, which take turns on being connected to the line. This method requires manual switching for the change to take place in both ends of the call, and is therefore quite inconvenient, and includes the obvious risk of the users getting out of phase with their modes. Simultaneous voice and text can be achieved using V.18 modems with V.61 modulation.

IP based systems can allow simultaneous voice and text.

A.8 Total Conversation

Total Conversation can provide simultaneous communication in video, text and voice by adding text conversation to multimedia protocols in a standardized way. For the text part, the Unicode based text presentation protocol ITU-T Recommendation T.140 [64] is used. It is transmitted with a specific transport mechanism standardized for each environment. Subsets can be used for example for text telephony enabled in the multimedia architectures. A Total Conversation service is defined in general terms in ITU-T Recommendation F.703 [47].

A.9 Special services

A.9.1 Emergency services

The Universal Service Directive [73] requires that all users are able to call the emergency service free of charge by using the single emergency call number "112". As the Directive refers to all users this implies that text emergency calls should use the same number or address as other emergency calls. Currently that is not the case in many European countries. Some do not have text support for emergency calls, some recommend calling through the regular text relay service, some have special emergency numbers for text access to the relay service and some have special emergency numbers for text calls which are directed to text terminals in emergency centres.

It is an important requirement that everybody should be able to make emergency calls with the communications equipment they use for day to day real time communication. For text users this requires text telephony access to emergency service.

When using the same number for text and speech emergency calls, it is necessary for the emergency service to be able to distinguish between text and voice calls. It is preferable that a text terminals sends a machine detectable signal indicating that it is a text call, so that the emergency centre system can act properly and ensure that the call is answered in text mode. If the signal is not detected automatically an emergency operator may receive a dangerous tone through the headset. Any V.18 or V.21 textphone sends such a tone as described in clause A.3.1 and illustrated in figure A.2. Text telephone systems such as EDT and DTMF (see clause A.3.1) do not.

When an emergency service is based on IP technology, it is important that calls with DUST protocols can be connected directly with the emergency centre, without converting to any PSTN service. In this way, the better functionality available to a DUST compliant service is maintained in calls with the emergency service, e.g. using text and voice simultaneously.

A.9.2 Relay services

Where the same communication mode or text protocol is not available at each end of a call, a relay service is needed to convert between the two modes in order to allow a conversation to be carried on between a user in one mode with a user in a different mode, (Requirement 16 of clause 6). TR 101 806 [10] gives guidance on the requirements for the provision of relay services in all telecommunications networks and sets out the minimum facilities that should be provided on a basic text relay service. It advises that such relay services should also provide a speech path in both directions so as to permit voice connection in either direction when desired.

The report also describes spoken to spoken relay services for users with impaired speech and a basic videophone service for signing users. It further sets out recommendations for call handling processes and system requirements, together with recommendations for traffic standards and quality.

The report sets out general requirements applicable to relay services accessed by all switched networks but does not deal with new requirements for access to text, voice or video over IP or any special needs which may arise from any requirement to handle SMS messages.

With IP based communication, there are increased opportunities to provide enhanced relay services. The following features are important enhancements possible when moving to IP environments.

- Simultaneous use of voice and text.
- One step dialling to the destination and invoking relay service in the same operation.
- Diversion of incoming calls on user demand to go through a relay service if the user is not capable of handling the preferred medium in the call.
- Arrangement for three party calls with all media available to all parties and the relay service being one of the participants.

Annex B (informative): Stage 2 description of DUST (simplified)

B.1 Introduction

ITU-T Recommendation I.130 [59] defines a staged method of development of protocols as follows:

- Stage 1: Specify requirements from the user's perspective.
- Stage 2: Develop a logical model to meet those requirements.
- Stage 3: Develop a detailed specification of the protocol.

The stage 1 definition of DUST is given in clause 7 of the present document. In the stage 2 development a model is specified based on logical blocks so that the flow of information necessary for meeting the specified requirements can be defined without concern for the detailed format that such information should take. The identification of possible normative interfaces between blocks is also simpler without the constraints imposed by a specific physical architecture. One advantage of making such an unconstrained design is that the detail protocol developed at stage 3 can comply to the requirements of stage 2 in a number of different ways. A number of considerations for the stage 3 design are given in annex C of the present document.

