 Framework of requirements for real-time text conversation using SIP.

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Abstract

   This document provides the framework of requirements for text
   conversation with real time character-by-character interactive
   flow over the IP network using the Session Initiation Protocol.
   The requirements for general real-time text-over-IP telephony,
   point-to point and conference calls, transcoding, relay services,
   user mobility, interworking between text-over-IP telephony and
   existing text-telephony, and some special features including
   instant messaging have been described.

Table of Contents

1. **Introduction**

     Text-over-IP (ToIP) is becoming popular as a part of total
     conversation among a range of users although this medium of
     communications may be the most convenient to certain categories of
     people (e.g., deaf, hard of hearing and speech-impaired
     individuals). The Session Initiation Protocol (SIP) has become the
     protocol of choice for control of Multimedia IP telephony and
     Voice-over-IP (VoIP) communications. Naturally, it has become
     essential to define the requirements for how ToIP can be used with

SIP to allow text conversations as an equivalent to voice. This
document defines the framework of requirements for using ToIP,
either by itself or as a part of total conversation using SIP for
session control.

2. Scope

The primary scope of this document is to define the requirements
for using ToIP with SIP, either stand-alone or as a part of a
total conversation approach. In general, the scope of the
requirements is:

a. Features in Real-Time ToIP
b. Real-time Multimedia Conversational Sessions using SIP
c. General Requirements for Real-Time ToIP using SIP
d. Interworking Requirements for ToIP
e. Text gateways to interconnect the different networks

The subsequent sections describe those requirements in detail.

3. Terminology

The key words "MUST", "MUST NOT", "REQUIRED", "SHALL", "SHALL
NOT", "SHOULD", "SHOULD NOT", "RECOMMENDED", "MAY", and "OPTIONAL"
in this document are to be interpreted as described in RFC 2119
[2].

4. Definitions

Audio bridging - a function of a gateway or relay service that
enables an audio path through the service between the users
involved in the call.

Full duplex - user information is sent independently in both
directions.

Half duplex - user information can only be sent in one direction
at a time or, if an attempt to send information in both directions
is made, errors can be introduced into the user information.

Interactive text - a term for real time transmission of text in a
character-by-character fashion for use in conversational services.

TTY - name for text telephone, often used in USA, see textphone.
Also called TDD, Telecommunication Device for the Deaf.

Textphone - text telephone. A terminal device that allow end-to-end real time text communication. A variety of textphone protocols exists world-wide, both in the PSTN and other networks. A textphone can often be combined with a voice telephone, or include voice communication functions for simultaneous or alternating use of text and voice in a call.

Text bridging - a function of a gateway or relay service that enables the flow of text through the service between the users involved in the call.

Text gateway - a multi functional gateway that is able to transcode between different forms of text transport methods. E.g. Between ToIP in IP networks and Baudot text telephony in the PSTN.

Text telephony - Analog textphone services

Text Relay Service - A third-party or intermediary that enables communications between deaf, hard of hearing and speech-impaired people, and voice telephone users by translating between voice and text in a call.

Transcoding Services - Services of a third-party user agent (human or automated) that transcodes one stream into another.

Total Conversation - A multimedia service offering real time conversation in video, text and voice according to interoperable standards. All media flow in real time. Further defined in ITU-T F.703 Multimedia conversational services description.

Video Relay Service - A service that enables communications between deaf and hard of hearing people with total conversation devices, and hearing persons with voice telephones by translating between sign language and spoken language in a call.

Acronyms:

2G   Second generation cellular (mobile)
2.5G   Enhanced second generation cellular (mobile)
3G   Third generation cellular (mobile)
CDMA   Code Division Multiple Access
CTM   Cellular Text Telephone Modem
GSM   Global System of Mobile Communication
ISDN   Integrated Services Digital Network
ITU-T International Telecommunications Union-Telecommunications standardisation Sector

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PSTN    Public Switched Telephone Network
SIP     Session Initiation Protocol
TDD     Telecommunication Device for the Deaf
TDMA    Time Division Multiple Access
ToIP    Text over Internet Protocol
UTF-8   Universal Transfer Format-8
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5. **Background and General Requirements**

The main purpose of this document is to provide a set of
requirements for real-time text conversation over the IP network
using the Session Initiation Protocol (SIP) [3]. The overall
requirement is that real-time text conversation can be part of a
conversational service like any other media. Participants can
negotiate all media including real-time text conversation[4, 5].
This is a highly desirable function for all IP telephony users,
and essential for deaf, hard of hearing, or speech impaired people
who have limited or no use of the audio path of the call.

It is important to understand that real-time text conversations
are significantly different from other text based communications
like email or instant messaging. Real-time text conversations
deliver an equivalent mode to voice conversations by providing
transmission of text character by character as it is entered, so
that the conversation can be followed closely and immediate
interaction take place, therefore providing the same mode of
interaction as voice telephony does. Store-and-forward systems
like email or messaging on mobile networks or non-streaming
systems like instant messaging are unable to provide that
functionality.

One particular application where real-time text is absolutely
essential, is the use of relay services between conversational
modes, like between text and voice.

Direct text emergency service calls, where time and continuous
connection are of the essence, is another essential application.

6. **Features in Real-time Text-over-IP**

While real-time Text-over-IP will be used for a wide variety of
services, an important field of application will be to provide a
text equivalent to voice conversation, in particular for deaf,
hard of hearing and speech-impaired users.
As such, it is crucial that the conversational nature of this
service is maintained. Text based communications exist in a

variety of forms, some non-conversational (SMS, text paging, E-mail, newsgroups, message boards, etc.), others conversational (TTY/TDD, Textphone, etc).

Real-time Text-over-IP will sometimes be used in conjunction with a relay service [I] to allow text users to communicate with voice users. With relay services, it is crucial that text characters are sent as soon as possible after they are entered. While buffering MAY be done to improve efficiency, the delays SHOULD be kept as small as possible. In particular, buffering of whole lines of text MUST NOT be used.

In order to make Real-Time Text-over-IP the equivalent of what voice is to hearing people, it needs to offer equivalent features in terms of conversation as voice communications provides to hearing people. To achieve that, real-time Text-over-IP MUST:

 a. Offer Real-Time presentation of the conversation. This means that text MUST be sent as soon as available, or with very small delays. The delay MUST not be longer than 300 milliseconds,

 b. Provide simultaneous transmission in both directions,

 c. Provide interoperability with text conversation features in other networks, e.g. PSTN, accepting functional limitations that this will lead to during interoperation.

 d. Support a transmission rate of at least 30 characters/second.

 e. Support suitable reliability of text transmission. A character error rate of 0.2% is regarded good, and 1% usable.

 f. Be possible to merge with video and voice transmission.

 g. The end-to-end delay in transmission MUST be less than 2000 milliseconds.

Many users will want to use multiple modes of communication during the conversation, either at the same time or by switching between modes e.g. between real-time Text-over-IP and voice. Native real-time Text-over-IP systems MUST support simultaneous use of modalities so that the text interface is always available.

