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This document proposes modifications to the stage 2 description for Global Text Telephony.

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Technical Specification

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Contents

Foreword

This Technical Specification has been produced by the 3GPP.

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Introduction

Real time, character by character text conversation is a component that can be of value in a distant conversation. Users may have an interest to use real time text conversation alone or in any combination with voice and video.

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

GTT is defined in a set of host environments, circuit switched as well as packet switched.

Interworking with corresponding features in other networks is an important part of Global Text Telephony. 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 consideration.

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 3GPP 22.226, Global Text Telephony, Stage 1, Annex A, [3]

1 Scope

This 3GPP Technical Specification defines the stage 2 description of the real time Text Conversation Feature called Global Text Telephony, GTT. GTT. Stage 2 identifies the functional capabilities needed to support the service described in GTT Stage 1.

This TS contains the core ~~transcoder~~ functions for a real time Text Conversation Feature GTT, to be used in combination with other media in conversational services.

GTT offers real time conversation in text, to be used alone or in combination with other conversational media, and interworking with current and emerging text conversation features in the fixed networks and other mobile networks.

GTT uses a number of functional entities to realise the requirements of the stage 1 description (3G TS 22.226). This TS describes how the service requirements are realised with these functional entities. As far as possible existing protocols shall be used for the realisation of the Global Text Telephony Feature. This may include e.g. , SIP, 3G.324, or Circuit Switched Voice service as protocol environments, and CTM, AL1 and RTP/text as transmission protocols. It also means usage of existing text presentation format ITU-T T.140, common to all GTT text conversation environments..

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.

- A non-specific reference to an ETS shall also be taken to refer to later versions published as an EN with the same number.

[1] 3G TS 22.101: “Service Principles”

[2] 3G TS 22.121: “The Virtual home Environment”

[3] 3G TS 22.226: "Global Text Telephony " Stage 1 service description.

[4] ITU-T V.18 Operational procedures for modems in the text telephone mode.

[5] ITU-T T.140Text conversation presentation protocol

[6] 3GPP 26.110 Codec for 3G CS Multimedia (2000)

[7] ITU-T H.323 Annex G, Text conversation and text SET (2000)

[8] ITU-T H.224 (2000) Very low bitrate multimedia system

[9] ITU-T H.248 Annex F, Facsimile, Text Conversation and Call Discrimination Packages

[10] IETF RFC 2793 RTP-Text. RTP Payload for Text Conversation.

[11] IETF SIP, Session Initiation Protocol

[12] 3G TS 26.226 CTM Cellular Text telephony Modem, General description

[13] ITU-T F.703 Multimedia Conversation Service Description (2000)

[14] 3GPP TS 26.235 Codec for packet switched conversation

3 Definitions and abbreviations

3.1 Definitions

Total Conversation A service offering standardised simultaneous text, video and voice conversation or a subset thereof. (see ITU-T F.703)

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.

3.2 Abbreviations

For the purposes of this document the following abbreviations apply in addition to those defined in [3]:

GTT Global Text Telephony

CTM Cellular Text telephone Modem, as specified in 3Gpp 26.226 [12]