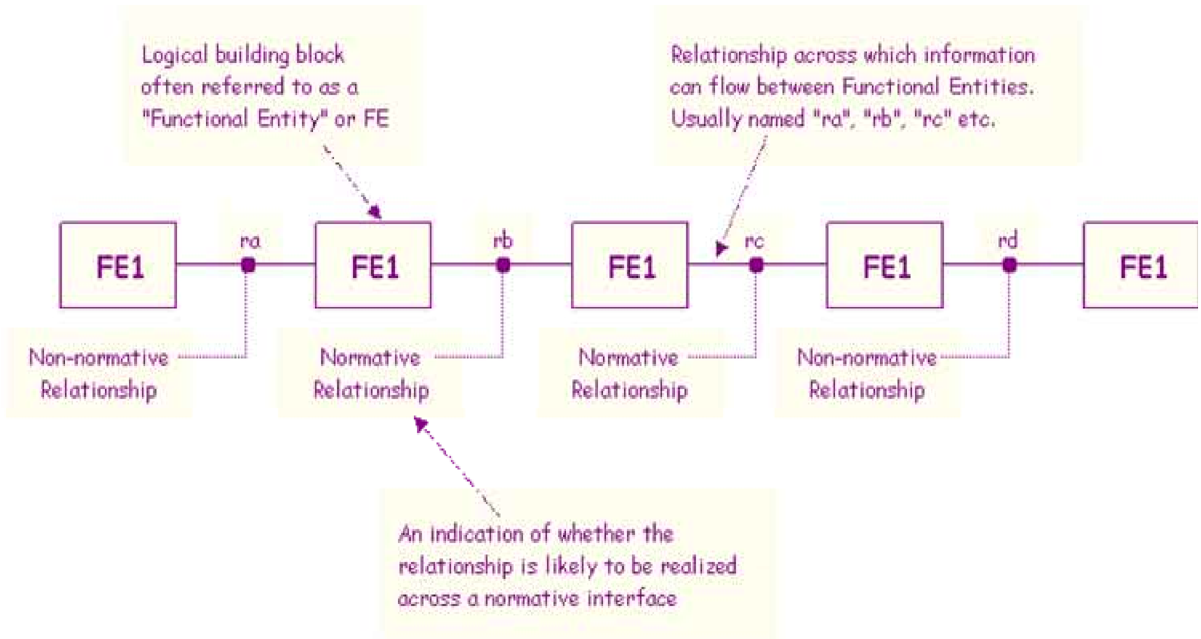


Figure B.1: Entity relationship diagram

The entity relationship diagram shown in figure B.1, together with information flow diagrams (sequence diagrams) are the most important elements of the logical model. The behaviour of each FE may be specified with UML or SDL state charts. However, stage two behaviour descriptions should remain significantly simpler then stage 3 behaviour descriptions.

The stage 2 information may be completed by providing a set of legitimate scenarios for the distribution of the logical blocks within a set of physical entities. Textual tables have traditionally been used quite effectively for this purpose but UML deployment diagrams can provide a graphical means of presenting these requirements.

NOTE: In the model presented in this annex whilst SDL was used in development the detail behaviour of each FE has been excluded, also excluded is a mapping of FEs to physical entities.

B.2 Functional entity model and information flows

B.2.1 Functional entity model

B.2.1.1 Description of model

NOTE: The model presented here is loosely based on the model presented in TS 101 882-3 [12] extended to allow more than one media selection in each call setup request.

This functional entity model covers the signalling aspects of call control for DUST multimedia calls (on the assumption that the signalling for such calls is essentially identical to that for voice calls) with call routing but does not include the flow of data in the media path. The model offered here allows for policy control of the call (i.e. the call progress may be controlled by checking the authority of the calling user to invoke the particular call model). The model offers the architectural capability to allow zero, one, or more, intervening networks in the signalling path.

The model comprises a number of functional entities split into the following key groups: Call control; Media control; Authorization; and, Routing.

The functional entity model for DUST is shown in figure B.2 (picture taken from the SDL model for DUST).

