ETSI TS 122 226 V16.0.0 (2020-08)

TECHNICAL SPECIFICATION

Digital cellular telecommunications system (Phase 2+) (GSM);
Universal Mobile Telecommunications System (UMTS);
LTE;
Global Text Telephony (GTT);
Stage 1
(3GPP TS 22.226 version 16.0.0 Release 16)

Reference
RTS/TSGS-0122226vg00

Keywords
GSM,LTE,UMTS

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Contents

Foreword

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Introduction

Real time, character by character text conversation is a component that increases usability in a conversational session.

Global Text Telephony is a feature that adds the capability to use a text conversation component in a session. It is called Global Text or GTT here.

Interworking with corresponding features in other networks is an important part of Global Text. Specifically, the different kinds of PSTN text telephone systems supported by the international text telephone modem standard ITU-T V.18 are included in the modes for interworking.

One important reason to offer the Global Text feature is to enable emergency service access to people who are depending on a written dialogue.

A more elaborated background is found in Annex A, and requirements in Annex B.

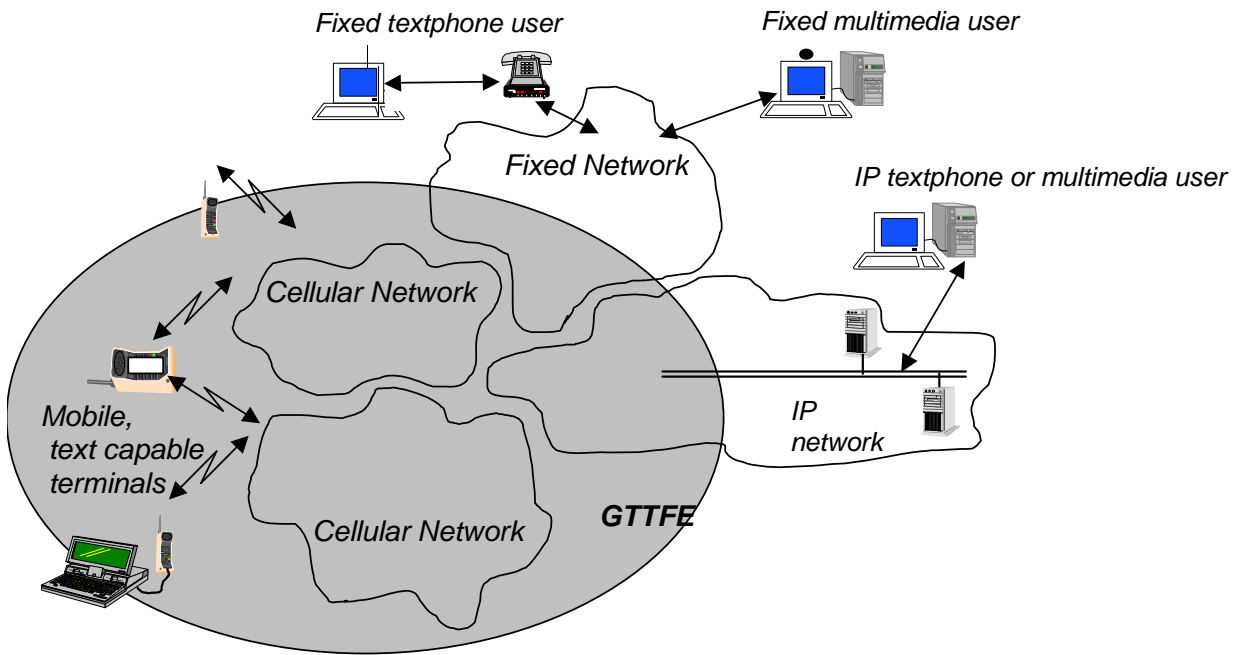Figure 1 gives an overview of the user environment for Global Text.

Figure 1. Global Text service environment

1 Scope

This Technical Specification defines the stage one description of the Global Text Telephone Feature, GTT.. Stage one is the set of requirements which shall be supported for the provision of the real-time text conversation feature, seen primarily from the subscriber's and service providers' points of view.