4 General Functions

4.1 Overview

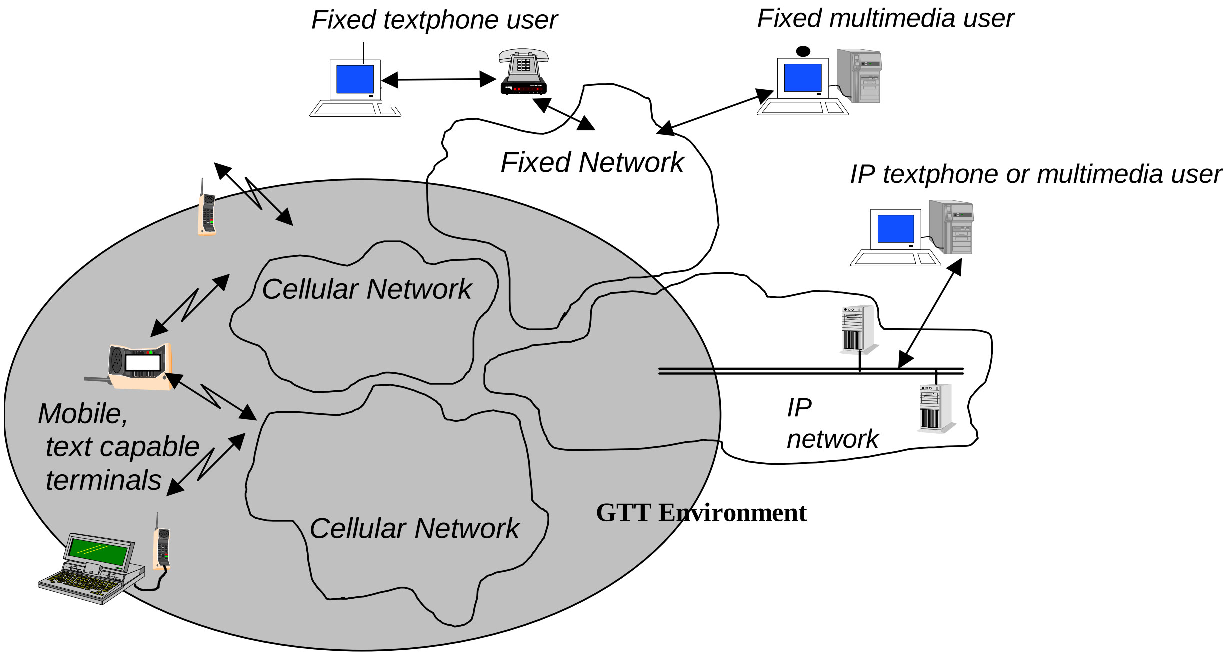

Figure 1: General view of GTT feature provision within the different networks

Figure 1 shows a generalised view of the Global Text Telephony feature architecture for a third generation conversation service system. It shall combine different networks and network types and shall integrate text conversation systems already existent within these networks.

Global Text Telephony is a 3GPP Feature that can be included in 3GPP conversation services such as circuit switched voice telephony, circuit switched multimedia and IP multimedia conversation. It makes use of the infrastructure of the host environment and includes elements necessary for GTT in these environments.

The following description and figures describe the text specific parts and leave out many details of the host environment conversation service where text is included.

4.1.1 General text transmission functions

On the route from user to user, the text passes a set of functional elements.

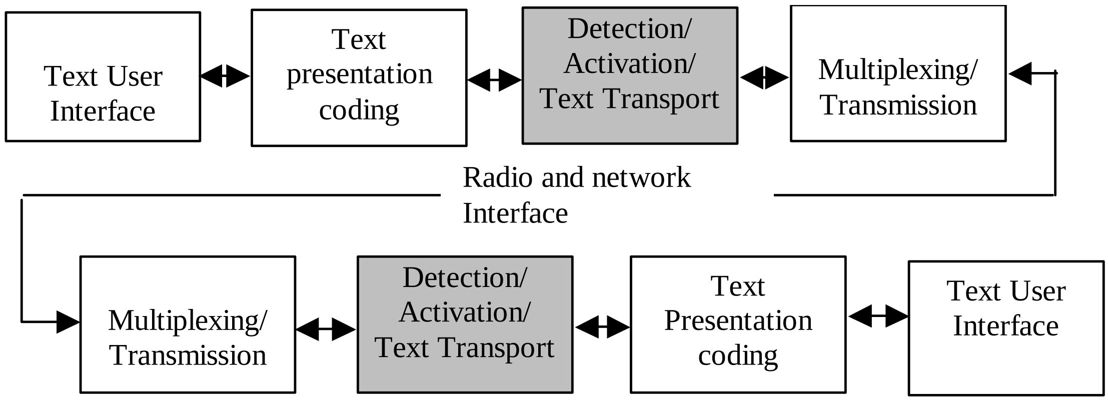

Figure 2, call flow for two mobile terminals with the same GTT transport method, using text conversation.

Figure 2 shows the function chain when communicating within the network, between terminals operating in the same mode.