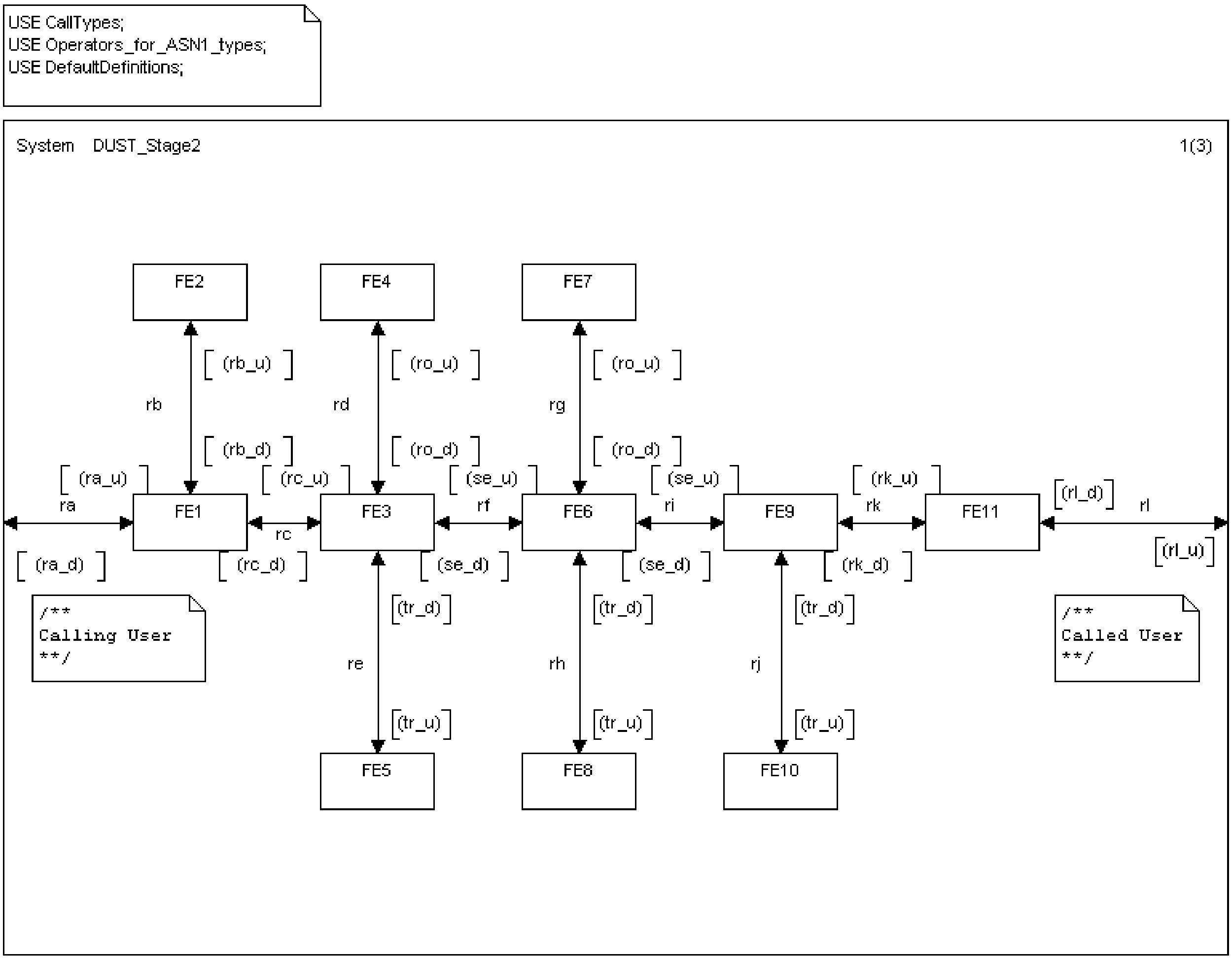


Figure B.2: Functional Entity Model for DUST

The functional entity model uses the Functional Entities (FE) described in table B.1.

Table B.1: Functional Entities for DUST

FE identity	Short description
Calling User	The application at the calling user's terminal which instigates the service request
$FE1_{OTC}$	The originating user service agent in the calling user's terminal that instigates the service request and processes call and bearer functions.
$FE2_{PE}$	The serving network policy control function associated with the calling user's service provider
$FE3_{ONC}$	The originating call and bearer coordination function that is responsible for establishing the call on behalf of the calling user.
$FE4_{OR}$	The originating call routing function, providing routing information and number/address translations (see note 1)
$FE5_{ONM}$	The originating media coordination function serving the calling user
$FE6_{INC}$	An intervening call and bearer control coordination function. This FE is responsible for establishing the call via the intervening domain
$FE7_{IR}$	An intervening routing function (see note 1)
$FE8_{INM}$	An intervening media coordination function
$FE9_{TNC}$	The destination call and bearer coordination function that is responsible for establishing the requested call on behalf of the called user
$FE10_{TNM}$	The destination media coordination function serving the called user
$FE11_{TTC}$	The service agent that processes an incoming call to the called user, processing call and bearer functionality
$FE12_{OTM}$	The media control function in the calling user's terminal. (see note 2)
$FE13_{TTM}$	The media control function in the called user's terminal. (see note 2)
Called User	The application in the called user's terminal at which the service request is terminated.
NOTE 1: Routing is based upon both the called user's identity and the requested service which includes QoS parameters. NOTE 2: The media control functional entities in the originating and terminating terminal are not part of the modelled behaviour.	

The functional relationships in table B.2 are part of the functional entity model for DUST.

Table B.2: Relationships between FEs

ra	between the Calling User and the Calling User's service agent ($FE1_{OTC}$);
rb	between the Calling User's service agent ($FE1_{OTC}$) and the Policy function ($FE2_{PE}$);
rc	between the Calling User's service agent ($FE1_{OTC}$) and the originating call and bearer coordination function ($FE3_{ONC}$);
rd	between the originating call coordination function ($FE3_{ONC}$) and the originating call routing function ($FE4_{OR}$);
re	between the originating call coordination function ($FE3_{ONC}$) and originating media coordination function ($FE5_{ONM}$);
rf	between the originating call coordination function ($FE3_{ONC}$) and an intervening call and bearer coordination function ($FE6_{INC}$);
rg	between an intervening call coordination function ($FE6_{INC}$) and an intervening routing function ($FE7_{IR}$);
rh	between an intervening call coordination function ($FE6_{INC}$) and an intervening media coordination function ($FE8_{INM}$);
ri	between an intervening call coordination function ($FE6_{INC}$) and the destination call and bearer coordination function ($FE9_{TNC}$);
rj	between the destination call coordination function ($FE9_{TNC}$) and the destination media coordination function ($FE10_{TNM}$);
rk	between the destination call coordination function ($FE9_{TNC}$) and the service agent that processes an incoming call to the called user ($FE11_{TTC}$);
rl	between the called user's service agent function ($FE11_{TTC}$) and the Called User;
rm	between the Calling User's service agent ($FE1_{OTC}$) and the media control function in the originating user's terminal (see note);
rn	between the called user's service agent function ($FE11_{TTC}$) and the media control function in the called user's terminal (see note).
NOTE: The relationship between the call control FE and the media control FE in the originating and terminating terminal is not defined in the present document.	

B.2.1.2 Description of functional entities

B.2.1.2.1 Calling User

The Calling User functional entity acts on behalf of an end user to request the establishment of a call.

B.2.1.2.2 Calling User's service agent, $FE1_{OTC}$

On receipt of a call set-up request from the Calling User, FE1 requests FE2 to validate the request e.g. to ascertain if current policy permits the requested QoS to be used in the call. If the request is valid FE1 forwards the call set-up request to FE3.

B.2.1.2.3 The policy control function associated with the calling user's service provider, $FE2_{PE}$

This FE checks if the calling user is authorized to invoke the service and may as part of the verification of authorization check its service database to determine if requested service parameters are permitted in the requested call, e.g. QoS parameters.

B.2.1.2.4 Originating network call coordination function, $FE3_{ONC}$

This FE requests FE4 for destination address information and requests the provision of appropriate media resource from the o FE5. If these requests are successful it then passes the call set-up request to the next call coordination function, FE6 (where an intervening network exists) or FE9 (where no intervening network exists).

B.2.1.2.5 Originating network routing function, $FE4_{OR}$

FE4 checks the provided destination address from the originating call coordination function FE3 and returns information on the completeness of the called address and next hop information.