This TS includes information applicable to network operators, service providers, terminal and network manufacturers.

This TS contains the core requirements for the Global Text Telephony feature, which are sufficient to provide a complete feature to incorporate in conversational services.

This TS defines the requirements for GTT to be understood as a framework to enable real-time transmission of text, for the purpose of a text based or text supported conversation between users. Text may be transported alone or in combination with other media in the session, especially video and voice.

Thus the GTT enables text conversation to be included in any mobile conversational service.

Interworking with existing text telephony in PSTN as well as emerging forms of standardised text conversation in all networks is within the scope of this document. Interoperation with Multimedia Messaging Services is also within scope of this feature.

Note: The Global Text Telephony feature may be enhanced due to e.g. operator's or regulator's requirements, however such additional functionality shall not compromise conformance to the core requirements documented in this TS.

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, edition number, version number, etc.) or non-specific.

- For a specific reference, subsequent revisions do not apply.

- For a non-specific reference, the latest version applies. In the case of a reference to a 3GPP document (including a GSM document), a non-specific reference implicitly refers to the latest version of that document *in the same Release as the present document*.

[1] 3GPP TS 22.101: "Service Principles".

[2] ETSI ETR 333; Text Telephony; User requirements and Recommendations

[3] ITU-T H.248 Annex F Facsimile, Text conversation and call discrimination packages (2000)

[4] ITU-T Recommendation T.140 (1998) – Text conversation protocol for multimedia application. With amendment 1 (2000).

[5] ITU-T Recommendation F.700 (2000) – Framework recommendation for audiovisual/multimedia services.

[6] ITU-T Recommendation V.18 (2000) – Operational and interworking requirements for DCE:s operating in the text telephone mode.

[7] 3GPP TS 22.140: "Multimedia Messaging Service; Stage 1"

[8] 3GPP TS 23.140: "Multimedia Messaging Service; Functional description; Stage 2"

[9] IETF RFC 2793. (2000) RTP Payload for text conversation.

[10]   3GPP TR 21.905: "Vocabulary for 3GPP Specifications".

3   Definitions and abbreviations

3.1   Definitions

Total Conversation: A service offering standardised simultaneous text, video and voice conversation or a subset thereof.

Host environment : The session environment where the text component is added. E.g. Circuit switched voice, IP Multimedia etc.

Text Conversation: A real time conversation in text with transmission character by character as entered.

Further definitions are listed in 3GPP TR 21.905 [10].

3.2   Abbreviations

For the purposes of this document the following abbreviations apply:

EDT	European Deaf Telephone
GTT	Global Text Telephony; The feature that adds real time text conversation to any 3GPP conversational environment
GTTFE	Global Text Telephony Feature Environment – The network components and functions forming GTT
TTY	Here used as the term for the text telephone type dominating in USA.
DTMF	Here used as a term for the text telephone type used in Holland, using DTMF tones.
VCO	Voice Carry Over: Alternating (or parallel) sending of Speech and receiving Text
HCO	Hear Carry Over: Alternating (or parallel) receiving Speech and transmission of Text
FCC	Federal Communications Commission (of United States of America)
PER	Printable character Error Rate
MMI	Man Machine Interface
SIM	Subscriber Identification Module
ITU	International Telecom Standardisation Union
GSM	Global System for Mobile communication
3GPP	Third Generation Partnership Project
CS	Circuit Switched
IP	Internet Protocols

Further abbreviations are listed in TR 21.905 [10].

4   High level Requirements

The following list gives the high level requirements of the GTT. These are requirements which are independent of the user's perception of the feature:-

- Service definition

   Global Text shall provide a real time conversational text feature. A general definition is found in Recommendation ITU-T F.700[5].

- Global Text Telephony host environments

   A standardised method for Global Text Telephony shall at least be specified for each host environment in the mobile networks that can carry voice.

- Standards compliant and forward compatible text conversation

Global Text Telephone mechanisms shall provide the capability to support current and evolving text telephony and text conversation features by re-using existing standards as far as possible and proposing extensions (as necessary) to existing standards (i.e. the global text telephony feature shall support the evolution of Total Conversation technologies in all networks).