When communicating via a gateway to other networks and protocols, the system MUST completely support the functionality for

alternating or simultaneous modalities as offered by the gateway.

When voice is supported on the terminal, the terminal MUST provide volume control.

7. Real-Time Multimedia Conversational Sessions using SIP

The Session Initiation Protocol (SIP) [3] provides mechanisms for creating, modifying, and terminating sessions for real-time conversation with one or more participants using any combination of media: Text, Video and Audio. However, participants are allowed to negotiate on a set of compatible media types (e.g., Text, Video, Audio) with session descriptions used in SIP invitations.

The standardized T.140 real-time text conversation [4], in addition to audio and video communications, will be valuable services to many. Real-time text can be expressed as a part of the session description in SIP and will be a useful subset of the Total Conversation (e.g., Real-time text, Video and Audio).

This specification describes the framework for using the T.140 text conversation in SIP as a part of the multimedia session establishment in real-time over a SIP network.

The session establishment using SIP defines procedures for how T.140 text conversation can be supported using the text/t140RTP payload defined in RFC 2793 [5]. The performance characteristics of T.140 will be determined using RTCP.

The session will not only define procedures between the SIP devices having text conversation capability, but will also define how sessions in SIP can be established between the text conversation and audio/video/text capable devices transparently.

If there is any incompatibility between the terminals, e.g. T.140 only and audio-only terminals, the necessary transcoding services will need to be invoked. This important service feature offers a variety of rich capabilities in the transcoding server. For example, speech-to-text (STT), text-to-speech (TTS), text bridging after conversion from speech, audio bridging after conversion from text, and other services can also be provided by the transcoding and/or translation server. The session description protocol (SDP) [6] used in SIP to describe the session also needs to be capable of expressing these attributes of the session (e.g., uniqueness in

media mapping for conversion from one media to another for each

communicating party).

Real-time text can also be presented in conjunction with video and audio. Making real-time text part of total conversation.

Visual and/or Tactile alerting for T.140 capable terminals should to be provided.

Users may set up text conversation sessions using SIP from any location. In addition, user privacy and security MUST be provided for text conversation sessions at least equal to that for voice.

The transcoding/translation services can be invoked in SIP using different session establishment models [7]: Third party call control [8] and Conference Bridge model [9].

Both point-to-point and multipoint communication need to be defined for the session establishment using T.140 text conversation. In addition, the interworking between T.140 text conversation and text telephony conversation [10] is needed.

The general requirements for real-time text conversation using SIP can be described as follows:

a. Session setup, modification and teardown procedures for point-to-point and multimedia calls

b. Registration procedures and address resolutions

c. Registration of user preferences

d. Negotiation procedures for device capabilities

e. Discovery and invocation of transcoding/translation services between the media in the call

f. Different session establishment models for transcoding/translation services invocation: Third party call control and Conference bridge model

g. Uniqueness in media mapping to be used in the session for conversion from one media to another by the transcoding/translation server for each communicating party

h. Media bridging services for T.140 real-time text, audio, and video for multipoint communications

i. Transparent session setup, modification, and teardown between text conversation capable and voice/video capable devices

j. Conversations to be carried out using T.140-over-RTP and RTCP will provide performance report for T.140

k. Altering capability using text conversation during the session establishment

l. T.140 real-time text presentation mixing with voice and video

m. T.140 real-time text conversation sessions using SIP, allowing users to move from one place to another

n. UsersÆ privacy and security for sessions setup, modification, and teardown as well as for media transfer

o. Interoperability between T.140 conversations and analogue text telephones

p. Routing of emergency calls according to national or regional policy.

[8](#). General Requirements for Real-Time Text-over-IP using SIP

The communications environments for ToIP using SIP to set up the conversation in real-time may vary from a simple point-to-point call to multipoint calls in addition to the fact that ToIP can be used in combination with other media like audio and video. In order to establish the session in real-time, the communicating parties SHOULD be provided with experiences like those of normal telephony call setup. There may also be some need for pre-call setup e.g. storing registration information in the SIP registrar to provide information about how a user can be contacted. This will allow calls to be set up rapidly and with proper addressing.

Similarly, there are requirements that need to be satisfied during call set up when another media is preferred by a user. For instance, some users may prefer to use audio while others want to use text as their preferred choice of conversational mode. In this case, transcoding services will need to be invoked for text-to-speech (TTS) and speech-to-text (STT). The requirements for transcoding services need to be negotiated in real-time to set up the session.

The subsequent subsections describe those requirements in great detail.

[8.1](#) Pre-Call Requirements

The desire of the users for using ToIP as a medium of

communications can be expressed during registration time. Two
situations need to be considered in the pre-call setup
environment:

a. User Preferences: It MUST be possible for a user to indicate a
preference for ToIP by registering that preference in a SIP
server. If the user is called by other party, preferences can be

invoked by the SIP server to accept or reject the call based on
the rules defined by the user. If the rules require that a
transcoding server is needed, the call can be re-directed or
handled accordingly.

b. Server to support User Preferences: SIP servers MUST have the
capability to act on users preferences for ToIP, based on the
usersÆ preferences defined during the pre-call setup registration
time.

8.2 Basic Point-to-Point Call Requirements

The point-to-point call will take place between two parties. The
requirements are described in subsequent sub-sections. They assume
that one or both of the communicating parties will indicate ToIP
as a possible or preferred medium for conversation using SIP in
the session setup.

8.2.1 General Requirements

The general requirements are that ToIP will be chosen from the
available media as the preferred means of communication for the
session. However, there may be a need to invoke some underlying
capabilities in some cases, for example, a transcoding server may
be invoked if one of the users want to use a communication medium
other than ToIP.
The following features MAY need to be involved to facilitate the
session establishment using ToIP as another medium:

a. Caller Preferences: SIP headers (e.g., Contact) can be used to
show that ToIP is the medium of choice for communications.

b. Called Party Preferences: The called party being passive can
formulate a clear rule indicating how a call should be handled
either using ToIP as a preferred medium or not, and whether a
designated SIP proxy needs to handle this call or it is handled in
the SIP user agent (UA).

c. SIP Server support for User Preferences: SIP servers can also

handle the incoming calls in accordance to preferences expressed
for ToIP. The SIP Server can also enforce ToIP policy rules for
communications (e.g., use of the transcoding server for ToIP).

8.2.2 **Session Setup**

Users will set up a session by identifying the remote party or the
service they will want to connect to. However, conversations could
be started using a mode other than real-time Text-over-IP. For
instance, the conversation might be established using voice and
the user could elect to switch to text, or add text, during the
conversation. Systems supporting real-time Text-over-IP MUST allow
users to select any of the supported conversation modes at any
time, including mid-conversation.

Systems SHOULD allow the user to specify a preferred mode of
communication, with the ability to fall back to alternatives that
the user has indicated are acceptable.

If the user requests simultaneous use of text and voice, and this
is not possible either because the system only supports alternate
modalities or because of resource management on the network, the
system MUST try to establish a text-only communication. and the
user MUST be informed of this change throughout the process,
either in text or in a combination of modalities that MUST include
text.