B.2.1.2.6 Originating network media coordination function, $FE5_{ONM}$

This FE attempts to establish a media connection with the requested capabilities between the indicated incoming user access point and an outgoing network access point providing a media connection between the calling and called user possibly via intervening media control functions. The media capabilities for a DUST call are as specified in the main body of the present document.

B.2.1.2.7 Intervening network call coordination function, $FE6_{INC}$

If present, this FE requests FE7 for destination address information and requests the provision of appropriate media resource from FE8 before passing the call set-up request to the next call coordination function, FE6 or FE9.

B.2.1.2.8 Intervening network routing function, $FE7_{IR}$

If present, FE7 checks the provided destination address from the intervening call coordination function FE6 and returns information on the completeness of the called address and next hop information.

B.2.1.2.9 Intervening network media coordination function, $FE8_{INM}$

If present, this FE attempts to establish a media connection between the indicated incoming user access point and an outgoing network access point providing an appropriate connection between the calling and called user possibly via intervening media control functions.

B.2.1.2.10 Destination network call coordination function, $FE9_{TNC}$

This FE requests the provision of appropriate media resource from the destination network media control function and then passes the call set-up request to the called user service agent.

B.2.1.2.11 Destination network media coordination function, $FE10_{TNM}$

The FE attempts to establish a media connection between the indicated network access point and the called user access point providing an appropriate connection between the calling and called user.

B.2.1.2.12 Called user's service agent, $FE11_{TTC}$

This FE informs the end user that a call set-up request is being made and conveys the Called User response to the destination network call control function, FE9.

B.2.1.2.13 Called User

The Called User functional entity acts on behalf of the called end user to respond to a simple call set-up.

B.2.2 Information flows

B.2.2.1 Definition of information flows

An information flow identifies the essential information required to establish a service between functional entities. There is no definition of element encoding implied in the structures that are given in the following clauses. There is also no implied definition of Protocol Data Units in the information flows.

The following conventions are used to specify information flows:

- An information flow is referred to as a "request" at the FE that sends it and as an "indication" at the FE that receives it. Only the request is defined in the tables.
- The corresponding confirmation is referred to as a "response" at the FE that sends it and as a "confirmation" at the FE that receives it. Only the response is defined in the tables. If the information flow is unconfirmed no response is defined.

B.2.2.1.1 Relationship ra

B.2.2.1.1.1 CC_OrigCallSetup

CC_OrigCallSetup is a confirmed information flow across relationship ra from the calling user to FE1 to indicate a request for call establishment. The elements required in the information flow are listed below.

- Call Identifier:
  - This element is in the form of an alphanumeric identifier returned by FE1 to allow the calling user to simplify later signalling by using it is a shorthand to uniquely identify the call to be modified or cleared. It is also used by FE1 when informing the calling user of any state change of the call.
- Called user ID:
  - This element is in the form of an E.164 [45] number to allow reachability from any terminal in the global telecommunications world. Alternative formats for use in local networks may include SIP-URIs.

- Calling User ID restriction / presentation preference:
  - This element is required to allow compliance with regulatory conditions for CLIP and CLIR and can take one of two values:
    - Available (CLIP);
    - Unavailable (CLIR).

NOTE: The presentation preference may be overridden when making emergency calls (112).

- Calling User ID:
  - This element is in the form of an E164 number to allow reachability from any terminal in the global telecommunications world. It is present only if the CLIP/CLIR preference allows it. Alternative formats for use in local networks may include SIP-URIs or a display name.
- Operator selection:
  - This element is provided to allow the calling user to pre-select a preferred service provider.
- QoS Service Class:
  - This element allows the calling user to indicate the QoS service class to be used for the media elements of the call. This may take the form of a predefined short name known to the service provider, or be offered as a set of QoS parameters.
- Media descriptor:
  - If the QoS service class identifies "parameterized" this element is provided. More than one element of this type can be included in the call setup if more than one media type is to be established (where the different media are combined at the end points then only one composite media needs to be requested). Each media element is defined in terms of Media peak rate, and Maximum media frame size, maximum allowed delay, packet loss rate that can be corrected by the application, and jitter sensitivity of the application (variation in packet transmission time across the network).
- Result:
  - This information element is returned to the calling user to indicate success or failure of the call setup attempt. In normal behaviour the expected result is "Call established with requested QoS" but where a rejection is given it should identify the reason for rejection which may be one of the following:
    - Transport not available.
    - Requested QoS not available.
    - Called user unknown.
    - No compatible codec available.
    - Policy Rejection.
    - Called user busy.

B.2.2.1.1.2 TCC_CallRelease

TCC_CallRelease is a confirmed information flow across relationship ra between the calling user and FE1 when a call is terminated. The elements required in the information flow are listed below.

- Call Identifier.
- Cause code:
    - This element is used to inform the receiving party if the release was initiated by the user or by the network.
- Result:
    - This element is used in response to the request to inform the user if the release was successful or not.

B.2.2.1.1.3 TCC_CallAlerting

TCC_CallAlerting is an unconfirmed information flow across relationship rb from FE1 to the calling user when the called party is being alerted to the call and contains only the Call identifier element.

B.2.2.1.2 Relationship rb

B.2.2.1.2.1 CallAuthorization

CallAuthorization is a confirmed information flow across relationship rb from FE1 to FE2 to request permission for a new call establishment with a specific end-to-end QoS. The elements required in the information flow are listed below.