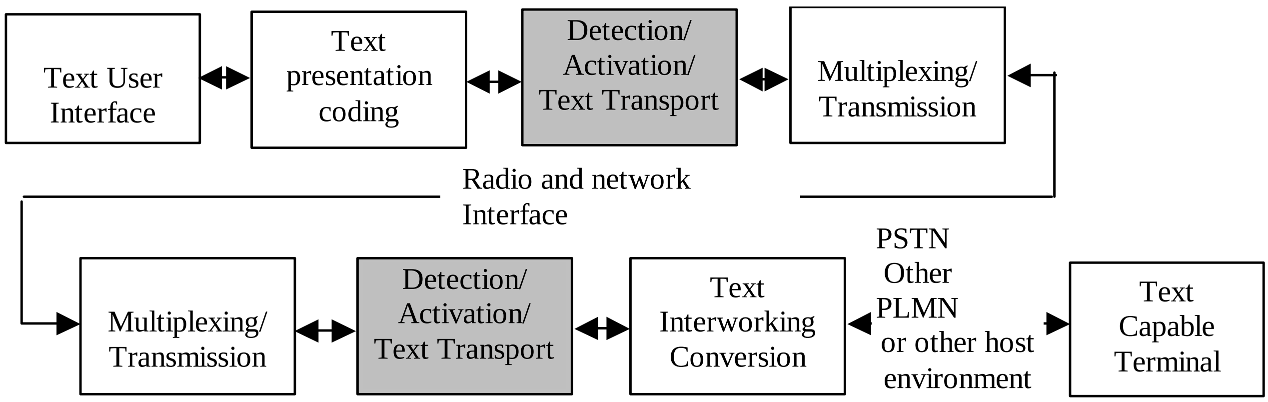

Figure 3 : GTT general text transmission functions showing interworking

The view in figure 3 is applicable to transport mechanisms and network environments when communicating in interworking mode between different environments.

4.1.2 Functions

4.1.2.1 Call Control and text media initiation

Before text conversation can begin, a call must be established. That is done with general call control functions of the host environment. ~~The desire of the user to use text in a call can either be indicated from the beginning of the call, or indicated~~ during the~~a~~ call ~~and nothing occurs at the beginning of the call.~~ GTT Elements detect the ~~desire to establish a~~ text ~~channel~~in the call, select a suitable transport mechanism, and activate transmission functions.

4.1.2.2 Presentation coding

The text in GTT shall be coded in a common presentation protocol, T.140 [7]. If necessary this presentation protocol shall be converted to or from any legacy mode character code used in other networks.

4.1.2.3 Transport and transmission host environments

When text transmission is activated, a suitable transmission method in the PLMN is selected. The appropriate method to use is selected according to the call environment. The environments valid for GTT are called GTT Host environments.:

1. IP Multimedia, according to IPMM subsystem with IETF SIP [11].
2. Circuit Switched Multimedia according to 3G.324.
3. Circuit switched voice channel.

Other methods to establish real time text conversation exist and may be used without further standardisation, using basic communication services of the PLMN. One example:

> Digital data transmission in a data channel can be used for real time text conversation.. For cases when only text conversation is wanted, there are a multitude of ways to implement that kind of communication. It can for example be done through a HTML based server, with an already established mainstream mechanisms that do not require text conversation specific functions to be stored in the terminals.

4.1.2.4 Multiplexing

The text transport is multiplexed in the network and radio interface according to normal procedures for the selected host environment.

4.1.2.5 Conversion.

For text conversation with text telephones and text capable terminals in different networks or using different transport mechanisms, conversion functions may be used. Functions and procedures suitable for the conversion functions are described in ITU-T H.248 Annex F, packages for Text telephony, Text Conversation and Call Type Discrimination [6].

4.1.2.6 Text capable terminal

By using the described GTT functions, a real time text conversation session can be conducted between GTT supported mobile Text Capable Terminals.. Different terminal function combinations and GTT host environments give different opportunities regarding combinations of text with voice and video. Valid combinations are:
text only,
alternating text and voice,
simultaneous text and voice,
simultaneous text and video,
simultaneous text, video and voice.

~~4.1.2.7~~4.1.2.7 Routing and location of functions

The detection/transcoding and transport function is carried out within the transcoder function. Thus for GSM it is in the transcoders in the BSS. For UMTS, it is in the transcoder function in the core network. Since the transcoder is present in every speech call, no specific routing is required.

~~Routing functions make sure that calls between different host environments where text functionality may be used are routed through GTT elements for conversion. Routing functions are also used to make sure that calls between PSTN Text telephones and GTT-Voice terminals get the required conversion. Routing can for example be based on teleservice or subscriber data.~~

4.1.3 Emergency service considerations

If an operator implementing GTT selects to offer access to Emergency Services through this feature for a specific host environment, the following must be considered.

