

Before the
Federal Communications Commission
Washington, D.C. 20554

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In the Matter of)
)
Transition from TTY to Real-Time Text) CG Docket No. 16-145
Technology)
)
Petition For Rulemaking To Update The)
Commission's Rules For Access To Support The) CG Docket No. 15-178
Transition From TTY To Real-Time Text)
Technology, And Petition For Waiver Of Rules)
Requiring Support Of TTY Technology)
)

**Reply Comments of the Rehabilitation Engineering Research
Center on Technology for the Deaf and Hard of Hearing, the
Rehabilitation Engineering Research Center on Universal
Interface and IT Access, and Omnitor**

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1. Introduction¹

The RERC on Technology for the Deaf and Hard of Hearing (DHH-RERC) is a project led by the Technology Access Program at Gallaudet University. The RERC on Universal Interface and IT Access (UIITA-RERC) is a project led by Trace Center at the University of Wisconsin-Madison. The RERCs are funded by the U.S. Department of Health and Human Services, Administration for Community Living, National Institute on Disability, Independent Living, and Rehabilitation Research, to carry out a program of research and development focused on technological solutions for universal access to systems and products for people with disabilities. Omnitor is a company in Sweden that focuses on developing accessible telecommunications solutions. The principal investigators of the DHH-RERC and UIITA-RERC have previously collaborated with Omnitor on the RERC on Telecommunications Access.

The RERCs and Omnitor (henceforth referenced to as the RERCs and Omnitor) would like to respectfully offer its reply comments on the FCC Order and Notice of Proposed Rulemaking on Transition from TTYs to Real-Time Text Technology. These reply comments contain additional information to comments from other parties and further details on the analysis of the NPRM.

¹ The contents of these comments were developed with partial support from grants from the National Institute on Disability, Independent Living, and Rehabilitation Research (NIDILRR numbers 90RE5020 and 90RE5015). NIDILRR is a Center within the Administration for Community Living (ACL), Department of Health and Human Services (HHS). The contents of this presentation do not necessarily represent the policy of NIDILRR, ACL, nor HHS, and you should not assume endorsement by the Federal Government.

2. Reply Comments

In general we are pleased to see the widespread support for RTT including among the carriers and manufacturers. There appear to be some misunderstandings and some places where changes are suggested that would defeat much of the impact of RTT and its benefit to people with disabilities, the relay services, the FCC, and to society as whole. To make the comments short and clear we will cite comments made along with our reply in a tight bulleted form. We will start with any comments made by more than one group and then group comments by commenter. We will try to give each comment a short name to help in locating them later.

Refrain from requiring standards

T-Mobile and CTA/CTIA/NCTA asked that no standards be required.

- The telecom industry cannot achieve any interoperability without required standards and does so routinely. It has had the opportunity to choose their own standard and require it for over a decade and have not done so.
- The IETF, the standards group for IP, SIP and most VoIP did create a standard (RFC 4103) for RTT and deprecated or archived (made historical) other standards so there is just one for RTT on SIP based systems.
- This standard was adopted and named in IMS and most every other SIP based telecom standard to date.
- However, carriers and manufactures still do not recognize it as the standard and ask for the standard to be unnamed - though no other is presented as the standard for SIP based system. Yet without a standard at least two companies who were ready to come to market years ago - were unwilling or unable to without knowing what the interoperability standard would be.
- We agree with the FCCs approach to name RFC 4103 as the safe harbor standard. Others can also be used as long as they interoperate with RFC 4103 when connecting to SIP based systems.

Application to all devices vs Narrowly

CTIA asks that the RTT rules be narrowly targeted and not include things like IoT.

- The telecom world is changing rapidly - and the name of the device that will be used for telecom will also change. The Smartphone could have easily been called the full feature PDA except for marketing. And the telecom transport format is already changing dramatically.
- IoT appears to have been chosen as an extreme to make a point, but it also makes the counter point. No IoT device will be subject to RTT unless it already has a screen and is already being used for two-way voice conversations at a distance. Something that does both of these is in fact a telecommunication device whether it is an IoT device or not. All smartphones are in fact part of the IoT universe.
- NOTE: In defining telecom, it will be increasingly important to **not** define telecom as being between two humans. In the very near future we will have AI voice interaction agents that will be completely indistinguishable from humans and will replace humans at the end of many (most) commercial telecommunication connections. It should not be possible to avoid all accessibility requirements by simply replacing people with these proto-people.

Application to WIRELINE

Although a number of commenters supported simultaneous implementation of RTT on wireline, several companies, including AT&T, CTA, CTIA, NCTA, did not support RTT on wireline any time soon.