- Consistent text conversation

Regardless of the selected host environment, the far end terminal, the interworking facilities involved, GTT shall be capable of providing a consistent way of handling the conversation.

- Global Text Telephone access

Within the capabilities of networks and terminals, the user shall be provided consistent access to the GTT regardless of the access point.

For example: the user shall be capable of accessing the text conversation features through a number of different access points, including networks based on 3GPP specifications, access through dedicated multimedia terminals, and access through combinations of mobile phones combined with text user interface devices.

- Interoperability

Global Text Telephony shall support interoperability with existing and emerging text telephone systems and text conversation features.

Global Text Telephony shall support a minimum set of environments where text conversation is supported to ensure full interoperability between different terminals and networks.

- Emergency calls

The implementation of Global Text Telephony shall enable a user to make emergency calls to, and receive calls from, an emergency call centre via a text telephony device used in conjunction with GTT enabled user equipment.

5　General Requirements

Network operators have many differing requirements, and GTT shall be supported in the network in a manner which allows network operators to consider different configurations depending on their network and commercial requirements. Thus, an identified set of functionalities and protocols shall be standardised to ensure interoperability across networks and terminals to support GTT.

The following general requirements shall be supported.

5.1　Text conversation host environments

The protocol environments for text conversation shall be the same as the ones used for other multimedia conversation calls and voice calls. The selected environment for a session is called the host environment.

Supported host environments are:

- Packet switched multimedia

- Circuit switched multimedia.

- The text conversation carried by an data transmission procedure, possible to combine with a voice session.

- Circuit switched voice telephony, with the text conversation carried in-band in the voice channel.

5.2　Text conversation management

It shall be possible to allow GTT use without GTT specific subscription.

Note: There is no requirement for GTT feature specific subscription, but for "normal GTT calls" a subscription for telephony is required. For "emergency GTT call" no subscription of any kind is required.

Session start up delay and loss of initial characters should be minimised as far as practicably possible.

The text transport method should remain transparent to the network, unless specific functionality is required to provide a satisfactory quality of service.

5.3 Registration of text conversation capabilities

The availability of text conversation capabilities in the user terminal may need to be communicated to the network at call establishment in order to access the GTT service. However, a user is expected to be able to access the GTT service without explicit GTT registration with the operator.

5.4 Session control

The session control (call control) functions shall use the same procedures as the selected host environment.

Additional operations needed to invoke the text component may be automated by the text conversation user interface.

It shall be possible to establish text-only calls as well as calls with other media components.

5.5 Invoking text conversation

Calls with terminals where text conversation is used shall bemonitored for requests to start the text feature.

When either party in a call begins usage of the text feature, an effort shall be made to establish text conversation. It shall be possible to start text conversation during a call already established in voice or video mode by adding the text component.

The mode selected for text conversation (IP Multimedia, CS Multimedia, Voice path etc.) shall be determined according to a prioritised procedure.

Modes allowing simultaneous voice and text shall by default have higher priority, while the user shall be given opportunities to prioritise text-only modes. The default order shall be: IP Multimedia, CS Multimedia, data path text plus voice, voice path text.

The user shall be given call progress information and text invocation progress information in text or other non-audible media.

5.6 Text conversation handling during calls

Text transmission shall be done character by character as entered, or in small groups transmitted so, that no character is delayed before transmission more than 0.5 seconds. (as stated in ITU-T T.140 [4])

The text transmission shall allow a rate of at least 10 characters per second so that human typing speed as well as speech to text methods of generating conversation text can be supported.

The end-to-end delay of characters shall be less than two seconds, measured between two mobile users, or between one mobile user and any interworking fixed network user, assuming that the fixed network does not contribute with more than one second to this figure.

The character corruption rate should be less than 1% in conditions where users experience the voice transmission quality to be low but useful.

The transmission of the text conversation shall be made according to the character set and operations defined in ITU-T T.140. [4]. This requirement enables smooth interworking with minimal loss of functionality between different text conversation environments. The allowed limitations in character set support specified in ITU-T T.140 shall be obeyed, so that two terminals always have a minimum common character set available for conversation.

