Proposal R1v3
for
Implementation of Real-Time Text Across Platforms

Built on the
Industry-Consumer Consensus Provisions on Real-Time Text, submitted by the Telecommunications and Electronic and Information Technology Advisory Committee (TEITAC) to the Architectural and Transportation Barriers Compliance Board (Access Board) in April 2008

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1.) Introduction

Real-time text (RTT) is a mode of communication that permits flowing text conversation, mixing of text and voice.

- **It works for people who are deaf,** providing continuous and instant communication similar to speech or signing.
- **It works for people who are hard of hearing**, providing the ability to use speech where it works, with text for numbers, addresses, or any other text that is not understood. It supports captioned telephony without requiring special technology. It also allows bidirectional captioning with speech recognition allowing the speaker to detect and correct speech recognition mistakes – allowing reliable use of speech recognition earlier (with text relay available when auto-captioning does not work).
- **It works with people who have impaired speech,** by allowing them to speak whenever it is understandable, and use text in parallel for words, topics, addresses, numbers or other times when their speech is not understandable.
- **It allows people who are deaf-blind** to communicate with anyone, using standard phones and their Bluetooth connections to braille displays.
- **It works for emergency calls** allowing anyone to call in with any ordinary phone and use text in parallel with speech for any parts that are not understood by either party. The real-time nature allows messages to get to 911 faster, ensures partial messages are received when they are interrupted mid sentence for any reason (caller passes out, caller is prevented from completing sentence by another person, etc), allows the 911 operator to optimize the information being sent, allows captioning of the 911 operators speech, etc. One particular advantage of ubiquitous real-time text is the ability to for 911 operators to caption their message for people who do not ordinarily use real-time text but are in a situation where the noise combined with their hearing does not allow them to understand the 911 operator.

The availability, reliability, and interoperability of real-time text communication are important for all of these groups. The increasing number of older people who are hard of hearing and will need text to supplement incoming speech in order to have reliable phone communication makes the need for real-time text all the more critical.

The purpose of this document, Proposal R1v3, is to provide a common point of discussion, and to clear up any confusion or misunderstandings about what is involved in implementing real-time text. This proposal builds upon a consumer-industry consensus agreement reached during the deliberations of the U.S. Access Board's Telecommunications and Electronic and Information Technology Advisory Committee (TEITAC) as well as the work of the FCC's EAAC, AT&T and others. More information about TEITAC is included in Appendix 1.

Specifically, Proposal R1v3 seeks:

1) to provide a concrete proposal, with details, around which discussion can be carried forward and agreements reached
2) to answer questions and clear up misunderstandings
3) to lay out the benefits of real-time text to all stakeholders including:
   consumers with disabilities - elders - all telecommunications users, especially in emergency situations - and industry

ORGANIZATION OF THIS DOCUMENT

This document contains the following Sections:

1) **Introduction –** (previous page)
2) **Proposal R1v3** – Concise **2 page proposal** for real-time text
3) **FAQ –** addresses key user and industry questions
4) **Benefits to Users** – outlines benefits of R1v3 to users (with & without disabilities)
5) **Industry Benefits and Considerations –** addresses issues of interest to industry
6) **Real-time Text Standards –** summarizes all existing real-time text standards
7) **RTT Implementations –** provides additional implementation information and options
8) **References and Extracts –** provides excerpts from standards and other documents describing implementation specifics for the real-time text feature

Appendices

(Appendix 1) TEITAC Consensus Language – that this proposal is built upon

(Appendix 2) Details on requirements and benefits for different user groups

2.) Proposal R1v3 - In two pages

Goal of Proposal R1v3

The goal of Proposal R1v3 is to ensure that, wherever voice is supported over IP for conversational communication, reliable and interoperable real-time text, is also available that can be used alone, or together with voice on the same call.

Quick Summary Benefits of Proposal R1v3

Cross-disability – Proposal R1v3addresses the needs of a wide range of users including people who are ***deaf***, ***late deafened*** (and communicate in speech and text), ***hard of hearing*** (and need captioned telephone), those who can hear but ***have a speech-disability,*** and those who are ***deaf-blind –*** *allowing all of them to communicate using standard phones*.

Standards based – Proposal R1v3 is based on industry standards from the standards groups (ITU and IETF) that created almost all of the standards used in VoIP today.

Mainstream technology based – – Proposal R1v3 proposes access via mainstream phone technologies – moving away from special devices (such as TTYs). This eliminates the need to support, distribute and maintain compatibility between mainstream devices or networks and special devices that use non-mainstream text formats. It would also allow users with disabilities to be able to take advantage of the same cost saving calling plans, bundles, etc from telecommunication vendors that everyone else enjoys.

Mainstream usable – Proposal R1v3 enables new mainstream applications and features on phones.

Flexible – Proposal R1v3 allows companies to choose their own format for the transport of real-time text within their closed or proprietary systems as long as it is reliable and interoperable with public SIP and IMS systems using RFC 4103.

Opens up communication – Proposal R1v3 allows people using text to be able to communicate with most mainstream callers; allowing text users to call friends, family, co-workers, and strangers without friends or family having to own special telecommunication devices and without them having to use relay services with it's loss of privacy.

Allows direct communication – By allowing people to communicate directly with each other in voice and/or text as needed, more personal and private communications are possible.

Promotes international harmonization – Proposal R1v3 uses international standards, so that companies can use one solution that will work in all countries, rather than the multiple standards used for TTYs and textphones today. This will also allow all of the groups who need to use text (see above) to be able to talk internationally as voice callers do today.

Supports character- and sentence/message-based text during a call – Real-time text transport standards like RFC 4103 allow users to send text character-at-a-time as it is typed for real-time conversations, for relay service use, for captioned telephone, and for emergency communication. RTT transport formats (and Proposal R1v3) **also** allow users to select (at ay time in a call) to switch to message mode – where the text is accumulated and only sent when the user is ready – for those who need or prefer this mode at times or always.

No increased phone hardware costs – Proposal R1v3 only requires real-time text support on phones and other terminal devices that in their design have text generation and display capabilities for other purposes.

Proposal R1v3

1. **Wherever voice communication over IP (VoIP) is supported, reliable and interoperable real-time text shall be available and supported along with voice on the same call.**

    - **Exception:** Terminal equipment that has no text generation and display capability shall not be required to generate or display real-time text.

    1.2. Real-time: Text shall be transmitted within 1 second or less of being generated and be transmitted as data, not tones.

    1.3. Reliable: There shall be less than 0.2% character-error rate (~1% word error rate) within the system and its gateways. (ITU-T F.703)

    1.4. Interoperable: One RTT transmission standard shall be specified as the required transmission format for each network, and all terminal and network, software and equipment, including gateways, connected to that network shall support that required transmission format.

    1.4..1. For SIP based networks, RFC 4103 shall be supported.

    1.4..2. For IMS based networks, RFC 4103 as specified in TS 126 114, TS 122 173 and TS 134 229 shall be supported.

    1.4..3. For all other networks, the company or organization who defines the majority of the specifications for that network shall specify the required real-time text transmission format that shall be used for that network.

    1.4..4. Where any network connects to another network and interoperates with voice, a common format for real-time text shall be negotiated for each call, or the gateways shall transcode all real-time text into the required real-time text format for the other network, maintaining the voice+text call integrity.

    NOTE: Other transmission formats may also be used on any network by any software or equipment as long as the specified required format is also supported and available for use by any caller.

    1.5. Real-time and block transmission support: Terminals shall provide the ability to transmit the text in real-time, and shall also allow the user to optionally hold text before transmission in block, (or "message") style in order to check and edit the text before transmission.

    1.6. Emergency real-time text default: If in block mode (message style), the mode shall automatically switch to real-time text mode when an emergency number is known to have been dialed.

    1.7. Single call: The user shall only have to make a single call on a single device in order to have voice and real-time text on the call.

    1.8. On the Same call: It shall be possible to initiate a call as voice only, RTT only, or voice and RTT, and to add voice or RTT to a call at any time during the call.

    1.9. Simultaneity and concurrency: It shall be possible for the users to use both real-time text and speech at the same time and in both directions (4 information flows) simultaneously. (If there is a PSTN leg in the call, then simultaneity is not required on that call.)

2. **TTY support:** TTY support shall not be required for VOIP networks or terminals. Where VoIP networks connect to the PSTN for voice, the gateway must transcode between the IP based real-time text and the TTY on the PSTN.

3.) Frequently Asked Questions

3.1 Questions About Real-time Text Usage

Q1) If users have IM, why would they want real-time text?