- Call Identifier.
- Called user ID.
- Calling User ID.
- QoS Service Class.
- Media descriptor.
- Result:
    - This information element is returned to FE1 to indicate success or failure of the authorization request. In normal behaviour the expected result is "Call permitted" but where a rejection is given it should identify the reason for rejection which may be one of the following:
        - Service not subscribed to.
        - Service not available.

B.2.2.1.3 Relationship rc

B.2.2.1.3.1 CallSetup

CallSetup is a confirmed information flow across relationship rc from the FE1 to the FE3. The elements required in the information flow are listed below.

NOTE: A bearer for media flow may be one of three types: unidirectional, bi-directional symmetric (default), and bi-directional asymmetric.

- Call Identifier.
- Called user ID.
- Calling User ID restriction / presentation preference.

- Calling User ID.
- Media descriptor.
- Result.

B.2.2.1.3.2 CallRelease

CallRelease is a confirmed information flow across relationship rc from FE1 to FE3. The elements required in the information flow are listed below.

- Call Identifier.
- Cause code.
- Result.

B.2.2.1.3.3 CallAlerting

CallAlerting is an unconfirmed information flow across relationship rc from FE3 to FE1 when the called party is being alerted to the call and contains only the Call identifier element.

B.2.2.1.4 Relationship rd, rg

B.2.2.1.4.1 CallRoute

CallRoute is a confirmed information flow across relationship rd and rg from FE3/FE6 to FE4/FE7 to indicate a request for address and routing information. A routing decision may be based on a number of criteria and these need to be included in the *CallRoute_request*. The elements required in the information flow are listed below.

- Call Identifier.
- Called user ID.
- Calling User ID.
- Operator selection.
- Media descriptor:
    - If the QoS service class identifies "parameterized" this element is provided. More than one element of this type can be included in the call setup if more than one media type is to be established (where the different media are combined at the end points then only one composite media needs to be requested). Each media element is defined in terms of Media peak rate, and Maximum media frame size, maximum allowed delay, packet loss rate that can be corrected by the application, and jitter sensitivity of the application (variation in packet transmission time across the network).
- Routing information:
    - This element is returned in *CallRoute_response* to FE3/FE6 to identify the path for signalling.
- Result.

B.2.2.1.5 Relationship re, rh, rj

B.2.2.1.5.1 MediaReservation

MediaReservation is a confirmed information flow across relationships re, rh, and rj to request the reservation of a media flow towards the called user's address. The media is allocated using a reserve-confirm pair of flows whereby the media resources are reserved pending completion of the call signalling and either released on failure or confirmed on success. The elements required in the information flow are listed below.

- Bearer Identifier.
- QoS parameters qualification:
  - This information element is used to indicate the cumulative QoS contribution of each network.
- Media identifier.
- Media descriptor.
- Result.

B.2.2.1.5.2 MediaEstablishment

MediaEstablishment is a confirmed information flow across relationships re, rh, and rj to request the establishment of a previously reserved bearer and media transport path towards the called user's address. The elements required in the information flow are listed below.

- Bearer Identifier.
- Media identifier.
- Result.

B.2.2.1.5.3 MediaRelease

MediaRelease is an unconfirmed information flow across relationships re, rh, and rj to request that a previously reserved bearer and media transport path is released as it is no longer required. The elements required in the information flow are listed below.

- Bearer Identifier.
- Media identifier.

B.2.2.1.6 Relationship rf, ri

B.2.2.1.6.1 NwCallSetup

NwCallSetup is a confirmed information flow across relationship rf and ri from FE3 to FE6 and from FE6 to FE9 (where an intervening network exists), or from FE3 to FE9 (where no intervening network exists). The elements required in the information flow are listed below.

- Call Identifier.
- Called user ID.
- Calling User ID restriction / presentation preference.
- Calling User ID.
- Media descriptor.
- Result.

B.2.2.1.6.2 NwCallRelease

NwCallRelease is a confirmed information flow across relationship rf and ri sent between originating, intervening or destination call coordination FEs. The elements required in the information flow are listed below.

- Call Identifier;
- Cause code;
- Result.

B.2.2.1.6.3 NwCallAlerting

NwCallAlerting is an unconfirmed information flow across relationships rf and ri from the destination or an intervening call coordinating FE to the originating or an intervening call coordination FE when the called party is being alerted to the call and contains only the Call identifier element.

B.2.2.1.7 Relationship rk

B.2.2.1.7.1 DestCallSetup

DestCallSetup is a confirmed information flow across relationship rk from FE9 to FE11. The elements required in the information flow are listed below.

- Call Identifier.
- Called user ID.
- Calling User ID restriction / presentation preference.
- Calling User ID.
- Media descriptor.
- Result.

B.2.2.1.7.2 CallRelease

CallRelease is a confirmed information flow across relationship rk between the destination call coordination function and the called user service agent. The elements required in the information flow are listed below.

- Call Identifier.
- Cause code.
- Result.

B.2.2.1.8 Relationship rl

B.2.2.1.8.1 TCC_DestCallSetup

TCC_DestCallSetup is a confirmed information flow across relationship rm from the FE11 to the Called User to indicate an incoming call. The elements required in the information flow are listed below.

- Call Identifier.
- Called user ID.
- Calling User ID restriction / presentation preference.
- Calling User ID.
- Media descriptor.
- Result.

B.2.2.1.8.2 TCC_CallRelease

TCC_CallRelease is a confirmed information flow across relationship rl between the called user and FE11 when a call is terminated. The elements required in the information flow are listed below.

- Call Identifier.
- Cause code.
- Result.

B.2.2.2 Timers

"Reservation Hold Timer": A reservation hold timer is used in the call coordination functional entity to make sure that resource is not held indefinitely.