Figure 2 gives an example of a possible layout of a real time text conversation according to ITU-T T.140.

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 2: A possible way to display a conversation with one window each

Display of the conversation shall be done according to the principles specified in T.140, unless the display is provided through interworking with a legacy mode text telephone, when the limitations of that device may govern.

NOTE: It shall be noted that support for any written language is possible with T.140, and that many languages require other writing directions than left-to-right.

5.7   Alerting

Since many GTT users are deaf or hard-of-hearing, the terminals used for GTT shall provide an interface that can be used to activate alternative alerting modes e.g. flash or vibration.

5.8   Addressing

GTT shall support the addressing formats of the host environments. to identify the different kinds of endpoints of the call used in the different host environments (see section 9, interworking.)

5.9   Roaming

Access to the GTT service while roaming is desirable.

Note: Operators may consider using a Virtual Home Environment toolkit to provide GTT services to roaming subscribers. It can be expected that the subscriber will need to specifically subscribe to this service and no access to the GTT service will be possible if the visited network does not support the appropriate VHE toolkit. It should also be noted that emergency calls bypass most VHE toolkits, therefore emergency calls will require GTT support in the local network.

6   Security and reliability

The user shall be able to use and access GTT in a secure manner. Within the limitations enforced by the host environments, the following apply.

-   It shall be possible for the contents of GTT sessions to be read only by the intended recipient(s).

-   The "Security Threats and Requirements" specified in 22.133 [3] shall not be compromised.

7   Charging

It is foreseen that users will not be expected to pay a premium for access to the GTT service. Hence the requirement to capture detailed information relating to the use of GTT is seen as optional.

Various charging mechanisms may be supported in addition to what is defined for the host environment, the following charging characteristics may be considered:-- Text was used in the call.- The duration for which text was used in a call.

8 External Interfaces

8.1 External interfaces in the terminal

It shall be possible to control and operate GTT from external digital devices, e.g, portable computers. For this purpose interfaces shall be standardised between the terminal and such external digital devices.

It shall be possible to control and operate GTT from external analogue text telephones. For this purpose, interfaces shall be standardised between the mobile terminal and such external analogue text telephones.

External interfaces specified for controlling and operating GTT between the terminal and an external device should make it possible to use the external device for both Multimedia Messaging and GTT.

8.2 External interfaces to other networks

No new network interfaces shall be specified for GTT, only the necessary elements to accomplish text interworking. These are specified in ITU-T H.248 Annex F, in the Text Conversation and Text Telephone packages [3].

The networks for specified interworking include:

- PSTN for interworking with PSTN text telephones.

- IP networks for interworking with IP conversational multimedia terminals and IP text telephones.

- CS networks for interworking with CS Multimedia terminals.

- Mobile networks out of scope of the GTT specifications interface through common interworking facilities.

9 Interworking

It shall be possible to use GTT to communicate with similar features in other networks as specified in ITU-T H.248 Annex F, in the Text Conversation and Text Telephone packages [3].

This includes:

- PSTN text telephones, using voice telephony as a carrier according ITU-T V.18 [6].

- IP text telephones, using IP multimedia protocols and text according to ITU-T T.140 [4].

- IP conversational multimedia terminals using IP multimedia protocols with text according to ITU-T T.140 [4].

- CS multimedia with text according to ITU-T T.140 [4].

There is no requirement that the host environment combinations for interworking shall be extended for GTT support beyond the host environment interworking established for other media.

For interworking with text conversation in mobile networks outside the scope of the GTT specifications, interworking shall be specified as for other external networks.

For mobile services within scope of the GTT specification, interworking shall be possible between different forms of GTT, within the limits of interworking between the different host environments.

GTT shall permit use of the supplementary services defined for the host environment.

Users of both GTT and Multimedia Messaging [7],[8], should be provided the following interoperation opportunities:

- If a call with a text component fails, it should be possible to offer the user to send a message to a MultiMedia Messaging facility such that the message may later be delivered in non-real time to the destination.

- External User Equipment interface specifications should enable both GTT and MMS text handling functions to be placed in the same device.