This question comes up often because IM has become widely used, and has been considered synchronous communications when compared to email. The chart below demonstrates the many advantages of RTT compared to email, SMS, and IM:

Email	+ Allows long messages + Can send when receiver not on-line + Complete interoperability (all email programs work with all other email programs) - Delivery is reliable but irregular and can be slow - Spam blocker or spam load can cause email to be lost
SMS	+ Can send when receiver is not on-line + Today SMS is interoperable – and most are also interoperable with email - Delivery is irregular and can take quite awhile sometimes - There is no guarantee of delivery (messages are dumped in overload or long delay situations) Note: as IP comes to phones, SMS is predicted by many to disappear in favor of IM which is free (though not currently interoperable)
IM & Chat	+ IM is delivered immediately (usually less than a second after the transmitting person sends) + Individual messages are generally not long but are not limited like SMS + Chat offers good multiparty functionality - In close or intense conversations, IM causes one user to wait with a blank screen, wondering what the other person is typing and when it will come. When the message is long or the typist slow this can lead to frustration or sending of another message. - Crossed messages can occur and lead to misunderstanding and loss of time. (Crossed messages are where a person sends a second message before you finish answering the first. In an emergency situation, a panicked caller may ask a second or third question if there is no immediate visible response from the 9-1-1 call-taker. This can lead to confusion, crossed answers, and error. - Typically, can only send messages when both sender and receiver are on-line (though some store messages and wait until both parties are on-line to deliver) - IM is not currently interoperable so messages cannot be sent from one system to another. There is software that enables logging into multiple IM systems at the same time – but it does not allow conferencing and other features of the individual IM systems. - IM cannot be used with captioned telephone relay services. (Many IM have voice but IM is not real-time except for AOL's IM with real-time text option.) - IM does not have the same 'wiretap' protections as voice phone calls do.
RTT	+ RTT is immediate: instant delivery and instant view of letters as they are typed + RTT starts with an alerting call and requires both parties to be on-line at the same time, like a phone call. Any pause in devotion to the call is seen as a stop in the flow of characters. - RTT displays errors in typing (unless used in block transmit mode) + RTT typing errors can be corrected after transmission. Erasure works across connection. + RTT is like voice and signed languages in that it is continuous and there is no waiting to see what is being said – or for the receiver to read the sender's message after completion. + In time-limited situations, RTT can save time. The receiver can see the text as it is being typed, and when they see that what has been typed is sufficient, they can interact to confirm – shortening message and eliminating typing that is not needed.

	+ Multi-party functionality is possible + RTT can look like IM. RTT systems CAN send character-by-character or line-by-line at the user's choice (though it should be character by character in 9-1-1 situations). + RTT systems provide the real-time text required by captioned telephone relay services. This mode presents voice and text transcription simultaneously for people who have voices and want to use their residual hearing. . **RTT also has advantages in emergency communications.** + RTT allows for the efficient exchange of information and a continued sense of contact. + With RTT, there is less risk of crossed messages because the receiver can read the other person's message as it is typed. There is no waiting or looking at a blank screen. + With RTT, all messages are delivered even if they are not finished. For example, the following messages would get through successfully with RTT: "I think I'm having a heart att__" "Help. My ex is breaking into my hous__" By contrast, no part of the above messages would be received using IM or SMS if, while the caller is still typing, the emergency prevents that person from finishing the message and pressing the send key. + With RTT, emergency call-takers can view the message as it is being typed and respond, refer, interrupt, or guide the information being sent to speed up communication and make it more helpful to emergency responders. . **Under Proposal R1v3:** + RTT would be as interoperable as voice. + RTT would always be available where voice is available, so real-time text could be added to any call on any phone that had a multi-line display to make information clear to listeners who are hard of hearing. + Because RTT will be part of the 'phone' call, it can be protected to the same extent that the voice content of calls are protected (wiretap restrictions, etc.).

Q2) How would this work with relay services?

As IP relay services move to the new FCC common platform, all of them will support real-time text. IP text relay services would support a real-time mode between the text user and the communication assistant. This would make it easier to provide simultaneous use of text and voice through relay services (known as voice carryover (VCO) and hearing carryover (HCO))., because it uses standards-based voice and text for bridging the two legs of the call.

Proposal R1v3 also allows older people who are deaf and can speak (and who want to use voice carryover) to use any VoIP phone with a multi-line display. This would eliminate the need to purchase special and more expensive textphones or to master use of a computer.

People using videophones that support real-time text could also use text as needed during a sign language call with a video relay service (VRS) to supplement sign language communications for numbers, codes, URLs, etc. This combination is called total conversation.

Q3) Can Proposal R1v3 support captioned telephone calls?

Yes. In fact it would allow captioned telephone to be used with any phone that has a display on it. One would not need a special phone. This would include captioned telephone relay services but would also allow commercial captioning of phone calls in the same or in different languages (see Industry Issues and Benefits section that follows). It would also allow bi-directional captioned telephone where two users (with or without disabilities) could both see captions for each other's speech.

Q4) Could people (particularly friends and family) with computers, tablets and smartphones use them to communicate with real-time text users?

Yes. There would be a number of ways that people can use computers, tablets and smartphones with real-time text. This includes communicating between such devices and communicating between such devices and phones (wired and wireless). In fact, any time you can use voice between two modern IP devices you would be able to use text between them as well,
with or without voice. Some real-time text options based on computers, tablets and smartphones would include:

- VoIP programs on computers tablets and smartphones ("soft phones")
- IM programs that include voice (and therefore real-time text, as well as IM)
- Real-time text programs
- Videophones/programs that include real-time text (also called Total Conversation phones)
- Web pages that provide video+audio+text communication

Q5) What if I use text prediction (e.g.T9) on my phone keypad. Is that allowed?

Yes, if the phone supports text prediction (e.g. T9), text could be sent in a word-by-word fashion under this proposal (rather than one letter at a time) since with text prediction the correct text for the word is not created until the user finishes pressing all of the keys for the word.

Q6) Is speech recognition input allowed?

Yes. If a phone (or computer) had speech recognition already on it, it could be used to send real-time text. With speech recognition, the text is often not generated until entire words or sometimes phrases or sentences are recognized. In this case, the text would be generated in those chunks and, as long as it can be automatically sent as it is generated and not accumulated, it would be real-time.

Q7) Can real-time text be combined with IM?

Yes. For example, AOL has combined IM with real-time text in AOL's Instant Messenger 6.8, and there is an extension to the XMPP IM standard to add the real-time text feature for the benefit of that environment.

In addition, under this proposal, any real-time text format can also be used to send message-at-a-time communication rather than just letter-at-a-time communication. Any real-time text format can be programmed to also allow the user to accumulate their text and then, when they are ready, send the whole message at once in a rapid sequence of packets of characters.

Proposal R1v3 only requires that every phone *be capable of* sending text in real-time. It does not require that all communications occur in real-time communication mode. The user should be allowed to choose between real-time text and message-based text on any particular call.

Q8) Could IM and other IP text users (who don't have a VoIP subscription) communicate directly in real-time text to hearing VoIP customers under Proposal R1v3?

Yes, in most cases. It would take a gateway between the IM service and the VoIP.

If the IM or IP text users were on a system that supported voice and connected to the phone system (as many/most do already) then Proposal R1v3 would mean that there would be a gateway that could translate the real-time text feature of the IM+Voice service.

If the IM they were using had no voice support whatsoever, then Proposal R1v3 would not apply and they could communicate with VoIP users only if there was an IM to VoIP gateway for that IM. This is not technically difficult, but would be needed and is not currently required.

Q9) Would users who are blind have access to real-time text under Proposal R1v3?

Since real-time text would be real text and is based on Unicode, it would be accessible and readable by any software the person who is blind is using to have the other aspects of their phone voiced to them. On a phone call this would be by demand so that text is only voiced when asked to. This also facilitates access by people who are deaf-blind

3.1 Questions of a Technical Nature

Q10) When you say a company can use any real-time text standard they wish within their own systems, what exactly do you mean?

Companies may find that there are technical or cost reasons for them to use a particular technology to implement real-time text within their systems. They may even want to use a proprietary method for transport. With Proposal R1v3, any method can be used to provide the real-time text capability within a system as long as it is reliable, and it transcodes the RTT into the RTT format of the other systems where their systems connect to other transport/call-handling technologies.

For example a company might have a comprehensive communication product that supports Voice + Video + IM within an enterprise. It would also allow calls out to other IP phone systems within the enterprise or outside. With Proposal R1v3 the company would provide real-time text in conjunction with their voice system. This might be separate from their IM or it might be a real-time text feature on their IM. They could use any technology or method they choose for the real-time text within their system as long as it was reliable, and, where they connect to other IP phone systems, they would convert their real-time text to the RTT format of the network they connect to. For example, if it was a SIP based system it would convert to RFC 4103. If it connected to the PSTN it would convert to TTY.