Session setup, especially through gateways to other networks, MAY
require the use of specially formatted addresses or other
mechanisms for invoking gateways.
Such mechanisms MUST be supported by the terminal.

8.2.3 **Addressing**

The SIP [3] addressing schemes MUST be used for all entities. For
example SIP URL and Tel URL will be used for caller, called party,
user devices, and servers (e.g., SIP server, Transcoding server).

The right to include a transcoding service MUST NOT require user
registration in any specific SIP registrar, but MAY require
authorisation of the SIP registrar in the service.

8.2.4 **Alerting**

Systems supporting real-time Text-over-IP MUST have an alerting
method (e.g., for incoming calls) that can be used by deaf and

hard of hearing people or provide a range of alternative, but
equivalent, alerting methods that are suitable for all users,
regardless of their abilities and preferences.

It should be noted that general alerting systems exist, and one
common interface for triggering the alerting action is a contact
closure between two conductors.

Among the alerting options are alerting on the user equipment and
specific alerting user agents registered to the same registrar as
the main user agent.

If present, identification of the originating party (for example
in the form of a URL or CLI) MUST be clearly presented to the user
in a form suitable for the user BEFORE answering the request. When
the invitation to initiate a conversation involving real-time
Text-over-IP originates from a gateway, this MAY be signalled to
the user.

8.2.5 Call Negotiations

The Session Description Protocol (SDP) used in SIP [3] provides
the capabilities to indicate ToIP as a media in the call setup.
RFC 2793 [5] provides the RTP payload type text/t140 for support
of ToIP which can be indicated in the SDP as a part of SDP INVITE,
OK and SIP/200/ACK for media negotiations. In addition, SIPÆs
offer/answer model can also be used in conjunction with other
capabilities including the use of a transcoding server for
enhanced call negotiations [7,8,9].

8.2.6 Answering

Systems SHOULD provide a best-effort approach to answering
invitations for session set-up and users should be kept informed
at all times about the progress of session establishment. On all
systems that both inform users of session status and support real-
time Text-over-IP, this information MUST be available in text, and
may be provided in other visual media.

8.2.6.1 Answering Machine

Systems for real-time Text-over-IP MAY support an auto-answer
function, equivalent to answering machines on telephony networks.
If an answering machine function is supported, it MUST support at
least 160 characters for the greeting message. It MUST support
incoming text message storage of a minimum of 16000 characters,

although systems MAY support much larger storage.

When the answering machine is activated, user alerting MUST still take place. The user MUST be allowed to monitor the auto-answer progress and MUST be allowed to intervene during any stage of the answering machine and take control of the session.

8.2.7 Session progress and status presentation

During a conversation that includes real-time Text-over-IP, status and session progress information MUST be provided in text. That information MUST be equivalent to session progress information delivered in any other format, for example audio. Users MUST be able to manage the session and perform all session control functions based on the textual session progress information.

The user MUST be informed of any change in modalities.

Session progress information MUST use simple language as much as possible so that it can be understood by as many users as possible.
The use of jargon or ambiguous terminology SHOULD be avoided at all times. It is RECOMMENDED to let text information be used together with icons symbolising the items to be reported.

There MUST be a clear indication, both visually as well as audibly whenever a session gets connected and disconnected. The user

should never be in doubt as to what the status of the connection is, even if he/she is not able to use audio feedback or vision.

8.2.8 Actions During Calls

Certain actions need to be performed for the ToIP conversation during the call and these actions are describe briefly as follows:

a. Text transmission SHALL be done character by character as entered, or in small groups transmitted so that no character is delayed between entry and transmission by more than 300 milliseconds.
b. The text transmission SHALL allow a rate of at least 30 characters per second so that human typing speed as well as speech to text methods of generating conversation text can be supported.

c. After text connection is established, the mean end-to-end delay of characters SHALL be less than two seconds, measured between two

ToIP users. This requirement is valid as long as the text input
rate is lower or equal to the text reception and display rate.

d. The character corruption rate SHALL be less than 1% in
conditions where users experience the quality of voice
transmission to be low but useable. This is in accordance with
ITU-T F.700 Annex A.3 quality level T1.

e. When interoperability functions are invoked, there may be a
need for intermediate storage of characters before transmission to
a device receiving slower than the typing speed of the sender.
Such temporary storage SHALL be dimensioned to adjust for
receiving at 30 characters per second and transmitting at 6
characters per second during at least 4 minutes [less than 3k
characters].

f. If text is detected to be missing after transmission, there
SHALL be an indication in the text marking the loss.

g. When used from a terminal designed for PSTN text telephony, or
in interworking with such a terminal, ToIP shall enable
alternating between text and voice in a similar manner as the PSTN
text telephone handles this mode of operation. (This mode is often
called VCO/HCO in USA).

h. The transmission of the text conversation SHALL be made
according to an internationally suitable character set and control
protocol for text conversation as specified in ITU-T T.140.

i. When display of the conversation on end user equipment is
included in the design, display of the dialogue SHALL be made so
that it is easy to read text belonging to each party in the
conversation.

8.2.8.1 Text and other Media Handling Between ToIP Devices

The ToIP devices do not need transcoding from speech to text and
can communicate directly using text/t140. The following
requirements are valid for media handling during calls:

a. When used between terminals designed for ToIP, it SHALL be
possible to send and receive text simultaneously with the other
media (text, audio and/or video) supported by the same terminals.

b. When used between terminals designed for ToIP, it SHALL be
possible to send and receive text simultaneously.

c. It should be possible to know during the call that ToIP is
available, even if it is not invoked at call setup (only voice
and/or video is used). To disable this, the user must disable the
use of ToIP.

8.2.8.2 **General Actions**

a. It SHALL be possible to establish a session with text
capabilities enabled at the beginning of a Call. Note: a call is
in this document defined as one or more sessions).

b. It SHALL be possible to place a call without text capabilities,
and to add text capabilities later in the call.

c. It SHALL be possible to transfer text at at least 30 characters
per second

d. It SHALL be possible to talk and listen simultaneously with
typing and reading.

8.2.8.3 **Call Action with Native ToIP Devices**

a. It SHOULD be possible to answer a call with text capabilities
enabled.

b. It SHOULD be possible to use video simultaneously with the
other media in the call.

c. It SHOULD be possible to answer a call in voice or video
without text enabled, and add text later in the call.

d. It MUST be possible to disconnect the call.

e. It SHOULD be possible to control IVR (Interactive Voice
Response) services from a numeric keypad.

f. It SHOULD be possible to control ITR (Interactive Text
Response) services from the alphanumeric keyboard.

g. It SHOULD be possible to invoke multi-party calls.

h. It SHALL be possible to transfer the call.

i. It MUST be possible to use text characters (numbers) instead of
DTMF tones (numbers) in interactions where the person is using a

keyboard to interact with a service and the service asks for a
number.

8.2.8.4 Audio/Visual/Tactile Indicators

It SHOULD be possible to observe visual or tactile indicators
about:
- Call progress
- Availability of text, voice and video channels.
- Incoming call.
- Incoming text.
- Typed and transmitted text.
- Any loss in incoming text.