Note: In order to fulfil the requirement to give a consistent view of the feature from different access points, it may be of interest to give fixed network text telephone users possibilities to send and receive MMS text messages. This is however outside the scope of this specification.

10 Additional services

10.1 Emergency services

It shall be possible to support emergency service calls with text and voice.

If an emergency service centre supports call back to the calling user, then the call-back function shall be available for GTT text calls without modification to the existing emergency service centre text telephony equipment.

GTT emergency calls shall be possible without the SIM/USIM module in the same host environments as such calls are enabled for voice.

All information that follows a voice call to the emergency centre, shall also be provided to the centre if GTT text is used in the call.

The emergency service requirements shall be possible to fulfil for emergency service centres equipped with PSTN text telephones, operated through a relay operator service or an interworking functions.

Support of GTT service for emergency calls shall not jeopardise normal emergency calls.

10.2 Relay services

Relay services offer translation between text and voice. It is most commonly implemented as manned services.

Access to relay services shall be possible for GTT users. Establishment of the relay service itself is outside the scope of GTT.

GTT shall enable addition of a relay service to the call. The relay service terminates the text path, and translates between text and voice.

Addressing the destination and requesting support from the relay service should be possible to do in one operation from the GTT user.

Other aspects of implementing relay services are outside the scope of this specification.

11 Network implementation considerations

The GTT feature may be implemented in a number of ways. The GTT specifications shall allow cost effective implementations both for networks with a small part of the subscribers using GTT as well as networks with relatively many subscribers using GTT.

Annex A (informative): Background

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. With proper implementation in mobile systems, the feature is expected to be of interest for any user.

The general user needs are described in ETSI ETR 333 Human Factors, Text Telephony; User requirements and Recommendations [2].

In PSTN, seven different, openly specified systems for text telephony exist, and are used in different regions. Some proprietary modes are also used.

The open specifications are called Baudot, DTMF, EDT, V.21, Bell103, Minitel and V.18. They all use different modem technologies and character coding for the transmission of text. They are briefly described in the annexes of ITU-T V.18 [6].

ITU-T V.18 [6] is an automoding mechanism that enables communication with all the legacy modes on the modem level. It also detects when both parties have V.18, and invokes an internationally useful character set, defined in ITU-T T.140, and optionally transmission of text and voice simultaneously. Native modulation for V.18 is V.21, while V.61 is used when simultaneous voice and text is wanted. Other modulations can be negotiated.

V.18 and T.140 are intended for use in new PSTN text telephones. They are also intended for use in gateways, bridging from the fragmented situation in the PSTN, into new services, designed according to the Total Conversation concept.

A.1 Total Conversation

Total Conversation adds text conversation to multimedia protocols in a standardised way, so that simultaneous communication in video, text and voice is accomplished. For the text part, the Unicode based text presentation protocol ITU-T T.140 is used. It is transmitted with a specific standardised transport channel for each environment. Subsets can be used for example for text telephony enabled in the multimedia architectures.

Total Conversation is currently defined in the following environments:

1. Packet networks, where the procedures described in ITU-T H.323 Annex G can be used for text conversation sessions, using TCP or RTP/T140 for the transport of T.140. A simple packet text telephone is defined, called Text SET.

2. Packet networks, where the IETF Session Initiation Protocol SIP can be used for setting up and conducting text conversation sessions using RTP/T140 for the transport of T.140 text media. RTP/T140 is MIME registered and specified in IETF RFC 2793 [9].

3. The H.324 multimedia environment in PSTN, ISDN and Mobile networks, where an AL1 channel connected by H.245 procedures is used for T.140.

4. The H.320 multimedia environment, where a H.224 channel with client ID=2 is specified for transport of T.140.

5. The T.120 data conferencing environment, that can be used alone or in conjunction with any of the environments above, where T.134 specifies the application entity and T.125 the data channel for T.140.

6. Text Telephony in the PSTN using the ITU-T V.18 modem and T.140 for the presentation without any further transport protocol.

A.2 Interworking

Interworking between these forms of Text Conversation can be achieved through the use of gateways or interworking functions. One gateway mechanism *(in the process of standardisation)* is ITU-T H.248, where Annex F, Facsimile, Text conversation and call discrimination packages [3] describes additions for handling text telephony and text conversation.