Q11) Is a Common Real-time text Interconnection Standard really needed?

Different approaches have been proposed for transporting real-time text. Because some are incompatible with others, a coordinated approach is necessary to ensure RTT interoperability.

Unless there is one interconnection format that is supported by everyone, the only way to ensure that everyone can connect to everyone else would be for each company to support all of the other companies' formats. Because new formats will need to be allowed in the future, it is not technically possible to support all formats going forward.

Q12) Which standard would be used as the required real-time text format?

Although there is not yet a universal agreement on which standard to use, the standard being overwhelmingly chosen by any company or group actually implementing or planning to implement RTT over SIP (including IMS) is RFC 4103. RFC 4103 is cited in most planning documents for next generation IP telecom and emergency response systems and is currently the most widely adopted and supported format for real-time text. (See References and Extracts Section).

RFC 4103 is therefore proposed as the interconnection standard in Proposal R1v3: Specifically:

- IETF SIP for call control,
- ITU-T T.140 for real-time text presentation, and
- IETF RFC 4103 for the real-time text transport.

For networks other than SIP and IMS, the controlling or defining entity for that network would determine the RTT standard (that meets the requirements in Section 2) that would be the required real-time text standard for that network.

Once established, the existence of an interconnection standard for each network provides companies a way to ensure interoperability of their products with other terminal and network equipment/software on the network.

This approach also provide flexibility to companies or organizations creating other closed or open networks. They can choose any format that meets Section 2 requirements for their network, and just provide transcoding as needed between their format and the format of the network they can connecting to.

Finally, since all products on a network are required to support the required RTT format , but not restricted to using the required RTT format only, products can use other real-time text formats when they work, and yet fall back on the required RTT interconnection format when the call is not successful with the other format.

Q13) Would it be better to keep the selection of a proposed interconnection standard open until all interested parties have made a choice?

The IP voice communication infrastructure is rapidly being deployed. In addition, a number of industry groups have already decided on RFC 4103 as their preferred real-time text transport formats, including the access to emergency services.

However, without officially establishing this as the interconnection standard there is

no way to ensure interoperability between these systems and individual products that may be introduced and used on these systems. The sooner this common standard is adopted, the less expensive it will be to support throughout the system, and the less likely it will be that companies will have to retrofit or update an installed base that is being created. Waiting only makes it harder and more expensive for everyone – and delays availability to those who need text to communicate. Waiting also raises the potential that companies will develop products and features that will be incompatible with the standard chosen, creating more expenses associated with retrofitting.

The use of common required transmission formats and codecs is already the practice of all telecommunication systems today. This is no difference from the case for real-time text except that industry has selected and enforced them for voice systems, but not for real-time text. In the absence of a requirement from industry for real-time text support in all voice systems, and to allow companies to move forward with deployment of real-time text, the real-time text for SIP and IMS are defined and required in proposal R1v3– and are defined to be the standards already specified for those systems (but not required) by industry.

Q14) Would it add cost to every voice communication product to require that it support real-time text?

The proposal would not add any hardware costs to support real-time text because it only requires products to receive and display real-time text if they already have a multi-line display (or software designed to run on a multi-line display); and it only requires products to send real-time text if they already have a mechanism for generating text for another purpose.

While this proposal would require initial software costs to develop, test, and connect a software text codec into their device software and to connect display and text generation software to the codec, both open source and commercial codecs and reference designs for implementation are available for this purpose.

Q15) Are there real-time text standards today?

Yes. Details can be found in the Real Time Text Standards section of this proposal.

Q16) Are these real-time text standards in use today on network and consumer products?

Yes, some are. For a listing of currently known products that use the standards, see Products using Real-Time Text section of this proposal.

Q17) Why not just use TTYs with VoIP and the new IP networks?

There are a number of reasons for not using the older TTY standards to transmit text on the new IP networks. They include:

- TTY is antiquated, even for the PSTN. Its technical and functional limitations – including its slow speed and half duplex mode – are well known. There have been no new TTY models introduced in years.
- TTY tones do not travel well using IP audio compression, transmission and repair techniques without introducing text errors.
- Many TTY standards (such as Baudot, used in the United States) do not include all of the characters used in modern text communication. This makes it hard for users to communicate some things like URLs, email addresses, etc.

- It is difficult to communicate with people using new phones, tablets, computers, etc. (The phones, computers, etc. would all need special tone decoding and TTY translator programs which use different data formats than TTYs.)
- Textphone (TTY) standards vary between countries, making international communication using textphone devices difficult or impossible.
- If TTYs were used as the standard, all devices would have to support its antiquated format – or people who are deaf would be limited to communicating solely with those individuals who also have TTY-compatible devices. This would significantly limit the sphere of people with whom they would be able to communicate in text (to a much smaller group than those who can use voice). It would also require friends, relatives and others to continue buying specialized equipment to communicate with their TTYs. The consequence would be increased reliance on relay services (rather than supporting direct text communication), with a resultant increased cost imposed on all telecom users.

Q18) Do all PSTN to VoIP gateways need to support translation of real-time text to TTY?

No. See Industry Aspects section of this document for a longer discussion of this topic.

Q19) Will real-time text work with normal SIP security measures?

Yes. There are a number of technical methods available for securing SIP calls, both for the call control security and the media security including real-time text. For the media, all agree that a method called SRTP, specified in IETF RFC 3711, shall be used.

Q20) Will real-time text be blocked by network security measures and firewalls?

Blocking of real-time text may occur where security or IT management personnel are not aware of real-time text and where it is not required to be supported. By using a SIP-aware firewall, the firewall understands the media stream setup and keeps the media port open just during the call and directed to the terminal that set up the call. If Proposal R1v3 is adopted, real-time text should be treated as a standard part of every SIP call and should not be blocked. The wide-spread used principles for pass-through of VoIP calls work also on calls with real-time text. Regular use of real-time text by all callers (see next section) will also reinforce requirements to have proper pass-through for the real-time text part of the calls.

Q21) Would non-VoIP legacy cellular (GSM, CDMA, etc.) be affected by R1v3?

No. Non-IP based legacy cellular (GSM, CDMA, etc.) would not be covered by Proposal R1v3.

Q22) How would this apply to DTV and IPTV based telecommunication?

With the advent of digital television and IPTV, there are efforts to create IPTV based telecommunication (video phone calls). If these calls are carried over IP networks then they would fall under Proposal R1v3 and real-time text capability would be needed in parallel with the voice. Again, they could use any format for the real-time text as long as it meets the requirements in Section 2 and is converted to the interconnection standard of the network it connects to.

4.) Benefits to Users

Proposal R1v3 would allow text communication on the Internet and VoIP telephone networks to be as fluid, interoperable, reliable, and ubiquitous as voice communications are.

It is important to note that text communication is used by different groups of people who have different functional requirements for accessibility – and who may need to use text in different ways. Some of the communication requirements for these individuals can be met through IM, SMS or real-time text alone while for others, concurrent real-time text and speech transmission is needed.

> NOTE: In reviewing the needs of different groups for real-time text, it is important to remember that real-time text systems can allow users to choose between communicating character-based (for conversational typing), word-based (for T9 or ASR), or sentence-based (for message-like communication).

4.1) The benefits of Proposal R1v3 for individual populations:

- **People who are deaf and who rely on text for both expressive and receptive communication …**
  - … would be able to use any standard phone or device that already has a multiline display and some ability to generate text for another purpose (for contacts, or speed-dial, etc.).

- **People who are deaf and communicate primarily in a signed language …**
  - … would be able to use real-time text to supplement signing in VRS or video peer to peer calls to more easily communicate addresses, serial numbers, difficult-to-spell names, etc.

- **People who use a captioned telephone service or voice-carryover in a relay service (e.g., late-deafened persons, hard of hearing persons) …**
  - … could use almost any VoIP phone to receive text because almost all will already have a multi-line display.

- **People who are hearing but have speech disabilities …**
  - … could use any standard phone or device that already has some ability to generate text (for contacts, speed-dial, etc.) and can communicate with anyone who has any standard VoIP phone with a multi-line display.

- **People who are hard of hearing …**
  - … could have the speech of the person with whom they are talking supplemented by text if misunderstanding arises on an IP voice call (so long as the other party has a phone that can generate text for any reason); and
  - … could use captioned telephone relay services using almost any standard VoIP phone or device they encounter because most have a multi-line display.

- **People who are deaf-blind …**
  - … would need a device that has a tactile display (or allows one to be connected) but could communicate with anyone else without requiring the other party to have a special phone or device (as long as it had a multi-line display and could generate text).