Annex C (informative): Stage 3 requirements - Session Initiation Protocol (SIP)

C.1 DUST Characteristics in SIP environment

The DUST communication model should work over a standard protocol stack defined for the NGN project by TISPAN. The key elements of this model are shown as follows:

- Control protocol SIP with SDP.
- Transport protocol TCP and UDP for signalling, RTP over UDP for media.
- Text presentation protocol ITU-T Recommendation T.140 [64].
- Text transport protocol RTP with RFC 4103 [42].
- Control protocol action to declare text capabilities and open text transmission.
  The actions are described in RFC 4103 [42].
- Speech encoding. Determined by service provider.
- Video encoding ITU-T Recommendation H.263 [76].

NOTE: The specific encoding and protocols are examples and developers may select alternatives.

C.2 Signalling characteristics

When using SIP (RFC 3261 [30]) to signal the session the media attributes and RTCP attributes may be established in the session definition using SDP. Where text support is according to ITU-T Recommendation T.140 [64] and transported as specified in RFC 4103 [42].

During call setup the SIP protocol also gives details on how the capability to handle text is declared. Text packets are only sent if both parties declare that they can handle text.

The use of real time conversational text in a telecommunications environment is supported in the IETF by RFC 4103 [42] and by recent work in the Sipping group as outlined in table C.1.

Table C.1: Internet draft prepared by IETF Sipping group

Name	Title	Author	Working Group	Date
draft-ietf-sipping-toip-00	Framework of requirements for real-time text conversation using SIP.	Arnold Van Wijk	sipping	22-Aug-05

C.3 Transmission characteristics

C.3.1 Control of transmission characteristics

Where RTP is used to carry the media RTCP should used to control the media session.

C.3.1.1 Transport mechanism

The characters entered by the user need to be transported from the transmitting terminal to the receiving terminal. They are composed of digital codes according to ITU-T Recommendation T.140 [64]. Transport mechanisms usually depend on the network and host environment, there being one transport mechanism specified for text for each host environment.

This clause describes the transport mechanism for IP networks using the IETF SIP host environment. A large part of the description is also valid for the ITU-T Recommendation H.323 [57] host environment.

C.3.1.1.1 Packetizing

In order to transmit information in the IP network, the data needs to be collected in packets. These packets always begin with information used for the transport. It is headers with addressing information, handling information for handling the packet and basic information about the contents. The packets are sent into the network, and the components in the network use the header information to decide on the delivery route to the destination and other factors to give the packet proper handling.

C.3.1.1.2 Text transmission with RTP

Media is transmitted with Real Time Protocol (RTP). By specifying a method based on RTP for the text transmission in SIP calls, the same mechanism is used as for other media. Audio and video are also sent using RTP, using other payload descriptions. This allows the text medium to be treated in the same manner as other media in the call and so reduces the risk of blocking. No extra servers are required, no special routing and no special protocol support. Currently, RTP is the only formally allowed protocol for media transmission in SIP calls, and therefore is also the natural choice for text.

For each type of medium and coding, a specification must exist, describing how this medium is put in packets and transmitted. These specifications are called RTP payload specifications. For real time text, the RTP payload specification is RFC 4103 [42], "RTP Payload for text conversation". This specification describes how to put the characters and other related information in packets, and also describes a procedure to make the transmission reliable.

C.3.1.1.3 Reliability

Characters that have been transmitted are saved at the transmitting side, and retransmitted twice in subsequent packets, with short intervals together with any new characters that the transmitting user has entered. The interval is recommended to be 300 ms in order to give the users a good real time experience without causing any cumbersome network load. The packets are assigned sequence numbers, so that at arrival, any missing packets can be detected, and the text that was contained in them can be retrieved from any of the following two packets. This scheme for introducing reliability in text transmission builds on a general method for redundant transmission called RFC 2198 [19].

C.3.1.1.4 Error recovery

In the rare circumstance that three packets in sequence are lost in transmission, this fact is revealed, and a character marking lost text according to ITU-T Recommendation T.140 [64] is inserted for information in the text stream by the receiving process. It is often said that packet loss in transmission appears in bursts. The effect of such bursts on total character loss is minimized by separating the packet transmissions by 300 ms.

A packet loss rate of 1 % is regarded to be bad conditions for IP networks. With the recommended two repetitions, the resulting total loss of text will be 0,0001 % or one character out of 1 million. Any loss will be marked with the special character indicating loss. Human typing errors outnumbers this error rate by many magnitudes.

C.3.1.1.5 Security

Security can be negotiated at call setup, where a method called Secure Real Time Protocol, SRTP can be initiated (when both parties are capable of handling it) for achieving protection against various security threats.

C.4 Support of supplementary services

Due to the fact that SIP is stateless and currently there is no agreed mechanism within the IETF to intercept and interpret the SIP signalling at the proxy server (see stateless proxy), there are currently problems known in implementing the following supplementary services:

- CLIR/CLIP.
- Call Forward on Busy.
- Call Forward on No Answer.
- Call Forward Unconditional.
- MCID (Malicious Caller Identity).
- Call waiting.
- Call HOLD.

Annex D (informative): Stage 3 requirements - ITU-T Recommendation H.323

D.1 H.323

In packet networks, the procedures described in ITU-T Recommendation H.323 [57] annex G can be used for text conversation sessions.. A simple packet text telephone is defined, called a Text Simple Endpoint Type (Text SET).

Use of ITU-T Recommendation H.323 [57], H.245 [52] and H.225.0 [51] are most commonly implemented within systems that also have an association with ISDN. Examples of this include IP PBX and Videoconferencing solutions where the call state is required and implements a stateful proxy server.