A.3 Additional services

Apart from the basic user – to – user conversation, the following are examples of important additional services that can be offered text telephone users and Total Conversation users.

Emergency services

The ability to call to emergency services, announce an emergency situation and discuss the actions can be offered text telephone users and Total Conversation users. Calls from the emergency service with support for text conversation is also needed.

Relay services

In order to have conversations with users of plain voice telephones, text relay services are established. Currently, manned relay services dominate, offering real time translation between written and spoken language. Automatic and semi-automatic versions are emerging. A through connection of the voice channel is usually offered as an option to satisfy users who can benefit from multiple modes.

With Total Conversation services, both traditional text relay services and other relay services are established. Examples are sign language relay services with text support during service establishment and speech-support relay services with visual enhancement.;

Annex B (informative):
Initial requirements

This section is informative and specifies the initial requirements that formed the base of definition of GTT. The 3GPP GTT specifications together define what requirements will be met.

Most requirements in this chapter are derived from the requirements expressed to be urgent by the FCC and the CTIA TTY forum in USA. The relation with these requirements are documented for each requirement.

Many of these requirements are written with the view that a PSTN text telephone is attached to the mobile station for implementation of the basic GTT functionality.

B.1 Transmission Performance

A1: The printable character error rate (PER) shall be less than 1% for stationary calls under nominal radio conditions, i.e. where also speech calls show an acceptable quality (TTY/FCC 1).

A2: The printable character error rate (PER) shall be less than 1% for calls with pedestrian speed or typical vehicle speed under nominal radio conditions (TTY/FCC 13, TTY Forum, extended).

A3: An output volume control should be provided in order to allow adaptation to existing Text Telephone equipment for optimal receiving quality (TTY/FCC 4).

A4: The input range shall be automatically adapted to the proper receiving level (new).

A5: The Text Telephone shall allow to transmit Speech and Text (VCO/HCO) in an automated way, without user interaction, either by alternating between speech and text mode or by transmitting speech and text in parallel (objective) (TTY/FCC 9, extended).

A6: The delay of the speech path shall not exceed that of a normal speech call in that situation by more than 40ms in one way, if the text telephone functionality is activated. Otherwise no additional delay shall occur to the speech signal. The delay of the speech path should not vary during a speech phrase.

A7: The delay of the text path shall not exceed that of the speech path by more than 1000 ms (one way) at typing speed of 7 characters per second. The delay of the text path may be variable.

A8: The time of switching between Text and Speech mode shall not exceed 1000 ms at normal typing speed.

A9: The printable character throughput shall not be smaller than 10 characters per second under nominal radio operating conditions (TTY/FCC 10, extended).

A10: Under impaired radio conditions the character throughput may be reduced in order to achieve the desired printable character error rate (TTY/FCC 10, modified).

A11: The signals used for transmission should pass existing fixed and wireless "analogue" speech channels with a printable character error rate below 1% under nominal radio conditions, if two devices are connected "back to back".

A12: The data transmission bit rate should be as high as possible under the given radio conditions with the objective of 300 Bit/s with a bit error rate below 10^{-3} and a residual bit error rate below 10^{-5} (after removing detected bit errors).

A13: It shall be possible to transmit both, single characters, as well as long text or data strings (from file) efficiently.

A14: The transmission mechanism should provide data flow control to adapt the source data rate (when in file transfer mode) to the varying radio transmission conditions, in order not to loose data on the path.

A15: The character coding shall allow transmission of characters from any language in a consistent way.

B.2 Man Machine Interface (MMI)

These Requirements and Objectives are to some extent depending on implementation and <u>not</u> on the transmission <u>standard</u>. This list shall by no means restrict the innovation capabilities of vendors, but give guidelines and define a minimum set, against which the transmission standard needs to be checked.

B1: The Text Telephone user must be able to monitor all aspects of call progress (same or more information as provided to voice users) by tones and visualisation (TTY/FCC 2). The transmission standard shall provide the necessary monitor information to the MMI.