- **People who do not have any disability – but would like to communicate with any of the people above …**
    - … would be able to communicate with those individuals without having to buy or carry any special devices. They could use their standard phone, which could be almost any portable VoIP phone, desk phone, smartphone, tablet, or computer which has a text display and/or generation capabilities for other reasons.

- **All of these groups …**
    - … could use many or most standard phones used by everyone else. This will be especially important in an emergency, if the individual's phone is not with them, or not functioning. In this situation, the person could use almost anyone's portable VoIP phone that is available. The ability to use any mainstream phone also means individuals will not have to pay high prices for specialized devices, or go through the difficulties of finding "specialty" phones.
    - ..because they can use mainstream phones, they could take advantage of special deals and bundles available for mainstream phones. They could also purchase used phones or use free, hand-me-down phones.
    - ... could use real-time text for emergencies with all of real-time text's advantages in that type of communication.
    - … could have direct (private) text (or mixed voice and text) conversations with others – without involving relay operators and without the other party having to buy or carry special phones or devices.
    - … in a disaster, could communicate with most anyone they need to in order to find their children or loved ones – without having to worry whether the person has a special phone with which they can communicate (because they would be able to communicate in voice and/or text with most any VoIP phone the other person might have).

Appendix 2 provides details on the requirements for each group as well as their benefits.

5.) Industry Benefits and Considerations

5.1) Ability to use real-time text to introduce new mainstream features and capabilities.

Because real-time text would be interoperable and implemented using standard IP technologies and methods, it creates the opportunity for companies to implement real-time text in a way that can contribute to the public good and provide them with new marketable mainstream functionality as well. Some potential new features would be:

1. On international calls, participants could choose to have the call captioned in the language of their choice.

2. A phone could have a feature that would allow a person to listen to one call and watch the captioning for a second call that was on hold.

3. A person could turn on captioning when they were interrupted in their office so that they could catch up when they were finished with the interruption. Similarly, people could take a health break and catch up when they get back.

4. A 'queue' query feature could be implemented by a teleconferencing company that would allow any participant to type "q+" to raise their hand (without disrupting the call). They could also type "q?" to have a waiting list sent to them so they can see where they are. This could even be done by people using the captioning feature.

5. In voice mail, the 'envelope' information including name of the caller, return number (from caller ID), length of call, time of call, etc., could be sent and viewable on screen while listening to voice mail.

6. If not otherwise being used for text conversation, it might be used to provide general advertisements.

7. Text on a screen might be a much friendlier way to while away time than listening to music on hold. Perhaps a scroll of today's headlines. Or of the company news or new products.

8. On a phone call, real-time text can be used to ensure that booking numbers, phone numbers, email addresses, codes and other information are conveyed accurately and without great effort. This can be of particular benefit on international calls or for persons with different accents.

9. If a manufacturer chose to provide real-time text along with an IVR's voice prompts, all users with a phone that has a display could quickly see all the choices in text instead of waiting for them to be read aloud – making IVR systems faster and less frustrating.

A few of these would require an on-demand speech-to-text service. But for executives, the cost of missing information (and their personal cost per hour) would far exceed these costs. And as speech recognition advances, the ability to use automated recognition with a moderately skilled person correcting it can move this feature to more users – but this can only occur if the capability is in the network and equipment.

5.3) Minimizing costs for access features

Proposal R1v3 does not require any additional hardware (displays, keyboards, etc.) beyond what a phone/device already has for other functions. Open source versions of the software (stacks and Codecs), as well as some commercial versions (including reference designs) needed to receive and send real-time text, are available. As a result, the costs associated with this proposal are primarily in the initial implementation for a company or transport technology. But even these costs will be kept down if real-time text design is incorporated in the beginning of the design process. At that stage, these costs should be merely a small fraction of the overall design costs, which can be amortized across all of the products sold – and carried forward to future designs. Indeed, given the capabilities of modern VoIP devices (the only type of devices to which this proposal applies), the small software (including firmware) changes needed should not add any significant cost to the manufacture of the products.

5.4) Ability to combine with instant messaging functionality

Proposal R1v3 allows real-time text to be combined with IM in a number of ways.

- First, Proposal R1v3 allows any format to be used within a system. So if an IM technology is used along with voice, the company can use a special real-time text format that is compatible or part of the IM format. AOL has done this with their AIM 6.8 IM client. The IM standard XMPP has an extension for real-time text (XEP-0301). This can be converted to the standard real-time text format where the text leaves the IM system.

- Second, any real-time text format following Proposal R1v3 would be able to function as both real-time text and as IM. There is nothing in Proposal R1v3 that prevents a company from allowing a user to send text as 'message-at-a-time' if the user desires. The only requirement is that the user also be able to send real-time text if they choose (and that the real-time text feature can be invoked as part of the voice call/connection and carried out in parallel with the voice communication).

- Third, a company with a voice communication function could implement both an IM feature and a real-time text feature side by side (as long as the real-time text feature can be invoked as part of the voice call and carried out in parallel with the voice communication).

Note: Proposal R1v3 does NOT require real-time text with IM products unless the products also support voice.

5.5) Concern about the responsibilities of carriers

In response to a question about whether carriers had to provide real-time text products or whether they just have to pass real-time text through their systems, under Proposal R1v3:

- Only IP transmission of voice is covered. Cellular wireless transmission that is not voice over IP is not covered under Proposal R1v3. It is covered under existing FCC requirements for TTY.
- Voice over IP carriers (including cellular carriers when they move to VoIP, but not before) would have to pass through real-time text.
- Carriers would not need to carry special VoIP phones. However, if a carrier is buying VoIP phones on the market, it should ensure the phones that it acquires which have

multi-line displays are capable of supporting receiving real-time text, and that phones that allow text entry will support sending real-time text.

- If a carrier designs or provides specs for a VoIP phone that does not have a multi-line display, it has no obligation to make it support real-time text.
- If a carrier designs or provides specifications for a phone that has a multiline display, it should also specify to the manufacturer that such phone needs to display real-time text that it receives on a call.
- If a carrier designs or provides specifications for a phone that has a text entry ability (and a multi-line display) for other purposes, it should also specify to the manufacturer that it needs to allow that text entry method to be able to be used to send real-time text on a call.

5.7) Concern about the availability of sufficient equipment

If Proposal R1v3 is implemented, there will be wide range of standard desk, wireless, smartphone, tablet and computer-based (softphone) VoIP phones that will support real-time text. In addition, specialized phones that are tailored to meet the needs of people with more severe disabilities (e.g., individuals who are deaf-blind) will also exist. Thus, it is anticipated that there will be sufficient equipment on the free market to meet the real-time texting requirements of all users.

NOTE: It is anticipated that real-time text will eventually be like captions on TV in that it will not just be people who are deaf who find this feature useful. Anyone on a phone call who wants to convey a credit card number, address, or other detailed information might find it handy to send it as real-time text rather than via voice when their phone has text capability.

5.8) Concern about every PSTN to VoIP gateway having to do TTY conversion.

Under Proposal R1v3, a carrier does *not* have to translate between IP real-time text formats and TTY formats at every gateway between the two. Rather, special gateways would be established, along with mechanisms that callers could employ to ensure that the calls are routed through these gateways.

Although the ideal way to handle this from a consumer perspective would indeed be that every gateway between VoIP and PSTN would translate between the VoIP real-time text standard format and the PSTN analog format, there is great concern among carriers that this is not technically feasible. There are too many such gateways and their technology level varies greatly. Also, real-time text support on the gateways may use up twice the ports on the IP side. For the relatively small amount of real-time text flowing through the gateways, this limitation of the ports would not be a problem if it only occurred on calls with text. But with calls coming from the PSTN side, it is not always possible at call origin to know whether the call will contain analog text (TTY). As a consequence, all calls would have to be set up to support text, or be able to re-invite with text, or have a mechanism to handle this. This is not too complicated for special gateways but it may be problematic for the installed base of network gateways, especially given the declining nature of TTY traffic.

Still, the fact is that everyone is not on broadband today. Until PSTN disappears, there will be people with hearing or speech disabilities who are on PSTN who still need to communicate in text. This is why support for analog text formats and their ability to communicate with the rest of the world is still needed.

Proposal R1v3 says that, instead of all gateways having to support TTY translation, special

gateways will be established that can handle the translation of text between analog formats and the VoIP standard format. Standard VoIP phones that do display IP real-time text would not have to support TTY text.

- For calls made from the IP side, calls that contain a text invite can be routed to one of the special gateways.

What does this look like to the text (IP) and TTY (PSTN) user?