8.2.9 Additional session control

Systems that support additional session control features, for
example call waiting, forwarding, hold etc on voice calls, MUST
offer equivalent functionality for real-time Text-over-IP
functions. In addition, all these features MUST be controllable by
text users at any time, in an equivalent way as for other users.

8.2.10 File storage

Systems that support real-time Text-over-IP MAY save the text
conversation to a file. This SHOULD be done using a standard file
format.

8.3 Conference Call Requirements

The conference call requirements deal with multipoint conferencing
calls where there will be at least one or more ToIP capable
devices along with other end user devices where the total number
end user devices will be at least three.

It SHOULD be possible to use the text medium in conference calls,
in a similar way as video is handled and displayed. Text in
conferences can be used both for letting individual participants
use the text medium, and for central support of the conference
with real time text interpretation of speech.

8.4 Transport

ToIP uses RTP as the default transport protocol for transmission
of real-time text medium text/t140 as specified in RFC 2793 [5].
Signaling and other media will use the transport protocol
specified in SIP [3] and/or their revised versions as specified in
standards.

The redundancy method of RFC 2198 SHOULD be used for making text
transmission reliable with transmission of three generations.

Text capability SHOULD be announced in SDP by a declaration in
line with this example:

```
m=text 11000 RTP/AVP 98 100
a=rtpmap:98 t140/1000
a=rtpmap:100 red/1000
a=fmtp:100 98/98/98
```

Characters SHOULD BE buffered for transmission and transmitted
every 300 ms.

By having this single coding and transmission scheme for real time
text defined, in the SIP call control environment, the opportunity
for interoperability is optimized.

However, if good reasons exist, other transport mechanisms MAY be
offered and used for the T.140 coded text, provided that proper
negotiation is introduced, and RFC 2793 transport MUST be used as
the defaut fallback solution.

8.5 Character Set

a. Real-Time Text-over-IP protocols MUST use UTF-8 encoding as
specified in ITU-T T.140 [12].

b. Real-time Text-over-IP SHOULD handle characers with editing
effect such as new line, erasure and alerting during session as
specified in ITU-T T.140.

8.6 Transcoding

Transcoding of text may need to take place in gateways between
ToIP and other forms of text conversation. ToIP makes use of
ISO 10646 character set.
Most PSTN textphones use a 7-bit character set, or a character set
that is converted to a 7-bit character set by the V.18 modem.

When transcoding between these character sets and T.140 in
gateways, special consideration MUST be paid to the national
variants of the 7 bit codes, with national characters mapping into
different codes in the ISO 10 646 code space. The national variant
to be used SHOULD be possible to select by the user per call, or
be configured as a national default for the gateway.

The missing text indicator in T.140, specified in T.140 amendment

1, cannot be represented in the 7 bit character codes. Therefore
these characters SHOULD be translated to be represented by the '
(apostrophe) character in legacy text telephone systems where this

character exists. For legacy systems where the character ' does
not exist, the character . (full stop) SHOULD be used instead.

[8.7](#) Relay Services

The relay service acts as an intermediary between 2 or more
callers.
The basic relay service allows a translation of speech to text and
text to speech, which enables hearing and speech impaired callers
to communicate with hearing callers. Even though this document
focuses on ToIP, we do not exclude video relay services for e.g.,
speech to sign language and vice versa and other possible relay
services. It will be possible to use ToIP simultaneously with
other relay services if desired.

It is very important for the users that a relay session is invoked
as transparently as possible. It SHOULD happen automatically when
the call is being set-up or by a simple user action. A transcoding
framework document using SIP [7] describes invoking relay
services, where the relay acts as a conference bridge or uses the
third party control mechanism.

Adding or removing a relay service MUST be possible without
disrupting the current call.

When setting up a call, the relay service MUST be able to
determine the type of service requested (e.g. speech to text or
text to speech), to indicate if the caller wants voice carry over,
the language of the text including the sign language being used.

The user MUST be provided with a method to indicate which service
is desired.

Relay services MUST be reachable all the time, even if the users
are visiting other operator?s networks.

It SHOULD be possible to route the call to a preferred relay
service even if the user makes the call from another region or
network than usually used.

It MUST be possible to identify ToIP sessions as emergency
sessions.

If it is decided that a relay service supports emergency calls,
the relay service operator MUST be able to process such a session
correctly and quickly with the following functionality:

a. The relay service operatorÆs network MUST give priority to this
incoming call.

b. The relay service operator MUST forward this session if they
are unable to process it to an alternative emergency relay
operator.

c. The relay service MUST label the transcoded stream as an
emergency call (in case of text to speech and/or vice versa).

d. The relay service MUST provide all session information to the
emergency centre (e.g., location information of the caller if
available).

8.8 Emergency services

a. It SHOULD be possible to support emergency service calls with
text only or simultaneously with voice.

b. All session information that accompanies a voice session to the
emergency centre, should also be provided to the emergency centre
if it is a ToIP session.(e.g, phone number and location
information of the user placing the emergency call).

c. A text over IP stream must be labelled as an emergency stream
to ensure that the emergency service center is able to receive
this call.

8.9 User Mobility

ToIP terminals SHOULD use the same mechanisms as other terminals
to resolve mobility issues. It is RECOMMENDED to use a SIP-adress
for the users, resolved by a SIP REGISTRAR, to enable basic user
mobility. Further mechanisms are defined for the 3G IP multimedia
systems.

8.10 Confidentiality and Security

UsersÆ confidentiality and privacy need to be met as described in
SIP [3]. For example, nothing should reveal the fact that the user
of ToIP is a person with a disability unless the user prefers to

make this information public. If a transcoding server is being used, this SHOULD be transparent. Encryption SHOULD be used on end-to-end or hop-by-hop basis as described in SIP [3] and SRTP [19]

Authentication needs to be provided for users in addition to the message integrity and access control.

Protection against Denial-of-service (DoS) attacks needs to be provided considering the case that the ToIP users might need transcoding servers.

8.11 Call Flows

ToIP is a way of establishing the real-time conversation. Call flow for ToIP SHOULD be similar to session establishment with audio and video. For example, ToIP services MAY be invoked in the following situations (among others):

- Noisy environment (e.g., in a machine room of a factory where listening is difficult)Busy with another call and want to participate in two calls at the same time.

- Text and/or speech recording services (e.g., text documentation/audio recording for legal/clarity/flexibility purposes)
- Overcoming of language barriers through speech translation and/or transcoding services

- Not hearing well or not at all (e.g., hearing loss due to aging, hard of hearing, deaf)

NOTE: In many of the above scenarios, text may accompany speech in a subtitling like fashion. This would occur for individuals who are hard of hearing and also for mixed calls with a hearing and deaf person listening to the call.

All call flows either for the point-to-point or for the multipoint situation need to consider that ToIP services may be invoked for many different reasons by users as explained. When the transcoding/translation services are needed, call flows will be shown for both session establishment models: Third-party call control model and Conferencing bridge model.