B2: There must be an indication, by tones and visualisation, when the call is connected or disconnected (TTY/FCC 3).

B3: Call information such as caller identification, where provided in mobile voice services, should also be provided for Text Telephony calls (TTY/FCC 11 adapted).

B4: The Text Telephone system must be able to send "Text Tones" to a normal telephone user, to indicate that he is using a text telephone, even if the other user has only a normal phone (TTY/FCC 6 adjusted).

B5: The Text Telephone user must have a means of tactile (vibrating) ring signal indication. (TTY/FCC 5), besides an acoustical and optical ring signal indication (new).

B6: Emergency calls (e.g. to 112 or 911 numbers) shall not require any further user interaction than for any normal voice call. (except to connect the possible additional equipment) (new).

B7: Call back from Emergency Call Centres should not require any further user interaction than for any normal voice call (except to connect the possible additional equipment) (new.

B8: Call setup to and from other Text Telephone users of the same type according to the new standard should not require any further user interaction than for any normal voice call (new).

B9: Call setup to and from other Text Telephone users of an other kind (e.g. those defined in ITU V.18) <u>may</u> require some user activity, like

- sending a short additional precode (e.g. 2..3 digits) or

- typing a short digit sequence to invoke the service (e.g. like #55*) or

- (on user´s preference) the permanent subscription to such a service

- a capability signalling by the mobile

- activating a text telephone application

- others

B10: The new standard should allow the usage of ordinary unmodified mobile phones as already available to the user or on the mass market, in order to get the advantage of high volumes price level.

B10: The new standard shall allow to hide the intermediate (with existing phones) necessary user interaction, as described in the items before, in modern equipment (automatic service).

B11: The use of the Text Telephone for emergency calls with unregistered phones or without SIM/USIM shall be possible as for normal voice calls.

B12: The interface to the possible adapters shall be exactly specified with

- mandatory interfaces (as to allow interaction to existing Text Telephones and the basic access to the mobile phone) and

- optional interfaces (as to connect microphone and loudspeaker, the control port of the mobile phones, …)

B.3 Compatibility

C1: The standard shall be compatible to equipment on the landline side as specified in ITU recommendation V.18 (TTY/FCC 12, extended), including all annexes of V.18 and V.18 with V.61 for voice and text simultaneously

C2: The landline party's Text Telephone equipment shall not need modifications or additions in order to be compatible and to achieve the desired error rate (TTY/FCC 7).

C3: It shall be possible to deploy the Text Telephone standard world wide in all systems based on 3GPP specifications (new).

C4: The wireless Text Telephone (i.e. on the Mobile user side) may require modifications or additions or the development of new equipment (TTY/FCC 8). The smaller the modifications or additions the better.

C5: Roaming between networks of different operators of the same kind of wireless technology shall be possible, provided the operators has installed the service.

C6: Is should be possible to connect equipment implementing the standard to any existing mobile phone of any existing wireless standard.

C7: Communication between Text Telephones in different kinds of wireless technologies shall be possible.

C8: Communication with text between mobile text telephones and multimedia devices with text shall be possible.

B.4 Complexity of Implementation and Roll Out

D1: The standard shall allow a fast first step implementation and roll out

D2: The standard shall allow integration into the mobile phone or the Text Telephone terminal or integration of everything together into a single device.

B.5 Referenced consumer requirements

In the previous sections of this annex, the "TTY /FCC" requirements refer to the points below. They are formulated in November 1999 as the consumer requirements by the TTY Forum administered by CTIA in USA. The comments are inserted here for clarification.

1. The character error rate should approximate that of AMPS, which has been demonstrated at <1% for stationary calls. More research on AMPS performance with TTY would be useful to assist in specifying a range of conditions.

2. The TTY caller must be able to visually monitor all aspects of call progress provided to voice users. Specifically, the ability to pass through sounds on the line to the TTY (so that the user can monitor ring, busy, answered-in-voice, etc.) should be provided.

3. There must be a visual indication when the call has been disconnected.

4. A volume control should be provided.

Comment: This item is intended to allow the TTY user to adjust volume for better reception of TTY tones into a mobile terminal attached TTY. It has no meaning for other solutions.