- People who are calling others and using the real-time text in VoIP would just make regular calls and use real-time text as needed, and their phone would support it.
- Those still using TTYs to call other people on TTYs can continue to just call as they have in the past.
- For calls between TTY and real-time text on VoIP, either side makes just regular calls as they have in the past. As soon as the IP user types any text (initiating or responding), the system would detect it and invoke the gateway via the media negotiation protocols on the VoIP side. This can be done at any point in the call.

What does this look like to carriers?

Standard gateways do not need to support real-time text to PSTN translation. A small number of special gateways would be set up that can handle the dwindling number of analog TTY calls until people are migrated off of PSTN and calls would be routed to them.

What does this look like to manufacturers?

- Standard gateways would need to do nothing new.
- IP trunks in PSTN networks would still protect TTY transmissions with the standards they have set up to do this.
- Special gateways can be constructed from standard gateways with a special services server attached that would provide the special translation tasks to and from VoIP real-time text and PSTN TTY formats.
- VoIP enterprise companies, such as those who have a VoIP system that connects to the PSTN, would have a mechanism built into their gateway to translate outgoing VoIP real-time text into TTY. For incoming they could have a special TTY line(s) – where incoming TTY would be translated into VoIP real-time text before being delivered to the phones internally.

5.9) Benefits from using RFC 4103 as the internal real-time text format in networks.

In Proposal R1v3, it is proposed that RFC 4103 be used as the Common Real-time text Interconnection Standard. This is both because of its inherent characteristics (light weight, uses RTP, built-in redundancy, international character set support (Unicode), robustness, etc.) and because of the level of implementation to date.

In addition to being used as the interconnection standard, RFC 4103 also has many advantages for use within networks, particularly SIP networks. These include:

- It eliminates the need to transcode at borders of the network.

- It has a wider range of hardware to choose from than if a proprietary format is used.
- It is based on the same transmission protocol, RTP, as audio and video. That suits trunking technology that usually is only designed for RTP carried media. It is also declared in call setup in exactly the same way as other media, and it is thus minimal work to include it in a trunking technology.
- Open source and commercial codecs for RFC 4103 already exist.
- It has very little overhead and therefore is bandwidth efficient.
- Special phones can look into SIP to detect what media is setup, then invoke assistive features for text if it is offered in the call setup.

6.) Real-Time Text Standards and specifications

For the PSTN , there are different standards in different countries: TIA-825-A transmission of Baudot (TTY) is used in the U.S., while a half dozen other standards (e.g., TTY, V.21, EDT, DTMF) are used in other countries. (This has caused great problems with interoperability between countries.)[1]

For traditional SIP based Voice over IP (VoIP), there currently is only one real-time text standard format in use today for SIP based products other than gateways: That is IETF RFC 4103. This is also the standards cited in all of emergency communication recommendations, the FCC ACE reference platform, and the AT&T deployment.

For Web based voice communication, WebRTC provides voice and video using RTP, but has chosen to implement real-time text using a WebRTC data channel. Standardization is in progress.

For XMPP based voice communication (e.g. by the Jingle extension to XMPP) there is an XMPP standard for real-time text: XEP-0301 In-Band Real Time Text. This real-time text is implemented as an enhancement of the XMPP Instant Messaging environment with real-time transmission.

Briefly, the real-time text standards for IP networks include:

6.1. Real-Time Text Presentation Standards

T.140 – is the ITU standard for encoding real-time text. It is based on Unicode and can handle text in all languages covered by Unicode. It is used in most real-time text implementations.

6.2. IP Real-Time Text Transport Standards

RFC 4103 – is an IETF standard for real-time text transmission using ITU T.140 text encoding. RFC 4103 is specified as the real-time text format for:

- **IMS** (3GPP TS 26.114 IMS Multimedia Telephony, media handling and interaction)
- **IMS LTE** (GSMA ir.92 IMS Profile for Voice and SMS)
- **IETF emergency services** (IETF RFC 5012 "Requirements for Emergency Context Resolution with Internet Technologies", RFC 6881 Best Current Practice for Communication Services in Support of Emergency Calling)
- **NENA and EENA Next Generation Emergency services**. NENA NG9-1-1 i3 Technical specification and EENA NG112 LTD.
- **Cable TV Broadband service** (ITU-T J.161 Audio and video codec

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[1] The term TTY is used in U.S. telecom policies to refer to text telephone devices using Baudot TIA-825-A over the PSTN. While some other countries also use this term to refer to such analog devices, many other countries use the term 'textphone.' For the purpose of interpreting this paper in other places around the world, the term 'analog textphone' could replace "TTY," and "ITU-T V.18 and its sub-modes" could replace "TIA-825-A."

requirements and usage for the provision of bidirectional audio services over cable television networks using cable modems)

- **SIP terminals** (IETF RFC 4504 "SIP Telephony Device Requirements and Configuration," describing best practices when designing SIP based IP telephony devices)
- VRS services SIPFORUM TWG 1.0: Video Relay Service (VRS); US VRS Provider Interoperability Profile, SIPFORUM October 2015. http://www.sipforum.org/component/option,com_docman/task,doc_download/gid,786/Itemid,261/

6.3. IM (Messaging) Standard Where Real-Time Text Extension is Specified

XEP-0301 In-Line Real-Time Text - is an extension for the Instant Messaging standard XMPP (IETF 6120, 6121), making it possible to transmit Instant Messages in a real-time text fashion, maintaining all other characteristics of Instant Messaging. http://www.xmpp.org/extensions/xep-0301.html

Although real-time text is not required for IM without voice, XMPP can, with the Jingle extension, be used for voice calls. In this case, the XMPP based VoIP could use XEP-0301 to provide real-time text. Where XMPP connects to SIP and IMS, XEP-0301 would be converted to RFC 4103.

6.4. Standards for Transmission of Analog Text Formats Over IP Networks or Legs

RFC 4351 – is an IETF standard for real-time text transmission using ITU T.140 text encoding. **RFC 4351** is similar to RFC 4103, except that the real-time text data is sent (as text data) in the audio channel. RFC 4351 is also referred to as audio/t140. The RFC 4351 standard states (emphasis is in original standard):

"The payload format for real-time text transmission with RTP described in this [standard] is intended for use between Public Switched Telephone Network (PSTN) gateways and is called audio/t140c. ----

"The audio/t140c payload specification is intended to allow gateways that are interconnecting two PSTN networks to interleave, through a single RTP session, audio and text data received on the PSTN circuit.

" The audio/t140c format SHALL NOT be used for applications other than PSTN gateway applications. In such applications, a specific profiling document MAY make it REQUIRED for a specific application. The reason to prefer to use audio/t140c could be for gateway application where the ports are a limited and scarce resource. Applications SHOULD use RFC 4103 for real-time text communication that falls outside the limited scope of this specification."
https://tools.ietf.org/html/rfc4351

TIA 1001 – is designed to be used between PSTN Gateways. It was designed to transmit TTY across an IP leg in a PSTN network. It defines several methods for translating analog TTY into an IP friendly form and then translating it back again at the receiving gateway. A revision to RFC 4351 is included as an annex to TIA 1001

and is listed as one of the methods in the standard. Voice Band Data is another format listed. TIA 1001 is similar to ITU V.151, except that it supports Baudot TTY only, rather than supporting international analog textphone formats.

ITU-T V.151 – is designed to transmit international analog textphone formats across an IP leg (between PSTN gateways) in a PSTN network (i.e., it is an international version of TIA 1001 that supports international analog textphone standards).

ITU-T V.152 (Voice Band Data or VBD) – was designed to send data between gateways using voice band data. It does not use an IP text format.

To use VBD to send text between consumers would involve using VBD to transmit old PSTN analog formats (e.g., TIA-825 with Baudot text). This is a complicated process that requires many layers of coding and decoding by the products at each end. It also requires that old equipment or formats be supported by all new equipment.

For example it would require that users continue to use TTYs – or it would require new equipment to 1) encode text into an analog text format, then 2) put it into an analog transmission format, then 3) encode as VBD, in order to send. On the receiving end, it would require 1) decoding from VBD back into the analog transmission format, 2) decoding the analog transmission format into the analog text format, and 3) decoding the analog text format back into the standard text format use on all modern IP devices.

VBD also perpetuates the limitations of the old PSTN text communication formats.

7.) Implementations

Known implementations of Real-time text in telecom.