8.11.1 Call Scenarios

There are 2 different terminal types possible:

1. The terminal itself has the intelligence to initiate a relay
service for incoming and outgoing calls (based on address book,
user preferences programmed on the terminal etc. This terminal can
be used in a conference bridge call as well as a third party
control call.

2. Dumb terminals, so that the relay service server actually
initiates the correct call handling (the dumb terminal can only
REFER the call to the relay center, which then sets up the call
using the conference bridge flow.).

The following call scenarios are shown:

- Communications between two ToIP/Multimedia capable, end user
devices using the same language.

- Communications between ToIP capable, end user devices using
translation services to provide language translation.

- Communications between ToIP/Multimedia capable and Audio (non-
ToIP) capable end user devices.

- Communications between ToIP/Multimedia and/or Audio (non-
ToIP)/Multimedia end user devices maintaining privacy.

8.11.2 **Point-to-Point Call Flows**

The point-to-point calls will contain at least one or both
ToIP/Multimedia devices in setting up the session. The detail call
flows need to be provided in the following scenarios:

- ToIP/Multimedia devices that use the same language.

- ToIP/Multimedia devices invoke translation services for using
different languages.
    * Third-party call control model.
    * Conference bridge service model.

- ToIP/Multimedia devices invoke translation services for using
different languages maintaining privacy.
    * Third-party call control model.
    * Conference bridge service model.

- ToIP/Multimedia device and Audio (non-ToIP)/Multimedia device
          invoking transcoding server.
             * Call initiated by Audio (non-ToIP)/Multimedia user
               - Third-party call control model.
               - Conference bridge service model.
             * Call initiated by ToIP user.
               - Third-party call control model.
               - Conference bridge service model.

          - ToIP/Multimedia device and Audio (non-ToIP)/Multimedia device
          invoking transcoding server maintaining privacy.
             * Call initiated by Audio (non-ToIP)/Multimedia user
               - Third-party call control model.
               - Conference bridge service model.
             * Call initiated by ToIP user.
               - Third-party call control model.
               - Conference bridge service model.

8.11.3 Conference Call Flows

     Conference call flows only contain the multipoint communications
     scenarios, and only the centralized bridge model is considered.
     The following multipoint conference call flow scenarios will
     contain at least one more ToIP/Multimedia devices:

     - ToIP/Multimedia devices that use the same language.

     - ToIP/Multimedia devices invoke translation services for using
     different languages.

     - ToIP/Multimedia devices invoke translation services for using
     different languages maintaining privacy.

          - ToIP/Multimedia device and Audio (non-ToIP)/Multimedia device
          invoking transcoding server.
             * Call initiated by Audio (non-ToIP)/Multimedia user.
             * Call initiated by ToIP/Multimedia user.

          - ToIP/Multimedia device and Audio (non-ToIP)/Multimedia device
          invoking transcoding server maintaining privacy.
             * Call initiated by Audio (non-ToIP)/Multimedia user.
             * Call initiated by ToIP/Multimedia user.

9. Interworking Requirements for Text-over-IP

     A number of systems for real time text conversation already exist

as well as a number of message oriented text communication systems. Interoperability is of interest between ToIP and some of these systems. This section describes requirements on this interoperability.

9.1 Real-Time Text-over-IP Interworking Gateway Services

Interactive texting facilities exist already in various forms and on various networks. On the PSTN, it is commonly referred to as text telephony. The simultaneous or alternating use of voice and text is used by a large number of users who can send voice, but must receive text or who can hear but must send text due to a speech disability.

9.2 Text-over-IP and PSTN/ISDN Text-Telephony

On PSTN networks, transmission of interactive text takes place using a variety of codings and modulations, including ITU-T V.21 [II], Baudot, DTMF, V.23 [III] and others. Many difficulties have arisen as a result of this variety in text telephony protocols and the ITU-T V.18 [10] standard was developed to address some of these issues.

ITU-T-V.18 [10] offers a native text telephony method plus it defines interworking with current protocols. In the interworking mode, it will recognise one of the older protocols and fall back to that transmission method when required.

In order to allow systems and services based on Real-time Text-over-IP to communicate with PSTN text telephones, text gateways are the recommended approach. These gateways MUST use the ITU-T V.18 [10] standard at the PSTN side.

Buffering MUST be used to support different transmission rates. At least 1K buffer MUST be provided. A buffer of at least 2K characters is recommended. In addition, the gateway MUST provide a minimum throughput of at least 30 characters/second or the highest speed supported by the PSTN text telephony protocol side, whichever is the lowest.

PSTN-Real-time Text-over-IP gateways MUST allow alternating use of text and voice.

PSTN and ISDN to real-time Text-over-IP gateways that receive CLI information from the originating party MUST pass this information to the receiving party as soon as possible.

Priority MUST be given to calls labeled as emergency calls.

9.3 Text-over-IP and Cellular Wireless circuit switched Text-
Telephony

   Cellular wireless (or Mobile) circuit switched connections provide
   a digital real-time transport service for voice or data.
   The access technologies include GSM, CDMA, TDMA, iDen and various
   3G technologies.

   Alternative means of transferring the Text telephony data have
   been developed when TTY services over cellular was mandated by the
   FCC in the USA. They are a) æNo-gainÆ codec solution, b) the
   Cellular Text Telephony Modem (CTM) solution and c) æBaudot modeÆ
   solution.

   The GSM and 3G standards from 3GPP make use of the CTM modem in
   the voice channel for text telephony.
   However, implementations also exist that use the data channel to
   provide such functionality. Interworking with these solutions
   SHOULD be done using text gateways that set up the data channel
   connection at the GSM side and provide real-time Text-over-IP at
   the other side.

9.3.1 "No-gain"

   The "No-gain" text telephone transporting technology uses
   specially modified EFR [15] and EVR [16] speech vocoders in both
   mobile terminals used provide a text telephony call. It provides
   full duplex operation and supports alternating voice and text.(
   "VCO/HCO"). It is dedicated to the CDMA and TDMA mobile
   technologies and the US Baudot type of text telephones.

9.3.2 Cellular Text Telephone Modem (CTM)

   CTM [17] is a technology independent modem technology that
   provides the transport of text telephone characters at up to 10
   characters/sec using modem signals that are at or below 1 kHz and
   uses a highly redundant encoding technique to overcome the fading
   and cell changing losses. On any interface that uses analog
   transmission, half-duplex operation must be supported as the
   æsendÆ and æreceiveÆ modem frequencies are identical. The use of
   CTM may have to be modified slightly to support half-duplex
   operation.

9.3.3 "Baudot mode"

This term is often used by cellular terminal suppliers for a GSM cellular phone mode that allows TTYs to operate into a cellular phone and to communicate with a fixed line TTY.

9.3.4 Data channel mode

Many mobile terminals allow the use of the data channel to transfer data in real-time. Data rates of 9600 bit/s are usually supported on the mobile connection.Gateways or the interworking function provides interoperability with PSTN textphones.

9.3.5 Common Text Gateway Functions

Text Gateways MUST cover the differences that result from different text protocols. The protocols to be supported will depend on the service requirements of the Gateway.