If the emergency services only has limited types of text conversation devices; conversion from the users host environment to the one used by the emergency service may be configured.

If the calling party address and location information are provided in voice emergency calls, this information must be preserved also in text emergency calls, and not changed by any conversion or routing mechanisms introduced.

Other host environment specific considerations for emergency calls are described in sections below.

5 Considerations for each host environment

5.1 GTT-IP IP Multimedia

IP Multimedia, supported by the IPMM subsystem, is a suitable environment for real time text conversation. It shall use IETF SIP [11], with text coded according to ITU-T T.140 and transported with IETF RTP-text [10] as indicated in 3G TS 26.235 [14]. This allows conversation in a selection of simultaneous media, such as text, video and voice.

Inclusion of the text conversation shall be done according to normal SIP and IPMM procedures, where the text media stream is handled as any other media. GTT-IP has no architecture influence on the 3G network, only that the components must allow handling of the standardised text media stream.

Note: This way of using mainstream procedures opens possibilities to utilize the flexibility of the SIP protocol for enhanced services. The user can interact in the service offering to optimise the stream handling. This may be used for flexible ways of invoking relay services for media conversion according to the desire of the users.

5.2 GTT-CS Circuit switched Multimedia

Text conversation in Circuit Switched Multimedia is called GTT-CS. The host environment is 3G.324, according to 3GPP TS 26.110.Codec for circuit switched multimedia [6].GTT-CS Text is T.140 coded and transported in an AL1 channel. Any combination of Video, Text and Voice can be supported and used simultaneously.

GTT-CS has no architecture implications on the 3G network, only that the network components must allow the standardised handling of the text media channel as well as any other media channel.

5.3 GTT-Voice Circuit switched voice channel

Voice channel transmission of text shall use CTM; Cellular Text telephone Modem [12]. It is possible to alternate between text and voice. Text shall be coded according to ITU-T T.140 on the presentation level.

5.3.1 Interworking between GTT-Voice and PSTN Text Telephony

If GTT-Voice is provided, interworking to PSTN text telephony can be provided by introducing conversion in the PLMN between CTM and PSTN based text telephony using ITU-T V.18 or any of its specific sub-modes.

A simple extension can be made of the conversion functionality described in H.248 annex F. CTM is included among the transport mechanisms for an extention of package txc.

For the CTM and PSTN text telephone case, a conversion function can be seen as a context containing a CTM termination and a V.18 text telephony termination. This combination is called the CTM channel. It is foreseen that the CTM channel needs control to be initiated to the proper state for each call etc. Such functions are here collected in an entity called interworking control.

~~CTM channels shall be clustered in CTM service nodes, and routing principles implemented to route the calls through the node.~~ The CTM channels are normally transparent but monitoring for text telephone signals or CTM signals. When a signal is discovered, the audio path is stopped and character conversion and transmission performed.

It can be symbolically documented as follows.