7.1) Phones and system components that have implemented real-time text based on RFC 4103 standard

- TIPCon1 open source softphone for real-time text, video and audio, using SIP for call control and RFC 4103 for real-time text.
  http://sourceforge.net/projects/tipcon1/

- Omnitor Allan eC, eCtouch and eCmobile apps for Total Conversation for Windows PCs, and Android and Apple IoS smartphones and tablets. Includes support for audio, real-time text and video, is used by deaf hard-of-hearing and deaf-blind people in Sweden. It uses SIP for call control and RFC 4103 for real-time text.
  http://www.omnitor.se/en/our-products/

- T-meeting TM-Touch, TM-Mobile, TM-PC and TM-9000 apps for Android and IoS smartphones and tablets and Windows PCs and dedicated hardware devices, All for Total conversation with audio video and real-time text. They all use SIP for call control and RFC 4103 for real-time text.
  https://tmeeting.com/products

- nWise myMMX apps for Windows PS, Android and Apple IoS smartphones and tablets for Total conversation with audio video and real-time text. They all use SIP for call control and RFC 4103 for real-time text.
  http://www.nwise.se/index.php?option=com_content&view=article&id=181&Itemid=320&lang=en

- Ivés Djanah apps for Windows PC, Apple IoS and Android smartphones and tablets for Total conversation with audio video and real-time text. They all use SIP for call control and RFC 4103 for real-time text.
  http://www.ives.fr/index.php/en/ives-solutions/ives-deaf-hard-of-hearing

- Accessaphone ipTTY. Windows softphone with real-time text using RFC 4103.
  http://www.accessaphone.com/iptty/

  Open eCtouch. Open source additions to the Linphone open source softphone to add real-time text to it, using RFC 4103 for real-time text. By Omnitor as participant in project RERC-TA.
  http://sourceforge.net/projects/ectouchopen/

- Asterisk IP-PBX open source soft IP exchange, includes support for audio, video and real-time text. It uses SIP for call control and RFC 4103 for real-time text.
  http://www.asterisk.org/

7.2) Phones and system components that have implemented real-time text based on real-time text technologies other than the RFC 4103 standard

- IPBlue VTGO 508 compliant softphone, with alternating real-time text and audio

intended for use together with a gateway featuring the Cisco text gateway protocol, and using IETF RFC 4351 for real-time text transmission.
www.ipblue.com/products_vtgo_508.asp

7.3) Network components

- Omnitor textphone gateways, between PSTN based text telephony and SIP with RFC 4103 based real-time text.

- Cisco IOS gateways providing the "Cisco text gateway" proprietary gateway protocol, using IETF RFC 4351 packetization of text in communication between TIA 825A TTYs and other similar gateways or terminals.

- Avaya system (introduced in 2003) based on RFC- 2833 for carrying descriptions of TTY tones through enterprise IP telephony networks and transcoding them on special text-enabled softphones. Proxy servers can also be used to split off the RFC-2833 information and send it to a TTY via a dedicated analog phone line.

7.4) Components

RTP text/T140 Library. Open source implementation of ITU-T T.140 and RFC 4103. One RFC 4103 part and one T.140 presentation part.
http://sourceforge.net/projects/rtp-text-t140/
http://sourceforge.net/projects/t140-pres/

7.5) Services and service elements available using RFC 4103 for real-time text

- IvèS total conversation communication services and systems with support for SIP and video, audio and RFC 4103 based real-time text
http://www.ives.fr

- nWise Total Conversation systems and services with support for video, audio and RFC 4103 real-time text protocol using SIP.
http://www.nwise.se

- Omnitor accessible SIP services and systems. SIP communication and additional accessibility features for SIP based Total conversation and real-time text.
http://www.omnitor.se

- T-meeting Total Conversation communication services and products for SIP calls with video, audio and RFC 4103 based real-time text.
http://tmeeting.com

- NG9-1-1 technology providers in USA, and NG1-1-2 technology providers in Europe are offering support for RFC 4103 real-time text in SIP based emergency calls.

- texttelefoni.se Web-based text relay service in Sweden with real-time text based on audio and RFC 4103 for real-time text in SIP calls. Procured by the Swedish Post and Telecommunications Authority PTS.
http://texttelefoni.se/en

- KPN Teletolk. Video relay and text relay in the Netherlands support SIP with audio , video and RFC 4103 - based real-time text. Procured by the Ministry of Innovation
https://www.kpnteletolk.nl/

- ORIGO. Video relay and text relay in France support SIP with audio , video and RFC 4103 - based real-time text. Procured by government authority. http://www.ives.fr/index.php/actualites-ives/123-origo-le-centre-relais-francais-pour-les-sourds-et-mals-entendants

- Bildetolk Video relay with Total Conversation in Norway, procured by the Labour and Social Authority NAV, Providing SIP support with video, audio and RFC 4103-based real-time. https://www.nav.no/no/Person/Hjelpemidler/Tjenester+og+produkter/Bildetolktjenesten

7.6) Services and service elements available using other technologies than RFC 4103 for real-time text

- Tess video relay service in Germany, providing calls with audio, video and real-time text.

- Swedish video relay service Bildtelefoni.net, using SIP and audio, video and a non-standard real-time text technology. Upgrade to RFC4103 is planned to 06/31/2016. http://www.bildtelefoni.net

- Beam messenger. Real-time text service for Apple IoS and Android devices, with proprietary real-time text protocol. http://www.beammessenger.com/

- Experimental XMP XEP-0301 based real-time text services or terminals. https://tap.gallaudet.edu/rtt/

8.) References and Extracts

Specifics on implementation of a real-time text feature in IP environments have been described in a number of standards and published documents. Some of them are referenced here, with an extract of the text.

Further descriptions of available components can be found in the real-time text forum group. at www.realtimetext.org.

EU COCOM 04-08 Report from the inclusive communications (INCOM) subgroup

"The favored solution is to adopt a single, IP-based set of preferred standards for all modes of accessible conversation facilitation. The set of preferred standards are IETF SIP for call control, ITU-T H.263 for video, ITU-T T.140 with transmission as specified in IETF RFC 2793 for text and ITU-T G.723.1 for audio. Nothing prevents implementations to include other coding standards, but the preferred ones should be maintained for interoperability."

Note: RFC 2793 is a compatible predecessor, replaced by RFC 4103.

○

IETF RFC 4103 RTP Payload for text conversation

"**Abstract**

This memo obsoletes RFC 2793; it describes how to carry real-time text conversation session contents in RTP packets. Text conversation session contents are specified in ITU-T Recommendation T.140.

One payload format is described for transmitting text on a separate RTP session dedicated for the transmission of text.

This RTP payload description recommends a method to include redundant text from already transmitted packets in order to reduce the risk of text loss caused by packet loss."
https://tools.ietf.org/html/rfc4103

IETF RFC 5012 Requirements for Emergency Context Resolution with Internet Technologies.

"Emergency calling must support a variety of media. Such media should include voice, conversational text (RFC 4103 [RFC4103]), instant messaging, and video."
https://tools.ietf.org/html/rfc5012

IETF RFC 4504 SIP Telephony Device Requirements and Configuration.

"Req-55: SIP telephony devices that include a display, or have a facility for connecting an external display, MUST include protocol support as described in RFC 4103 for real-time interactive text."
https://tools.ietf.org/html/rfc4504

IETF RFC 4733 RTP Payload for DTMF Digits, Telephony Tones, and Telephony Signals.

"The gateway or end system can change to a higher-bandwidth codec such as G.711 when tone signals are to be conveyed. See new ITU-T Recommendation V.152 for a formal

treatment of this approach. Alternatively, for fax, text, or modem signals respectively, a specialized transport such as T.38 , RFC 4103, or V.150.1 modem relay may be used."
https://tools.ietf.org/html/rfc4733

IETF RFC 4734 Definition of Events for Modem, Fax, and Text Telephony Signals.

"2.7.1. Signal Format Indicators for Text Telephony Legacy text telephony uses a wide variety of terminals, with different standards favored in different parts of the world. Going forward, the vision is that new terminals will work directly into the packet network and be based on RFC 4103 packetization of character data. In anticipation of this migration, it is RECOMMENDED that text carried in the PSTN by legacy modem protocols be converted to RFC 4103 packets at the sending gateway."
https://tools.ietf.org/html/rfc4734

IETF RFC 4351 Real-Time Transport Protocol (RTP) Payload for Text Conversation Interleaved in an Audio Stream.

"The payload format for real-time text transmission with RTP described in this [standard] is intended for use between Public Switched Telephone Network (PSTN) gateways and is called audio/t140c . ----

"The audio/t140c payload specification is intended to allow gateways that are interconnecting two PSTN networks to interleave, through a single RTP session, audio and text data received on the PSTN circuit.