Different data rates of different protocols MAY require text buffering.

Interoperation of half-duplex and full-duplex protocols MAY require text buffering and some intelligence to determine when to change direction when operating in half-duplex.

Identification may be required of half-duplex operation either at the æuserÆ level (ie. users must inform each other) or at the æprotocolÆ level (where an indication must be sent back to the Gateway).

A Text Gateway MUST be able to route text calls to emergency service providers when any of the recognised emergency numbers that support text communications for the country or region are called eg. æ911Æ in USA and æ112Æ in Europe.

A text gateway MUST act transparently on the IP side. It acts then as a virtual end-point terminal.

9.4 Text-over-IP and Cellular Wireless Text-over-IP

Text-over-IP MAY be supported over the cellular wireless packet switched service. It interfaces to the Internet. For 3GPP 3G services, the support is described to use ToIP in 3G TS 26.235 [20].

A Text gateway with cellular wireless packet switched services MUST be able to route text calls into emergency service providers when any of the recognized emergency numbers that support text communication for the country are called.

9.5 Instant Messaging Support

Instant Messaging is used by many people to communicate using text via the Internet. Instant Messaging transfers blocks of text rather than streaming as is used for real-time Text-over-IP. As such, it is not a replacement for real-time Text-over-IP and in particular does not meet the needs for real time conversations of deaf, hard of hearing and speech-impaired users. It is unsuitable for communications through a relay service [I]. The streaming character of real-time Text-over-IP provides a better user experience and, when given the choice, users often prefer real-time Text-over-IP.

However, since some users might only have Instant Messaging available, text gateways might be developed that allow interworking between Instant Messaging systems and real-time Text-over-IP solutions.

Because Instant Messaging is based on blocks of text, rather than on a continuous stream of characters, such gateways need to transform between these two formats. Text gateways for interworking between Instant Messaging and real-time Text-over-IP MUST concatenate individual characters originating at the real-time Text-over-IP side into blocks of text and:

a. When the length of the concatenated message becomes longer than 50 characters, the buffered text MUST be transmitted to the Instant Messaging side as soon as any non-alphanumerical character is received from the real-time Text-over-IP side.

b. When a new line is received from the real-time Text-over-IP side, the buffered characters up to that point, including the carriage return and/or line feed characters, MUST be transmitted to the Instant Messaging side.

c. When the real-time Text-over-IP side has been idle for at least 5 seconds, all buffered text up to that point MUST be transmitted to the Instant Messaging side.

It is recommended that during the session, both users are constantly updated on the progress of the text input. For example, many Instant Messaging protocols signal that a user is typing to the other party in the conversation. Text gateways between Instant Messaging and real-time Text-over-IP MUST provide this signaling to the Instant Messaging side when characters start being received, or at the beginning of the conversation. Also at the real-time text-over-IP side, an indicator of writing

the Instant Message MUST be present. For example, the real-time
text user will see . . . waiting for replying IM. . . And per 5
seconds a . (dot) can be shown.
Those solutions will reduce the difficulties between a streaming
versus blcoked text.

Even though that the text gateway can connect Instant Messaging
and real-time Text-over-IP. The best solution is to take advantage

of the fact that the user interfaces and the user communities for
instant messaging and real-time text-over-IP telephony are
extremely similar.

After all, the character input, the character display, Internet
connectivity, SIP stack, etc are the same for Instant Messaging
and real-time Text-over-IP.

Devices that implement Instant Messaging SHOULD implement real-
time text-over-IP telephony, using standard SIP and text/t140
mechanisms.

9.6 IP Telephony with Traditional RJ-11 Interfaces

Analogue adapters using SIP based IP communication and RJ-11
connectors for connecting traditional PSTN devices SHOULD enable
connection of legacy PSTN text telephones [18]. These adapters
SHOULD contain V.18 modem functionality, voice handling
functionality, and conversion functions to/from SIP based ToIP
with T.140 transported according to RFC 2793, in a similar way as
it provides interoperability for voice calls. If a call is set up
and text/t140 capability is not declared by the endpoint (by the
end-point terminal or the text gateway in the network at the end-
point), a method for invoking a transcoding server shall be used.
If no such server is available, the signals from the textphone MAY
be transmitted in the voice channel as audio with high quality of
service.
NOTE: It is preferred that such analogue adaptors do use RFC2793
on board and thus act as a text gateway. Sending textphone signals
over the voice channel is undesirable due posible filtering and
compression and packet loss between the end-points. Which can
result in dropping characters in the textphone conversation or
even not allowing the textphones to connect with each other.

9.7 Interworking Call Flows

<< this chapter will change depending on how chapter 9.10 works
out>>

The call flows in chapter 8.11 deal with end to end ToIP. These
call flows do not change on the IP network when one end-point is
actually a text gateway. The text gateway actually acts like a
ToIP/Multimedia device. Separate call flows will show the
interworking between the ToIP/Multimedia devices [4] over the IP
network and the text telephony devices [10] over the PSTN/ISDN
network using the IP-PSTN/ISDN interworking functional (IWF)
entity. It is assumed that the IWF will provide ToIP and text
telephony interworking in addition to other capabilities. Thus
acting as a Text gateway.

The point-to-point call flows will contain at least one
ToIP/Multimedia and one text telephony/multimedia (or POTS) device
for the following cases:

- ToIP/Multimedia device and text telephony/multimedia device that
use the same/different language.
- ToIP/Multimedia device and PSTN/ISDN-based POTS/Multimedia
device.

For multipoint conferencing calls, it is assumed that only the
centralized conferencing will be considered, and the media bridge
is supposed to be located somewhere in the SIP network. However,
it is considered that the ToIP and text telephony interworking
function will be located in the IWF.

The multipoint conference call flows will contain at least one
ToIP/Multimedia device, at least one text telephony/multimedia
device, and other devices where total number of devices will be
three or more for the following cases:

- ToIP/Multimedia and text telephony/multimedia devices that use
the same/different language.
- ToIP/Multimedia devices, telephony/multimedia devices, and/or
PSTN/ISDN-based POTS/Multimedia devices.

9.8 Multi-functional gateways

The scenarios described in this document deal with single pairs of
interworking protocols or services. However, in practice many of
these interworking systems will be implemented as gateways that
combine different functions. As such, a text gateway could be
build to have modems to interwork with the PSTN and support both
Instant Messaging as well as real-time ToIP. Such interworking
functions are called Combination gateways.

Combination gateways MUST provide interworking between all of
their supported text based functions. For example, a text gateway
that has modems to interwork with the PSTN and that support both
Instant Messaging and real-time ToIP MUST support the following
interworking functions:

- PSTN text telephony to real-time ToIP.
- PSTN text telephony to Instant Messaging.
- Instant Messaging to real-time ToIP.

9.9 Gateway Discovery

To get a smooth invocation of the text gateways, where those
gateways are transparant on the IP side, it requires a method how
and when to invoke the text gateway. As described previously in
this draft. The text gateways must act as the end-terminal. The
capabilities of the text gateway will in that call be determined
by the call capabilities of the terminal that is using the
gateway. For example, a PSTN textphone is only able to receive

voice and streaming text. Thus the text gateway will only allow
ToIP and audio.