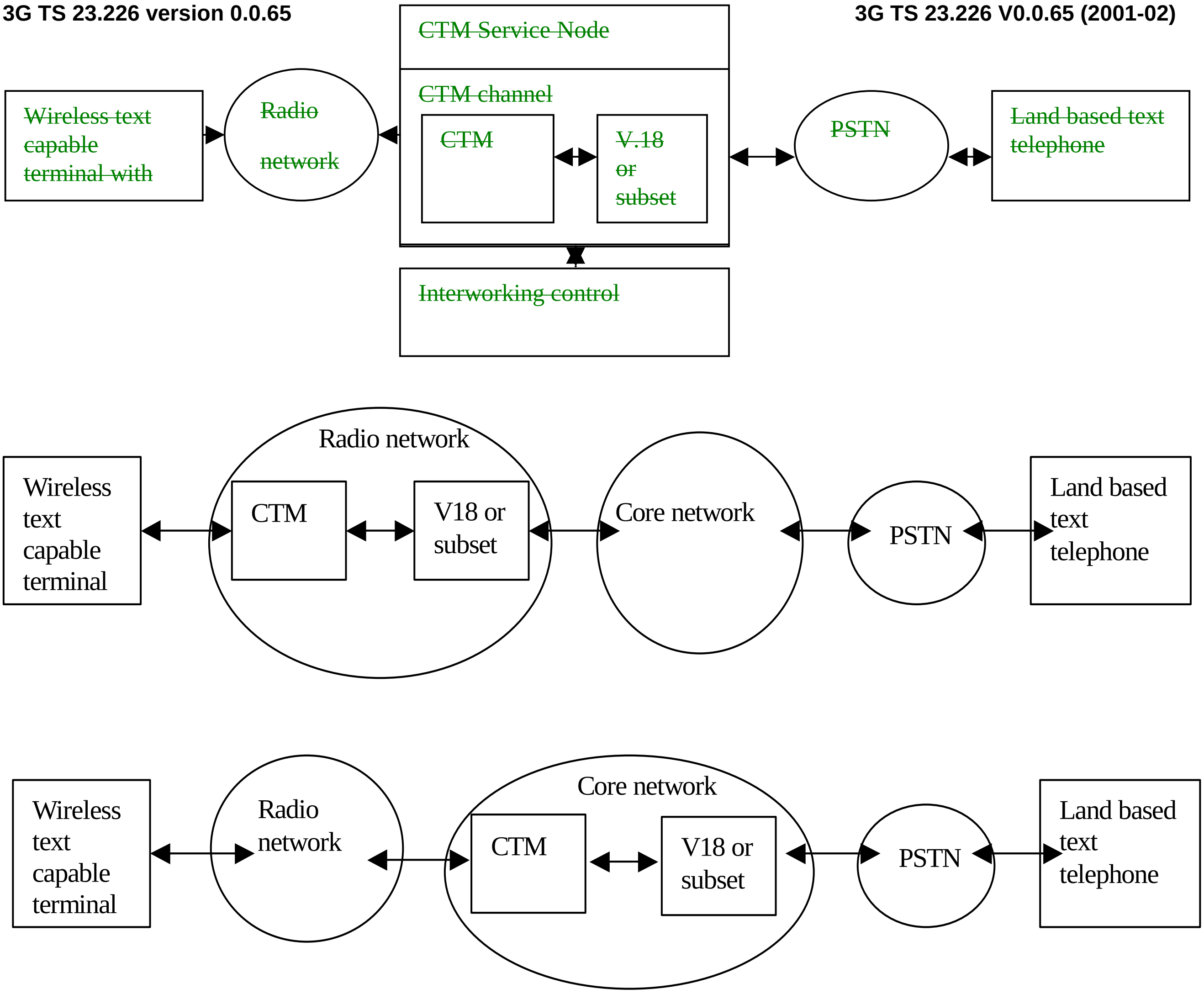

Figure 4. Interworking with PSTN

Since the PSTN text telephone protocols have states, the CTM channel should be placed so that the state will be maintained regardless of handover.situations. Resetting the state during the call should be avoided because it can cause loss or corruption of characters or loss of communication.

~~Already standardised routing and invocation mechanisms should be used to invoke CTM channels in the calls that may require text conversation. A specific CAMEL based procedure is specified for the general user to user calls~~.

The mechanisms for invocation of the CTM channel act on both Mobile Originating calls and Mobile Terminated calls.

~~The CTM service node should be positioned functionality in front of the Media Gateway or in other ways so that the audio is not coded for the wireless environment when it is brought to the text telephone modem in the CTM channel.~~

5.3.2 Emergency call considerations.

It may be required to let any user use any phone for a text call to the emergency service without any additional configuration option due to CTM presence. ~~To meet this requirement, one configuration option shall be to identify all emergency calls as a potential text calls, so that the calls can be routed through CTM channels~~.

It may be required to handle emergency calls also when the call comes from a SIM-less phone or a phone with an invalid subscription without needing any additionnal configuration option due to CTM presence. ~~To meet this requirement, one configuration option shall be to route emergency calls independent of subscription status through the CTM channel.~~

~~Thus one option for routing calls through the CTM channel,shall be to route based on the emergency call teleservice indication~~

History

Document history		
V. 0.0.2	May 2000	Initial Draft
V.0.0.3	Aug 7, 2000	Changes according to evolving total R´2000 architecture
V 0.0.4	Sep 1, 2000	Improved structure and detailing of some procedures
V.0.0.5	Feb 21, 2001	Alignment with R-4 and R-5 architecture

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