In this environment text communication is supported according to ITU-T Recommendation H.323 [57], annex G with ITU-T Recommendation T.140 [64] transported as specified in RFC 4103 [42].

- Control protocol H.245 with parameters for T.140 text H.225.0 for signalling.
- Transport protocol TCP and UDP for signalling, RTP over UDP for media transport.
- Text presentation protocol ITU-T Recommendation T.140 [64].
- Text transport protocol RTP with RFC 4103 [42], described in annex C.
- Control protocol action to declare text capabilities and open text transmission.
  Use T.140 parameters in H.245 to declare, negotiate and open a logical channel for text.
  The actions are described in H.323 annex G.
- Speech encoding ITU-T Recommendation G.723.1 [77].
- Video encoding. ITU-T Recommendation H.263 [76].

NOTE: Firewall and NAT traversal requires careful planning,

Annex E (informative): Stage 3 requirements - ISDN

E.1 H.320

In ISDN networks, the ITU-T Recommendation H.320 [55] multimedia environment offers an opportunity to declare and transport text in combination with audio and video.

- Network protocol — ISDN.
- Control protocol — ITU-T Recommendations H.320 [55] and H.221 [49].
- Transport protocol — Not defined.
- Text presentation protocol (usually ITU-T Recommendation T.140 [64]) — ITU-T Recommendation T.140 [64].
- Text transport protocol — ITU-T Recommendation H.224 [50], client ID=2.
- Control protocol action to declare text capabilities and open text transmission.
  H.224 channel declared and negotiated in H.221, and client ID=2 channel opened for text.
- Speech encoding. — ITU-T Recommendation G.728 [78].
- Video encoding — ITU-T Recommendation H.263 [76].

NOTE: ISDN connection usually established as unrestricted digital.

E.2 ITU-T Recommendation H.324 and H.225

The ITU-T Recommendation H.324 [58] multimedia environment can be used in PSTN and ISDN networks.

- Network environment — PSTN or ISDN.
- Control protocol — ITU-T Recommendation H.245 [52].
- Text presentation protocol — ITU-T Recommendation T.140 [64].
- Text transport protocol — AL1.
- Control protocol action to declare text capabilities and open text transmission.
  H.245 capability negotiation and logical channel opening with T.140 parameters in H.245.
- Speech encoding. — ITU-T Recommendation G.723.1 [77].
- Video encoding — ITU-T Recommendation H.263 [76].

NOTE: PSTN transmission quality is at risk by IP transit

Annex F (informative): Stage 3 requirements - 3GPP

F.1 3GPP IP Multimedia Subsystem

In the IP MMultimedia Subsystem of 3G mobile networks, multimedia connections can be established with the SIP protocol.

The protocol stack for DUST calls in this environment is similar to for other SIP networks, described in this clause, except that the default audio coder is decided to be AMR.

The codecs to use are specified in TS 126 235 [18].

The stage 2 specification is found in the parts of TS 123 226 [14] describing GTT-IPMM.

The service description is found in the parts of TS 122 226 [13] that are IP specific or of general nature.

- Network environment IP in UMTS network.
- Control protocol SDP.
- Text presentation protocol ITU-T Recommendation T.140 [64].
- Text transport protocol RTP with RFC 4103 [42], described in annex C.
- Control protocol action to declare text capabilities and open text transmission.
  In SDP, declare and open the text channel with m = statement as described in RFC 4103 [42].
- Speech encoding. AMR.
- Video encoding ITU-T Recommendation H.263 [76].
- Network transmission considerations IP in the IP Multimedia Subsystem of 3GPP networks.

F.2 3GPP Circuit-Switched CS MM

The Circuit switched multimedia environment in 3GPP Mobile networks, can be used as a DUST host environment. This environment is described in TS 123 226 [14], under the name GTT-CSMM.

- Network environment 3G Circuit switched network.
- Control protocol H.324M with H.245.
- Text presentation protocol ITU-T Recommendation T.140 [64].
- Text transport protocol AL1.
- Control protocol action to declare text capabilities and open text transmission.
  H.245 capability negotiation and logical channel opening with T.140 parameters in H.245.
- Speech encoding. AMR.
- Video encoding ITU-T Recommendation H.263 [76].
- Network transmission considerations Use 3GPP network.

Annex G (informative): Stage 3 Requirements - Data conferencing

G.1 T.120

Text conversation in the ITU-T Recommendation T.120 [61] data conferencing environment, is specified in ITU-T Recommendation T.134 [63]. It is a simple encapsulation of ITU-T Recommendation T.140 [64]. It can be used in conjunction with any standardized protocol for audio or video, thereby forming a suitable DUST host environment.

- Network environment Any.
- Control protocol ITU-T Recommendation T.124 [79].
- Text presentation protocol ITU-T Recommendation T.140 [64].
- Text transport protocol ITU-T Recommendation T.125 [62].
- Control protocol action to declare text capabilities and open text transmission.
  Specified in ITU-T Recommendation T.134 [63].
- Speech encoding. Depends on the environment where T.134 [63] is applied.
- Network transmission considerations Must be selected to meet user requirements.

Annex H (informative): Reference software

H.1 Reference software information

Software implementing a basic IP text telephone available from http://sourceforge.net/. It is based on the SIP host environment, and was developed with the aim of supporting a DUST terminal in accordance with the guidelines given in annex C. Please search the Sourceforge open source library for the items "SIPcon1", "T.140" and "text/t140" to get the latest versions.