The PSTN devices or other non IP multimedia devices that require
the text gateways to connect to the IP must be able to locate the
text gateway, and ensure that the correct call capabilities of the
non IP multimedia device is used by the text gateway.

The following possible solutions for using the text gateway are:

- PSTN Textphone users using a prefix number before dialing out.
- In band text dialogue, where the gateway asks the user for the
destination address.
- separate text subscriptions, linked to the phone number or
terminal identifier/ IP address.
- text capability indicators.
- text preference indicator.
- listen for text activity in all calls.
- call transfer request by the called user.
- placing a call via the web, and use one of the methods described
here
- text gateways with its own telephone number and/or SIP address.
(this requires user interaction with the text gateway to place a
call).

- ENUM address analysis and number plan
- number or address analysis leads to the gateway for all PSTN calls.
- etc

9.10 Text Gateway in the call Scenarios

9.10.1 IP terminal calling an analogue textphone (PSTN)

The ToIP stream will be converted into an analogue text telephone protocol (using the voice channel) and vice versa by the text gateway.

The PSTN knows that it may be a textphone call thanks to the SDP description (for example: m=text 11000 RTP/AVP 98 a=rtpmap:98 t140/1000 for T.140 text on port 11000). It can then activate text gateway functions on the PSTN side listening for a text answer.

The PSTN will also know that all those incoming calls are only for analogue textphones. Thus the speed of the text stream is adjusted to the selected analogue textphone protocol.
If there is no analogue textphone on the called number, the call setup will be terminated by the text gateway.

The text gateway can be implemented in two ways: The PSTN has its own text gateway (the IWF), or it redirects the media stream to the nearest IP-PSTN gateway with text transcoding abilities.

Text gateway detection: In the SIP messages.

9.10.2 IP terminal calling a mobile text telephone (CTM)

The ToIP stream will be converted into CTM and vice versa by the text gateway located in the network of the cellular/mobile operator. It is similar to the PSTN.

Text gateway detection: In the SIP messages.

9.10.3 IP terminal calling a mobile telephone (GPRS based)

A text gateway located in the mobile network converts the incoming T.140/RTP stream into for example T.140 over TCP (T.140/TCP) or tunnels the T.140 stream over HTTP (T.140/HTTP). Or any other temporarily non standard solution necessary to connect the text gateway with the text telephone client on the mobile phone.

This is necessary, since RTP over GPRS is not possible in many
mobile phones.
Note, those server-client solutions are ONLY acceptable for the
GPRS and non RTP stack phones. It is encouraged to use T.140/RTP
as soon as possible for all mobile phones.
Allowing UDP transport over the GPRS link will enable RFC2793 text
over GPRS.

Text gateway detection: In the SIP messages.

9.10.4 IP terminal calling a mobile telephone(UMTS)

No text gateway is required here since this will be end to end IP.

9.10.5 Analogue textphone (PSTN) user calling an IP terminal using
prefix

The PSTN is unable to distinguish between an analogue voice call
and an analogue textphone, both use the voice channel. The text
gateway needs to transcode the analogue textphone protocol into
T.140/RTP.

One way for a PSTN to separate an incoming voice call into text
telephony or normal voice is by using a prefix number for all
incoming text telephone calls to the PSTN. For example , the text
telephone user (e.g Boudot) places a call and enters a prefix e.g.
600 and then continues with the original number. The PSTN will
recognize all incoming 600 calls as an analogue textphone call and
redirects the call to a text gateway (unless it is a number
connecting the same PSTN).

It is undesirable to allow a PSTN to transport all the analogue
textphone tones/signals through a VoIP stream! (In band text
dialogue).

Text gateway detection: Prefix number for incoming textphone
calls.

9.10.6 Mobile text telephone (CTM) user calling an IP terminal

The voice channel of the cellular network is used. The MSC is able
to separate between the text call and voice only, it is just a
matter of redirecting the voice channel to the text gateway.

Text gateway detection: CTM signal detection.

9.10.7 Mobile telephone user (GPRS) calling an IP terminal

The text telephone client on the mobile telephone connects the
text gateway located in the network. The text gateway transcodes
the text stream into ToIP.

Text gateway detection: pre-programmed in the mobile textphone
client.

9.10.8 Mobile telephone (UMTS) user calling an IP terminal

No text gateway is required here since this will be end to end IP.

9.10.9 Voice over DSL user using an analogue text telephone.

Voice over DSL is a widespread service. When connecting analogue
text telephones to this service there is a risk that they just use
the voice channel that result in corrupted text transmission. The
VoDSL gateway located in the network of the (A)DSL operator itself
should connect with a text gateway as soon it turns into VoIP.

Text gateway detection: prefix number similar to the PSTN.

9.10.10 VoIP user via a building telephone switch (at an apartment
building) owning an analogue text telephone.

This is the case where only VoIP is possible and no other IP
traffic between the telephone switch and the apartments.
The only solution would be a forced analogue text telephone
protocol over the Voice channel, in band text dialogue . If that
must happen. Then the telephone switch MUST convert the analogue
text telephone protocol into ToIP and vice versa before the
telephone switch connects the IP network.
Note: The in band text dialogue is undesirable. This scenario
SHOULD be avoided at any cost.

Text gateway detection: prefix number or in band text signalling.

9.10.11 VoIP user via a gateway/box connected to his/her own
Broadband connection owning an analogue text telephone.

The gateway box should natively transcode analogue text telephony
into ToIP and vice versa when an analogue text phone is plugged in

the RJ-11 socket [18].

Text gateway detection: RJ-11 socket preconfigured by the box via jumpers or software, or listen for textphone tones and perform V.18 text telephone detection.

10. Terminal Features

Implementers of products that support interactive Text-over-IP SHOULD NOT assume that all users of text are able to use mainstream input and output devices. People with arthritis or other dexterity problems might not be able to use very small keyboards. Visually impaired people might not be able to use standard sized characters on a display. Colour-blind people might suffer from badly chosen colour-schemes. People with motor disabilities might require specialised input devices.

Implementers SHOULD make their products as open as possible with regard to this wide range of abilities and preferences and they MUST use standard interfaces wherever they provide such interfaces.

10.1 Text input

Systems that support real-time interactive Text-over-IP SHOULD support suitable input mechanisms, either built-in or connectable through the use of a standard interface: PS/2, USB, Bluetooth, or virtual keyboard. In particular Braille users should be able to connect Braille keyboards to the terminal. Terminals MAY support a web interface for input and output of text.

It is recommended that systems that fixed terminals that support real-time interactive Text-over-IP allow the user to enter the standard alphanumerical characters directly, rather than through a cycle of key presses or other indirect means. This could be done using full-sized keyboards, smaller sized keyboards or fastap keyboards for example. It is highly recommended to provide a standard interface to allow attachment of an external input device, especially for terminals that have only limited input systems built-in.

Systems should provide means to add voice-to-text translation as text input.