However, if RTT is not supported on IP wireline then:

- Wireless callers will not be able to directly call wireline phones
- There will need to be many more RTT-TTY gateways
- TTYs will need to continue to be provided for wireline users if IP RTT will not work
- TTY communications will fail on IP wireline - as has been shown in IP wireless.
 - (We found it somewhat confusing when a company admits TTYs don't work on IP then advocates for delay in implementation of RTT on IP wireline.)

- Relay Services will need to continue to be relied upon to mediate between RTT callers and everyone else.
 - (If all IP wireline phones supported RTT, then anyone who needs RTT could call anyone (pizza place, doctor, babysitter, police, neighbor) and talk to them without the neighbors needing to have a special phone -- and without having to use relay services.)

Fixed Wireless

Of special concern is the category of “fixed wireless” that uses wireless technologies to provide fixed phone service to a home. In many of these applications, the phones inside the house remain analog (PSTN-like) but TTYs won't work reliably because there is no Gateway at the border to the house where it becomes IP.

In order to maintain equivalency of access, Fixed Wireless installations would need to meet the following requirements.

1. The wireless connection to the house must support RTT
2. The gateway at the house must provide WiFi connection to devices in the house to allow IP-RTT devices to be connected without requiring internet wiring to be installed in the house
3. The service plan must allow voice+RTT conversations with devices that are connected to the IP connection from the house (in the same way and at the same rates as calls just through the Analog to IP gateway).

OR - the gateway must be a full TTY to RTT Gateway at the house to allow TTYs to be used.

OTT or installed Apps (for providing RTT)

AT&T, Verizon, CTA/CTIA/NCTA, suggested the use installed RTT applications either temporarily or permanently instead of having RTT be build into the native telephony features.

We see this as very problematic for a number of reasons.

- First - see the OTT comments above.
 - Apps will not be on all phones - so relay must continue to be relied upon

- Users cannot directly call anyone - like voice callers can. And they cannot try a direct voice call and then just use RTT to supplement when not understood (other person won't have RTT).
- Users can't follow the lead of everyone around them but must use a different way to call
- When you click on phone numbers in web or other apps, it is the native app that pops up and makes the call
- Phone books, and other features are all tied into the native apps

We suggest that carriers and manufacturers focus on building the RTT into their native phone functions (only software is needed to do this) vs spending time creating different Voice+RTT applications. Users can still find Voice+RTT apps if they choose during any gap, but having manufacturers focus on them themselves will slow down and provide a more confusing path for users who need to learn one way and then another to make calls.

Access to all call related functionality (transfer, multiparty

AT&T claims that there are no standards for how to make call transfer, Multi-party calls and voicemail include RTT support with voice.

However,

- If needing RTT on a call means you can use it only for 1-to-1 live calls but
 - **you can't have your call transferred** - then how do you call most companies that have someone you talk to first before you get to the person you need to call.
 - **you can't use call menus** - then how do you call a store or company that has a robot secretary?
 - **you can't use voicemail** - then how do you call a company that sends all calls to voicemail - and then calls back those who don't hang up?
 - **you can't use conference calling** - then how do you sit in on all of the virtual meetings that occur at work?

- Furthermore there appear to already be standards in 3GPP for call *transfer* (i.e. TS 22.173, section 8.2.15) and *multi-party calls* (i.e. TS 22.173, section 8.2.13) that are part of the standards for the 3GPP Multimedia Telephony concept. Also note that TS 22.173 says in 4.1 “*When a supplementary service is invoked it applies to all media components of an IMS Multimedia Telephony communication.*” In addition, standards exist from service level through to technical level (TS 22.173, TS 23.173, TS 24.229). And these standards use RFC 4103 for RTT, as specified in TS 26.114.
- Multi-media answering machines are not standardized by 3GPP. However, it is an application and easy to extend to support more media than audio.

We also note that AT&T says on Page 17 that “*RTT OTT application will not meet: (5) transfer RTT calls and initiate conference calls using the same procedures used for voice communication; (6) use RTT to communicate with and retrieve messages from messaging, automated attendant, and interactive voice response systems*”

- We do not understand why an OTT RTT application would have any more trouble transferring or initiating a conference call than the build in application - unless the service is for IMS only and the OTT RTT application is not IMS. However OTT (non-native) RTT applications that are IMS could still be used.
- The same would be true for the ability to communicate with and retrieve messages from messaging, automated attendant, and interactive voice response systems.
- Their concerns here, and to the extent they feel it is over-burdensome to create solutions that work with OTT/non-native applications, however appear to reinforce the point we make about carriers and manufacturers not using APPS to fill in and delay building the functionality directly into the native call and call related functions of the devices.