" The audio/t140c format SHALL NOT be used for applications other than PSTN gateway applications. In such applications, a specific profiling document MAY make it REQUIRED for a specific application. The reason to prefer to use audio/t140c could be for gateway application where the ports are a limited and scarce resource. Applications SHOULD use RFC 4103 for real-time text communication that falls outside the limited scope of this specification."
https://tools.ietf.org/html/rfc4351

IETF RFC 5194 Framework for Real-Time Text over IP Using the Session Initiation Protocol (SIP)

"The Real-Time Transport Protocol (RTP) is the protocol of choice for real-time data transmission, and its use for real-time text payloads is described in RFC 4103."
....and...
"ToIP services MUST support the Real-Time Transport Protocol (RTP) according to the specification of RFC 4103 for the transport of real-time text between participants. RFC 4103 describes the transmission of T.140 real-time text on IP networks."
https://tools.ietf.org/html/rfc5194

RFC 6881 "Best Current Practice for Communications Services in Support of Emergency Calling (BCP 181)".
Specifying RFC 4103 for real-time text use in IP-based emergency service access.
https://tools.ietf.org/html/rfc6881

RFC 6443 Framework for Emergency Calling Using Internet Multimedia
Specifying RFC 4103 for real-time text use in SIP-based emergency service access.
https://tools.ietf.org/html/rfc6443

ETSI EN 301 549
Accessibility requirements suitable for public procurement of ICT products and services in Europe, 2015.
Specifies RFC 4103 as one mechanism for real-time text.
http://www.etsi.org/deliver/etsi_en/301500_301599/301549/01.01.02_60/en_301549v010102p.pdf

Access Board Section 255/508 refresh
Specifies requirement for RFC 4103 anywhare by: " Where ICT interoperates with Voice over Internet Protocol (VoIP) products or systems using Session Initiation Protocol (SIP), they shall support transmission of real time text that conforms to RFC 4103"
http://www.access-board.gov/guidelines-and-standards/communications-and-it/about-the-ict-refresh

NENA NG9-1-1 08-0003 Detailed Functional and Interface Standards for the NENA i3 Solution.
Specifying RFC 4103 for real-time text use in IP-based emergency service access.
"**4.1.8.3 Real-Time Text**
All call handling elements in the ESInet must support Framework for Real-Time Text over IP Using the Session Initiation Protocol (SIP) (RFC5194 [117]).
(To be renamed to STA-10)
https://www.nena.org/?page=i3_Stage3

EENA NG112 LTD
Specifying RFC 4103 for real-time text use in IP-based emergency service access.
http://www.eena.org/ressource/static/files/2013-03-15-eena_ltd_v1-1_final.pdf.

XEP-0301 In-line Real-Time Text
Specification for adding a real-time text mode in the dominating standard XMPP for IM.
http://www.xmpp.org/extensions/xep-0301.html

GSMA ir.92 IMS Profile for Voice and SMS
Specifies use of RFC 4103 for real time text, but makes it depending on national regulation.
http://www.gsma.com/newsroom/all-documents/ir-92-ims-profile-for-voice-and-sms/

FCC ACE platform specifications (formerly VATRP, an RFP from the FCC, August 14, 2014)
Specifies that real-time text must be among the supported media.
"4.2.7 Call signaling and media: [...] the VATRP shall implement the basic call flows specified in RFC 3665, including: at least H.264 and H.263-1998 at 30 frames per second for video; video resolution of 640x480 and better, audio (G.711 and G.722), message-based (SIP MESSAGE) and real-time text. [RFC 4103]"
Attachment A in https://www.fbo.gov/index?id=609c2196c044d005a639de874a055d6f

FCC EAAC User Survey
Survey indicating users need for communication modes, highlighting need for real-time text, voice and video and subsets of that in the same call.
https://www.fcc.gov/document/eaac-survey-review-and-analysis-report

FCC EAAC Report and Recommendations
The main output from EAAC, recommending real-time text access to emergency services.
https://www.fcc.gov/document/eaac-report-and-recommendations

FCC EAAC TTY Transition report
A document with details on why and how an IP based replacement and improvement of the TTY is proposed to be done, having SIP based implementations with RFC 4103 as the main alternative.
https://www.fcc.gov/document/emergency-access-advisory-committee-eaac-report-tty-transition

ETSI TS 101 470 Total Conversation Access to Emergency Services
A standard describing how Total Conversation access to IP based emergency services can be arranged,
http://www.etsi.org/deliver/etsi_ts/101400_101499/101470/01.01.01_60/ts_101470v010101p.pdf

3GPP TS 26.114 IP Multimedia Subsystem IMS; Multimedia Telephony; Media handling and interaction.

"7.4.4 Real-time text
The following RTP payload format shall be used:

- T.140 text conversation RTP payload format according to RFC 4103."
  http://www.3gpp.org/DynaReport/26114.htm

3GPP TS 26.235 Packet Switched Conversational Multimedia Applications; Default codecs

"RTP payload format for the ITU-T Recommendation T.140 text conversation coding is specified in IETF RFC 4103."
http://www.3gpp.org/DynaReport/26235.htm

3GPP TS 23.226 Global Text Telephony; Stage 2.

"IP Multimedia, supported by the IPMM subsystem, is a suitable environment for real time text conversation. It shall use IETF SIP, with text coded according to ITU-T T.140 and transported with IETF RTP-text as indicated in 3G TS 26.235. This allows conversation in a selection of simultaneous media, such as text, video and voice."
http://www.3gpp.org/DynaReport/23226.htm

3GPP TS 29.163: " IP Multimedia Subsystem (IMS); Interworking between the IP Multimedia (IM) Core Network (CN) subsystem and Circuit Switched (CS) networks",

Specifies how a RTT/TTY gateway is included in the path between RTT and TTY.
http://www.3gpp.org/DynaReport/29163.htm

ETSI EG 202 320 Duplex Universal Speech and Text (DUST) communication.

"C.3.1.1.2 Text transmission with RTP
...
For real time text, the RTP payload specification is RFC 4103 "RTP Payload for text conversation". This specification describes how to put the characters and other related information in packets, and also describes a procedure to make the transmission reliable."
http://www.etsi.org/deliver/etsi_eg/202300_202399/202320/01.02.01_60/eg_202320v010201p.pdf

ETSI ES 202 975 Relay Service Requirements

A.2.2 Real-time text Real-time text performance, protocols, and interoperability aspects are described in ETSI EN 301 549 [1], clause 6.2, which gives specific protocols for PSTN, SIP and IMS, as well as a mechanism for establishing interoperability by using other published and publicly available protocols.
http://www.etsi.org/deliver/etsi_es/202900_202999/202975/02.01.01_60/es_202975v020101p.pdf

ATIS-1000068 Support of TTY Service Over IP Using Global Text Telephony
August 2015. A standard for interoperability between TTY and RTT.
https://www.atis.org/docstore/product.aspx?id=28244

Appendix 1
TEITAC Consensus Language
(April 2008)

Proposal R1v3 builds on the consensus agreement reached between industry and consumers as part of the deliberations of the U.S. Access Board's Telecommunications and Electronic and Information Technology Advisory Committee (TEITAC) in April 2008. The TEITAC report consensus language provided below is quoted from http://www.access-board.gov/sec508/refresh/report/#656.

6-A: Real-Time Text Reliability and Interoperability

If hardware or software provides real-time voice conversation functionality it must provide at least one means of REAL-TIME TEXT communication where the following reliability requirements are met:

1. Products must use a REAL-TIME TEXT (RTT) system that meets the following requirements:
   a. RTT format must be a standard REAL-TIME TEXT format for the voice platform that is supported by all TERMINAL, router, gateway and other products on that platform;
   b. RTT format must transmit characters with less than 1 second delay from entry;
   c. RTT system must transmit TEXT with less than 1% Total Character Error Rate at the peak network traffic specified for intelligible speech transmission (TEXT must work on the network as long as speech does);
   d. The RTT system, together with the audio system, must support speech and TEXT in both directions in the same call session (and support speech and TEXT simultaneously in both directions in the same call session if IP based)
   e. RTT system must not utilize audio tones for transmission of REAL-TIME TEXT over IP. Note: this is subject to a waiver of the TTY support requirement from the FCC for systems that implement IP based RTT. Also subject to consumer acceptance of prefixes or phone numbers to direct TTY traffic to gateways capable of handling TTY translation.
2. Where products or systems interoperate outside of their closed systems, they must:
   a. If product interfaces with PSTN, it must use TIA 825A Baudot where it interfaces to the PSTN.
   b. If product interfaces with other VoIP products or systems (outside of a self-contained product-system) using SIP it must support transmission of TEXT as per XXX where it interfaces with other VoIP products or systems. Note: this is subject to a waiver of the TTY support requirement from the FCC for systems that implement IP based RTT. Also subject to consumer acceptance of prefixes or phone numbers to direct TTY traffic to gateways capable of handling TTY translation.
   c. If product connects to other products or systems using a protocol other than SIP it must use the standard REAL-TIME TEXT protocol that meets provision 1 above that has been established for that protocol.