All IP phones with a display of 12 or more characters MUST support at least text input through the regular phone keypad (and display of any incoming text) in order to provide basic emergency text communication from any IP phone.

Input devices that have automatic key repeat MUST allow the user
to specify the key-repeat rate.

[10.2](#) Text presentation

Systems that support real-time interactive Text-over-IP SHOULD
support suitable displays, either built-in or connectable through
the use of a standard interface: S-VGA, USB, Bluetooth or IP.
Braille readers should be connectable to the terminal using a
standard interface.

Terminals MAY support a web interface for input and output of
text.

A variety of handsets and terminals might be developed for a
number of equally varied scenarios.

In the case of fixed terminals or software applications on
Personal Computers, implementers MUST:

a. Use either separate screen areas for displaying sent and
received text OR clearly indicate the difference between sent and
received text. Systems MAY allow the user to chose either on of
these presentation methodologies.

b. Provide at least 5 lines of 35 monospaced characters each for
each direction (sent and received text) OR at least 10 lines of 35
characters when sent and received text are presented together.

In the case of Mobile terminals, implementers MUST:

c. Use either separate screen areas for displaying sent and
received text OR clearly indicate the difference between sent and
received text. Systems MAY allow the user to chose either on of
these presentation methodologies.

d. Provide at least 3 lines of 20 monospaced characters each for
each direction (sent and received text) OR at least 6 lines of 20
characters when sent and received text are presented together.

On both types of terminals, scrolling back through both sent and
received text MUST be supported, even after the conversation has
ended. Lines SHOULD be wrapped at word boundaries .

There MUST be an easy-to-use function to clear the screen at any
time during the session, and if the implementation has chosen to
present sent and received text separately, clearing the screen
SHOULD be possible as a separate function for sent and received

text.

The function of the new line and erasure controls as explained in
[section 9.5](#). MUST be supported by the presentation in the

consistent way described by T.140. Presentation layers MUST
support the full UTF-8 character set.

When real-time Text-over-IP is used in conjunction with other
modalities, like voice, the presentation MUST clearly indicate
this to the user in an area outside the display region for send
and received text.

Identification information for other parties in the conversation,
like URLÆs, user-friendly names from an address book, or CLI in
the case of conversations with text telephones, SHOULD be
displayed throughout the entire conversation in a region outside
the sent and received text area.

[10.3](#) Call control

Call (Session) Control procedures MUST use the SIP protocol. Text
sessions MUST be identified in accordance with requirements
described earlier.

Text services SHOULD be part of a Total Conversation environment
in which voice, text and video sessions can be added, modified or
deleted individually.

To enable interworking with Textphones in telephone and cellular
(mobile) networks, terminals MUST be able to access Gateways
automatically when a PSTN or cellular (mobile) E.164-based
telephone number is used as the called address.

Users MUST be able to establish text sessions to emergency service
providers using the widely recognised emergency numbers in use in
the country or region of operation of the terminal eg. æ911Æ in
USA and ?112?in Europe.

The ability to transfer Location information SHALL be provided if
the information is available from the terminal.

[10.4](#) Device control

ToIP devices shall support multiple means of setting up and

performing calls as well as controlling the device itself. The built-in controls and presentation systems shall take accessibility aspects into account as far as possible. The device shall include external interfaces that makes it possible to attach user interface devices for people with needs beyond what the built-in user interface can support. It is preferrable if such external interfaces are wireless.

10.5 **Alerting**

The form of Alerting indication(s) provided to the user should be selectable to suit particular users. Alerting indications MAY include Sound, Tactile (eg. vibrational), Visual (on-screen symbols; separate flashing light), Motion (eg. movement of something).

The ability to send an Alerting signal to an external interface SHOULD be provided. This will allow Alerting devices that are specific to users requirements to be attached.

As many as possible of the following alternatives for alerting SHOULD be provided:
    * Internal flash.
    * Two-pole connector for external alerting systems triggered by contact between the two poles when a ring signal is generated (if necessary with 1.5-9 V battery power for alerting systems requiring electrical currents to activate).
    * Bluetooth serial profile with AT command interface, sending the "RING" message, intended for a Bluetooth alerting receiver with flash, vibration or sound action.
    * SIP connected alerting device, that get its stimuli by being registered on the same sip address as the terminal.

10.6 **External interfaces**

Terminals for ToIP SHOULD provide external interfaces for the following functions:
    * Text input.
    * Text display.
    * Terminal control.
    * Session control.

10.7 **Power**

As terminals could remain active for very long periods of time, the electrical power requirements of all the terminals SHOULD be as low as possible.

If the terminal is to be used for calling Emergency services or where the mains power supply is unreliable, back-up power systems SHOULD be provided for the terminal and all equipment used to provide the ToIP service. This can be implemented in many different ways eg. via the line powering option on some Ethernet interfaces, or by using a æno breakÆ power supply (a battery back-up system with inverters that can recreate a limited amount of mains power).

11. Security Considerations

There are no additional security requirements other than described earlier.

12. AuthorsÆ Addresses

The following people provided substantial technical and writing contributions to this document, listed alphabetically:

Barry Dingle
ACIF, 32 Walker Street
North Sydney, NSW 2060 Australia
Tel +61 (0)2 9959 9111
Fax +61 (0)2 9954 6136
TTY +61 (0)2 9923 1911
Mob +61 (0)41 911 7578
email barry.dingle@bigfoot.com.au

Guido Gybels
RNID, 19-23 Featherstone Street
London EC1Y 8SL, UK
Tel +44(0)20 7294 3713
Txt +44(0)20 7296 8019
Fax +44(0)20 7296 8069
EMail: guido.gybels@rnid.org.uk

Gunnar Hellstrom
Omnitor AB
Renathvagen 2
SE 121 37 Johanneshov
Sweden
Phone: +46 708 204 288 / +46 8 556 002 03

        Fax: +46 8 556 002 06
        Email: gunnar.hellstrom@omnitor.se

        Paul E. Jones
        Cisco Systems, Inc.
        7025 Kit Creek Rd.
        Research Triangle Park, NC 27709
        Phone: +1 919 392 6948
        E-mail: paulej@packetizer.com

        Radhika R. Roy
        AT&T
        Room C1-2B03
        200 Laurel Avenue S.
        Middletown, NJ 07748
        USA
        Phone: +1 732 420 1580
        Fax: +1 732 368 1302
        Email: rrroy@att.com

        Henry Sinnreich
        MCI
        400 International Parkway
        Richardson, Texas 75081
        Email: henry.sinnreich@mci.com

        Gregg C Vanderheiden
        University of Wisconsin-Madison
        Trace R & D Center
        1550 Engineering Dr (Rm 2107)
        Madison, Wi 53706
        USA
        gv@trace.wisc.edu
        Phone +1 608 262-6966
        FAX +1 608 262-8848

        Arnoud A. T. van Wijk
        Viataal (Dutch Institute for the Deaf)
        Research & Development
        Afdeling RDS
        Theerestraat 42
        5271 GD Sint-Michielsgestel
        The Netherlands.
        Email: a.vwijk@viataal.nl

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