5. The TTY user must have a means of tactile (vibrating) ring signal indication.

6. The caller must be able to transmit TTY tones independent of the condition of the receiving modem.
Comment: This is to permit Text Telephone signalling, to let a hearing person know that the incoming call is from a text telephone.)

7. The landline party's TTY must not require retrofitting in order to achieve the desired error rate.

8. The wireless party's TTY may require retrofitting, or a new model TTY to be developed, or the use of a portable data terminal such as a personal digital assistant.

9. VCO and HCO should be supported where possible.

Comment: VCO= Voice carry over. HCO= Hearing Carry Over. These are terms for alternating voice and text in the same call.

10. Reduction of throughput (partial rate) on Baudot is highly undesirable and should not be relied upon to achieve compliance (see #7). It may be useful as a user-selectable option to improve accuracy on a given call.

11. Call information such as ANI and ALI, where provided in wireless voice, should also be provided for TTY calls.
Comment: ANI= Automatic Number Identification, ALI= Automatic Location Identification.

12. On the landline side, the solution need not support little-used or obsolete TTY models, but in general should support the embedded base of TTY's sold over the past ten years. The landline equipment supported must not be limited to that used in Public Service Answering Points (911 centers).

13. Drive conditions must be supported, again using AMPS as a benchmark.

Annex C (informative): Change history

Document history		
V. 0.0.1	April 2000	First Draft (Presented at TSG-S1 10th – 15th April 00)
V.0.3.0	July 2000	Includes changes required by S1#9.
V.0.3.1	Aug. 2000	Align with 3GPP doc structure
V.0.3.2	Aug. 2000	Comments from Ericsson and Siemens accepted
V.1.0.0	Sept. 2000	Updated to version 1.0.0 for presentation to SA #9
V.1.1.0	May 2001	Updated with output of GTT ad-hoc. To be sent for approval at SA#12
V.2.0.0	May 2001	Raised to version 2.0.0 for approval at SA #12
V5.0.0	June 2001	Approved at SA #12
V5.1.0	Oct. 2001	Creation of 5.1.0 by inclusion of CRs at SA #13

Change history											
TSG SA#	SA Doc.	SA1 Doc	Spec	CR	Rev	Rel	Cat	Subject/Comment	Old	New	WI
Dec 1999			02.01					Transferred to 3GPP SA1	8.1.0	3.0.0	
SP-13	SP-010433	S1-010861	22.226	001		Rel-5	F	CR to 22.226 version 5.0.0 GTT Stage 1 as requested by SA	5.0.0	5.1.0	GTT
SP-15	SP-020045	S1-020457	22.226	002	-	Rel-5	F	Editorial CR to correct terms and references	5.1.0	5.2.0	CORRECT
SP-26	SP-040744	S1-040997	22.001			Rel-6		Updated from Rel-5 to Rel-6	5.0.0	6.0.0	
SP-36			22.001			Rel-7		Updated from Rel-6 to Rel-7	6.0.0	7.0.0	
SP-42	-	-				Rel-8		Updated from Rel-7 to Rel-8	7.0.0	8.0.0	
SP-46	-	-	-	-	-	-	-	Updated to Rel-9 by MCC	8.0.0	9.0.0	
2011-03	-	-	-	-	-	-	-	Update to Rel-10 version (MCC)	9.0.0	10.0.0	
2012-09	-	-	-	-	-	-	-	Updated to Rel-11 by MCC	10.0.0	11.0.0	
2014-10								Updated to Rel-12 by MCC	11.0.0	12.0.0	
2015-12	-	-	-	-	-	-	-	Updated to Rel-13 by MCC	12.0.0	13.0.0	
2017-03	-	-	-	-	-	-	-	Updated to Rel-14 by MCC	13.0.0	14.0.0	
2018-06	-	-	-	-	-	-	-	Updated to Rel-15 by MCC	14.0.0	15.0.0	
SA#88e	-	-	-	-	-	-	-	Updated to Rel-16 by MCC	15.0.0	16.0.0	

History

Document history		
V16.0.0	August 2020	Publication