The software can be used as a reference design for product developments, as an interoperability test partner, as a source of components and as an operational basic IP text telephone. However, no guarantees are provided with respect to the function or compliance. The software has the following main characteristics:

- Developed in Java.
- Portable to various environments, but prepared for use in Microsoft Windows®.
- Using the SUN JMF® communications package.
- Offering the possibility to register in a SIP registrar for SIP address resolution and call control.
- Real time text according to RFC 4103 [42].
- Compatible with IP telephony using SIP.
- Extendable through further development to support other functions, such as video, presence etc.

The software is provided with open source licences. Please see information stored in each package for conditions.

Provision of this software has been made possible by the European Commission, who partly funded development in the IST project WISDOM (IST 2000-27512).

This software was also partially funded by the National Institute on Disability and Rehabilitation Research, US Dept of Education under Grant H133E040013 as part of the Telecommunication Access Rehabilitation Engineering Research Center of the University of Wisconsin -Trace Center jointly with Gallaudet University.

The software is available in three parts. Information on usage is provided in each package.

H.2 Executable IP text telephone SIPcon1

The package "SIPcon1" contains an executable package implementing a complete IP text and voice phone.

This application is suitable for anyone who wants to try communication with ITU-T Recommendation T.140 [64] and RFC 4103 [42].

To run, the application requires a PC running Windows® with at least a 500 MHz CPU and an Internet connection (if a firewall is in use, it has to be SIP capable).

It can be installed by executing the installation program named SIPcon1-v1.3.2.exe.

H.3 Source code for T.140 text presentation protocol

The package "T.140 Presentation library" contains source code in Java for the T.140 text presentation level implementation.

This library is suitable for developers.

H.4 Source code for IP transport of text

The package "RTP text/t140" is a reference implementation for RTP payload Type for Text Conversation (RFC 4103 [42]). It contains the source code in Java for encoding and decoding RFC 4103 [42] text and may be used as a plug-in to JMF or in a separate RTP sender/receiver.

This library is suitable for developers.

Annex I (informative): Bibliography

- ETSI "Making better standards": http://portal.etsi.org/mbs/home.asp.
- IETF draft-rosenberg-midcom-turn.txt "Traversal Using Relay NAT (TURN)".
- IETF draft-ietf-mmusic-ice-04.txt: "Interactive Connectivity Establishment (ICE) is a Methodology for Network Address Translator (NAT) Traversal for Multimedia Session Establishment Protocols" (J. Rosenberg).
- IETF draft-mspencer-dundi-01: "Distributed Universal Number Discovery (DUNDi)".
- Ballabio, E: "COST 220 If I call, who can answer? Text telephone systems in use in Europe - A way forward to compatibility" - Office for the official publications of the European Communities 1991.
- Martin, M.C 1: "Telecommunications and the hearing impaired" - Proc 13th International Symposium on Human Factors in Telecommunications (1990) p.189.
- Martin, M.C 2: "Communication by text" COST 219 Issues in Telecommunication and disability - Office for the official publications of the European Communities 1991.
- Oleson, K.G: "Survey of text telephones and relay services in Europe" Office for the official publications of the European Communities 1992.
- ANSI TIA/EIA-825: "A Frequency Shift Keyed Modem for use on the Public Switched Telephone Network".
- ETSI EG 201 872: "Methods for Testing and Specification (MTS); Methodological approach to the use of object-orientation in the standards making process".
- ETSI EG 202 067: "Universal Communications Identifier (UCI); System framework".
- ETSI EG 202 072: "Universal Communications Identifier (UCI); Placing UCI in context; Review and analysis of existing identification schemes".
- ETSI TR 101 767: "Human Factors (HF); Symbols to identify telecommunications facilities for deaf and hard of hearing people; Development and evaluation".
- ETSI TR 101 878: "Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON) Release 5; Service Capability Definition; Service Capabilities for a Multi Media Call".
- ETSI TR 101 882: "Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON) Release 5; Protocol Framework Definition and Interface Requirement Definition; General".
- ETSI TR 103 073: "Universal Communications Identifier (UCI); Improving communications for disabled, young and elderly people".
- ETSI TS 101 329-2: "Telecommunications and Internet Protocol Harmonization Over Networks (TIPHON) Release 3; End-to-end Quality of Service in TIPHON systems; Part 2: Definition of speech Quality of Service (QoS) classes".
- IETF RFC 1034: "Domain names - concepts and facilities".
- IETF RFC 3103: "Realm Specific IP: Protocol Specification".
- IETF RFC 3403: "Dynamic Delegation Discovery System (DDDS) Part Three: The Domain Name System (DNS) Database".
- IETF RFC 3482: "Number Portability in the Global Switched Telephone Network (GSTN): An overview".
- ITU-T Recommendation H.223: "Multiplexing protocol for low bit rate multimedia communication".

- ITU-T Recommendation I.140: "Attribute technique for the characterization of telecommunications services supported by an ISDN and network capabilities of an ISDN".
- ITU-T Recommendation I.210: "Principles of telecommunication services supported by an ISDN and the means to describe them".
- ITU-T Recommendation T.123: "Network specific data protocol stacks for multimedia conferencing".
- ITU-T Recommendation V.80: "In-band DCE control and synchronous data modes for asynchronous DTE".
- ITU-T Recommendation V.34: "A modem operating at data signalling rates of up to 33 600 bit/s for use on the general switched telephone network and on leased point-to-point 2-wire telephone-type circuits".

History

Document history				
V1.1.1	August 2005	Membership Approval Procedure (Withdrawn)	MV 20050930:	2005-08-02 to 2005-09-30
V1.2.1	August 2005	Membership Approval Procedure	MV 20051021:	2005-08-23 to 2005-10-21