Carriers not responsible for RTT on the hardware they use.

AT&T (P 14) commented that carriers not be responsible for RTT on hardware in their networks because manufacturers are in control of the devices.

- However, carriers have long been known to have large impact on phone design and often require certain features exist or not exist on the phones before they will allow them on their networks.
- The exemption should be allowed only if the carriers certify that they have not been able in the past to require certain software only features or capabilities be in phones or not in phones before they will put them into their lineups.

Dropping TTY support in Circuit Switched Mode when RTT is implemented in VoIP mode.

AT&T asks the Commission to vacate the requirement to support TTY in services and devices introduced after the deployment of RTT, but support backwards compatibility for landline TTY users. They also asked for obligations to support connected TTY devices to be dropped.

- We support dropping of requirements to connect TTYs to phones that support RTT natively on VoIP.
- However, while VoIP is rolling out, there are many times when VoIP is not available and phones drop back into circuit switched mode. When this happens voice callers continue to be able to make calls. But if TTY connectivity is dropped then there is no way for voice+text callers to make calls. For voice+text callers, it would be useful if there was the ability for the device itself to act as a TTY when there is no VoIP connection available. We do not expect phones to be able to handle a transition seamlessly on a call. But for individuals to have no text availability when in areas away from VoIP coverage, built in support for TTY would be useful. We do not suggest that the FCC require this - but it would be good to mention it as desirable.

Text presentation Requirements

AT&T on page 9 says that “Commission rules should avoid imposing mandates for specific RTT user features” and support “minimum functionality requirements to ensure that RTT replaces the functions of and allows for an orderly transition from TTY, such as

backwards compatibility (i.e., interoperability) with legacy TTY and simultaneous voice and text capabilities, other features of RTT should be consumer driven.” That “block mode,” simultaneous transmission of video, voice, text, and data, the transmission of emoticons and other special characters; and the ability to control settings, such as font size and color, conversation windows, and text location “should be neither required nor prohibited”. That they extend “far beyond the performance objectives in Rule Parts 6, 7, and 14.”

- We partially agree. The regulation should not be too specific with regard to form and format of RTT. And the requirements should not name every feature that can be thought of that would be helpful.
- However, not specifying essential features can lead to an implementation that is RTT in name only, and not useable or used.
- Many of these comments indicate a misunderstanding of the purpose of the features and their importance to different user groups. Because of their importance, they do not extend beyond Rule Parts 6,7 and 14. Treating each in turn:

AT&T: “Adding video to an RTT call session is especially problematic because it would require development of new user interfaces, triggers significant concerns about capacity in a service provider’s managed networks, and is more likely to increase error rates for RTT transmissions over Wi-Fi or other unmanaged networks.”

- Depending on how you interpret the FCC request - we partially agree with AT&T.
- IF it is interpreted to mean that video should be a required option on all voice calls, then we agree - this goes beyond providing an alternative to TTY.
- IF, however, it is interpreted as saying that RTT should be able to be used on a Voice+Video call, then it would not -- since TTYs could be used on a voice+video call over PSTN.
- Thus the requirement is not that video must be on all calls - but that if video is on the call- it should be done in a way that does not prevent RTT from also being used on the call.

- We also urge that the video be done in a fashion that is sufficient to allow accurate use of sign language including fingerspelling.

AT&T: “block mode,” ... “should be neither required nor prohibited”

- There is much misunderstanding of this feature. The goal is not to create a ‘messaging mode’ but to allow those with cognitive, language, and learning disabilities or for whom english is a second language, to be able to “pause instant transmission to allow review and editing” before it is presented to a stranger. This is important for some users but not others who do not have a problem constructing accurate language (albeit with typos).

AT&T: the ability to control settings, such as font size and color, conversation windows, and text location “should be neither required nor prohibited”.

- We agree that *local font size and color* are not part of the requirements here -- but instead should be covered under the regular accessibility requirements for the device. So they are important to some users but should already be covered in other regulations
- We do not know what “conversation window” means but
 - We agree that side by side vs intermixed etc. presentation formats should be left to designers and users - since what will work on different form factors and for different interfaces will vary.

There ARE some aspects of the interface functionality that are essential to use, especially by elders, and should be required.

1. RTT Window appearing automatically when RTT is received.

- It is essential that when RTT is received, the window or screen that will display it will automatically appear and display the incoming text. Without this both those less technically inclined, but more importantly, everyone else who is called, will not know that RTT is being sent, nor how to bring up its display.

- NOTE: A concern that RTT will be used for pushing advertising or other unwanted text can be addressed in the same fashion as voice calls with unwanted content. The users simply hangs up and/or blocks the number.