Note 1: RFC-4103, TIA 1001, and MSRP (RFC4975) are being explored to fill the role of XXX. The intention is that XXX will be replaced by one interconnection format in all places it was used.

Note 2: All products may support and use other protocols in addition to these as long as they meet the 5 requirements of 5-B(1) above.

Note 3: A self-contained SIP system that uses the same real-time text protocol can be treated as a single product and can use any protocol internally as long as it supports XXX where the system-product connects to other systems or products.

6-B: Voice Terminal Hardware and Software

TERMINAL hardware or software that is capable of providing voice communications in real-time must comply with the following:

1. Receive only: If hardware or software TERMINAL provides voice conversation over IP in any form, and has a user interface with a multi-line display or a user interface that runs on devices that have a multi-line display, then that TERMINAL must display any REAL-TIME TEXT that is

received if it is received in the format for the voice and REAL-TIME TEXT system being used on the network on which it is installed.

2. Send and Receive: If TERMINAL hardware or software provides voice conversation over IP in any form, and has TEXT generation capability, then the TERMINAL must allow users to send REAL-TIME TEXT in the format for the voice and REAL-TIME TEXT system being used on the network on which it is installed.
3. If IP TERMINAL hardware or software does not provide REAL-TIME TEXT send and receive capability then the TERMINAL must support the addition of TERMINALS and TERMINAL peripheral equipment that support REAL-TIME TEXT functionality in conjunction with the voice call functionality, in the same location and with the same permissions for use as their voice TERMINAL. If the TERMINAL is in a public or shared area and not in an individual's private work area then the connection must be possible [without requiring system-administrator intervention]. Note: the "without system-administrator intervention" is a serious concern due to security issues, but removal would prevent people from connecting devices outside of their home system. Additional work is needed to address this issue.]
4. If TERMINAL is analog or TDM-digital wired TERMINAL then it must support the connection of a TTY via an RJ-11 jack in the same location and with the same permissions for use as the telephone and it must be capable of allowing simultaneous speech and TEXT conversation without interference or its microphone must be capable of being turned on and off to allow the user to intermix speech with text use.

Note 1: Provision of the RJ-11 jack may be accomplished through one of the following techniques:

a. provision of the RJ-11 jack on the telephone,
b. the use of a Y-adapter that allows both the analog telephone and the TTY to be plugged into the same line outlet,
c. having built in capability to support an RJ11 module that can provide a connection point for TTYs.

Note 2: The standard format for PSTN is TIA-825A. For SIP is it XXX. For other voice transport protocols the format is to be determined by the entity responsible for the voice transport protocol.

Rationale:

This provision, along with 6-A, allows people with disabilities to communicate using standard IP methods rather than continuing to support TTY within IP networks and devices.

Appendix 2
Details on Requirements and Benefits for Different User Groups

People who want to use TEXT IN BOTH DIRECTIONS.

- This includes
  - **People who are deaf and also cannot speak**
  - **All people who are deaf** when communicating with other people who are deaf
  - **Hearing people who cannot speak** – communicating with others who cannot speak or cannot hear

REQUIREMENTS FOR THIS GROUP	USER GROUP BENEFIT

• Need text in both directions
• Messaging or real-time text can be used. Both should be available to meet user needs and preferences. In emergencies real-time text should be used.
• Text only technologies and service plans (no speech support) are sufficient.
• People at both ends of the call must have means to create and display text on their devices.
 	
Technical Requirements
TEXT SEND
TEXT RECEIVE
[**NO VOICE SUPPORT** required]
	For these callers Proposal R1v3 would mean that
	– they could communicate with each other using REAL-TIME text or MESSAGE based text
	– Special devices would not be needed. Standard devices could be used – allowing these people to benefit from mainstream special prices, deals etc.
	  – They could use any device that has a multiline display and text generation capability.
	  – This would include almost all cell phones and some desk phones as well as all computers, smarthones, tablets etc.
	– Text only plans could still be used – but the array of potential communication devices would be wider.
	– In emergency situations – if they didn't have a phone (or had one with a dead battery or out of range) they could use almost anyone's phone because most phones would support text.

People who want SPEECH OUT and TEXT BACK

- This includes
  - **Some people who are deaf and can speak –** communicating a person who can hear where person who is deaf would like to talk and have text back.
  - **Late deafened adults** - who prefer to talk to the person they are calling and have the other person (or a relay service) text back to them.

REQUIREMENTS FOR THIS GROUP	USER GROUP BENEFIT
• The form communication for these callers is to speak and see text coming back. • Can use their speech and a relay service so that both they, and the person being called, do not need to type anything (especially important for elders). • Both people must have a means to display text. Neither needs to have a means to create text if relay is used. (A phone that displays incoming text but does not send text is sufficient). Technical Requirements **TEXT RECEIVE** **TEXT + VOICE** support needed on the call [**NO TEXT SEND** capability needed]	For these users – Proposal R1v3 would mean that – They could use almost all portable and stationary VoIP phones since almost all have a multiline display. – They could use smartphones, tablets or computers as well. – They could use relay services without any need for special equipment on either end. – Because standard devices could be used – these people can benefit from mainstream special prices, deals, etc. – In emergency situations – if they didn't have a phone (or had one with a dead battery or out of range) they could use almost anyone's phone because most phones would support text.

People who need to TEXT OUT but want SPEECH BACK

- **Hearing people who cannot speak –** often need to communicate by text but would like to hear the other person (and not make them type).

REQUIREMENTS FOR THIS GROUP	USER GROUP BENEFIT

• For direct communication – the person who cannot speak would use text – and the other person would to talk back to them.
• This would require this person's device to generate text but the person they are communicating with can use any device that receives text.
Technical Requirements
TEXT SEND (only) for person without speech
TEXT RECEIVE (only) – required by other person
TEXT + VOICE required by both people.
• Alternately – the person without speech could use a relay operator. The needs of the person without speech and the relay operator would be the same as the above. The person being called by the relay operator would need no text ability.
	For these users, Proposal R1v3 would mean that
	– they could communicate with each other using REAL-TIME text or MESSAGE based text.
	– Special devices would not be needed. Standard devices could be used – allowing these people to benefit from mainstream special prices, deals, etc.
	  – They could use any device that has any text generation capability.
	  – This would include almost all cell phones, some desk phones, as well as computers, smartphones, tablets etc.
	– Loved ones, friends or even strangers could get calls from them without any of them having to have/buy a special (higher cost) phone. They could use the standard bargain phones since almost all have a multiline display.
	– In emergency situations – if they didn't have a phone (or had one with a dead battery or out of range) they could use almost anyone's phone

	because most phones would support text.

People who need TEXT OUT and TACTILE BACK.

- **People who are deaf and blind** – communicating with any of the other groups mentioned.

REQUIREMENTS FOR THIS GROUP	USER GROUP BENEFIT
• For many in this group it is necessary to communicate in text in both directions. Messaging or real-time text can be used. Both should be available to meet user needs and preferences • The person who is deaf-blind would need a specialized device with a tactile display – or a (specialized) phone that allows received text to be sent to an external tactile display. • The person being called requires a device that can send and receive text (or make the call through a relay that has these capabilities). Technical Requirements **TEXT SEND** **TEXT RECEIVE** [**NO VOICE SUPPORT** required]	For these users Proposal R1v3 would mean that – Though they might need a special device or tactile display, they could communicate with others using REAL-TIME text or MESSAGE based text – Special devices would not be needed by the other person. Standard devices could be used – allowing the person who is deaf-blind to call almost anyone who has a device with a multiline display and text generation capability. – That would include almost all cell phones and some desk phones as well as all computers, smartphones, tablets etc. In addition, it is likely that tactile displays could be connected to an increasing number of standard phones meaning that even people who are deaf-blind could take advantage of the lower costs of mainstream phones and phone plans (be they standard voice plans – that would now include real-time text) or text only plans.

- **People (without disabilities) –** who want to communicate with any or all of the groups above.

An important stakeholder group that often gets left out is "all those who don't yet have disabilities" that would like to, or need to, communicate with those that currently do.

Proposal R1v3 would mean that this group would be able to use their standard phones and phone plans and be able to communicate with their friends, family, or strangers who have disabilities. Most phones would provide the abilities needed to allow them to communicate with all of the groups above without any special features or phone plan additions.