2. The Control for Showing the RTT window (turning on RTT in a call)

- The “RTT” control (to turn RTT on in the middle of a call) should be available with the same simplicity as other “in call features” such as MUTE, or SPEAKERPHONE, or ADD PERSON, etc.
- Without this, people who are having trouble being understood have no way to easily add RTT to a call. The result is that they use RTT or Relay for example in a preventive fashion on all calls because it is too complicated to use RTT as needed.

On point 78, RTT support in various forms of TRS

In stark contrast to the argumentation from Sorenson in their comments to the NPRM we find that all forms of TRS benefit from providing access for standardized interoperable RTT. This may be as one of the two main media in text relay, IP-relay and IP-CTS, or as an important complementing media in VRS services.

Sorenson comments: “ *RTT cannot replace VRS or IP CTS, and the Commission should not require that VRS and IP CTS providers incur the unnecessary costs associated with implementing RTT. ”*)

- In Sorenson’s comment - they seem to keep posing RTT as an alternative to the various forms of relay.
- However in no place does the FCC (or anyone else in their comments) talk about replacing VRS or CTS with RTT, but rather having RTT be used as one of the media along with speech or signing in the relay services. And we agree with the FCC in this.

- In fact CTS uses real-time-text and using RTT for the real-time text component of CTS would provide benefits to users, allow more CTS interoperability, and facilitate direct person to person communication.

Arguments for support of the view of the FCC in point 78, that TRS of all forms should provide access for RTT, is expressed from many sources in the comments, including from us, from VTCSecure and from Hamilton.

Implementation of RTT in all end user devices provides an additional opportunity to reduce cost and increase accessibility and interoperability compared to the current situation, when TRS terminals are specific to the services.

Two events, that occurred after the last comment date, are of note, and illustrate some advantages of having a standard RTT as a common part of different relay services.

One is the Swedish VRS provider Bildtelefoni.net's decision to provide, as of July 1st 2016, VRS using total conversation with video, audio and (RFC 4103 based) RTT. The RTT medium is mainly used for occasional communication between the interpreter and the user, especially for numbers, addresses and items that need to be remembered. Because it uses the same standard as used for the RTT based text relay service from Texttelefoni.se, (which also uses RFC 4103) users can, with the same terminals, select the type of relay service (Text or Sign) that fit them and the person they are calling best for the moment.

These terminals are standard and are supplied independent communication service providers. Even if they were provided by relay services any terminal+service can be used with any relay service, and for direct person to person communication. Also, since it is the same standard as mainstream devices, as video makes its way into mainstream devices, they mainstream devices could also be use directly both for Relay and for person-to-person communication.

This structure (where terminal/service providers are independent of relay providers and terminals can also be used for person-to-person communication) is what VTCSecure

argues for, and is worth considering also for US - a structure where the terminal communication providers have access to the iTRS system for routing calls and taking decisions on TRS use for each call without being TRS provider.

The other event supporting interoperable RTT and VRS is a draft standard that has been published in IETF that defines a standard interface between VRS user terminals and VRS services. This draft, which is currently US focused, is available at <https://tools.ietf.org/html/draft-vrs-rue-dispatch> and is based on IETF RFC 4103 for RTT, together with G711 audio and H264 video (with other codecs as possible in addition to these three required codecs).

RTT Definition - Transmit instantly

On page 58 in the NPRM, the definition of RTT includes "(3) transmit text instantly, so that each text character appears on the receiving device within one second of when it is generated on the sending device, with no more than 0.2 percent character error rate

CTIA was concerned that the proposed rule 67.2(d) only allows character-by-character keyboard input and no prediction, autocorrection, swyping etc.

- We suggest the addition of a note to the definition that says
 - NOTE: Some types of text generation result in text being generated in chunks of text such as voice recognition, swiping keyboards, text prediction, encoding etc. In these cases, all of the block of characters are generated at the same time - and must, as per the definition, be transmitted instantly upon generation.
- Because the phone manufacturer cannot control transmission delay, we suggest the definition be changed to “transmitted from the device within 500 ms of creation”.

Appendix A, final rule

§64.604 should be considered to be modified.

Current wording:

"(vii) TRS shall transmit conversations between TTY or RTT callers and voice callers in real time."

- Our concern here is that this must apply to both the caller and the called person. We therefore suggest the following edit or some other edit to achieve the same result.
 - Change "callers" changed to "users"
"(vii) TRS shall transmit conversations between TTY or RTT users and voice users in real time."

Respectfully submitted,

On behalf of the UIITA-RERC², DHH-RERC, and Omnitor:

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