This comment will focus specifically on Real-Time Text (RTT).

I am happy to read: "*NPRM, the Board proposes to require that ICT support RTT functionality whenever such ICT also provides real-time, two-way voice communication. This proposal represents a significant shift in approach for both the 508 Standards and the 255 Guidelines to better align with current technology*."

We have now ONE chance to ensure full and complete interoperability with Real-Time Text aligned with VoIP and Video over IP. It is crucial to ensure that RTT must not only be included with only VoIP, but also with Video calls/Video telephones. The concept of using voice, video and RTT is called Total Conversation. There is no issue using RTT without video, so it does not matter that video telephony is out of scope for this refresh.

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Question 8. *If the XEP-0301 standard is finalized, the Board is considering incorporating it by reference as an alternative standard for XMPP networks. We seek comment on the benefits, costs, and possible drawbacks associated with referencing this standard in addition to the RFC 4103 standard.*

Anwer: The drawback is that with 2 RTT standards, the developers and manufacturers will not understand which RTT standard to use and this will lead to interoperability issues. This can be prevented by clearly describing that XEP-0301 to be used for XMPP services and systems/networks and when connecting with a non XMPP system, that it <u>must interoperate</u> with RFC4103.

"*Commenters to the 2011 ANPRM noted that other standards aside from RFC 4103—such as XMPP and XEP-0301—were currently in use and could be referenced as specifications for ICT interoperability with VoIP using SIP*."

That is not advisable. XEP-301, while now a standard for RTT (no longer pending standard), it is specifically created as an improvement on XMPP/Jabber chats by adding the RTT component to it and should be limited to XMPP networks and systems!

"*Yet despite its potential benefits, the Board cannot incorporate XEP-0301 until it becomes a final standard. However, should the XEP-0301 standard be finalized before publication of the final rule, the Board plans to incorporate it by reference as an alternative technology to support transmission of RTT when interoperating with VoIP products or systems using XMPP. RFC 4103 would, in any event, be retained for ICT interoperating with VoIP products or systems using SIP technology*."

For interoperability reasons it is NOT advisable to use XEO-0301 for transmission of RTT when interoperating with VoIP products. It is perfectly fine for systems using XMPP but when an XMPP network connects with a non XMPP VoIP network/system. It MUST interoperate with RFC4103.

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The European standard, EN 301 549 would allow the use of multiple standards for RTT. As discussed in 4.6, Harmonization with European Activities above, EN 301 549 lists several standards for RTT, as well as an unspecified "common specification" for RTT. The common specification must indicate a method for indicating loss of corruption of characters. The Board seeks comment on whether other standards should be incorporated by reference. The other standards are:

- *ITU-T v.18, Recommendation ITU-T V.18 (2000) "Operational and interworking requirements for DCEs operating in the text telephone mode" (see EN 301 549 6.3.3(a)). This Recommendation specifies features to be incorporated in data carrier equipment intended for use in, or communicating with, text telephones primarily used by people who are deaf or hard of hearing.*
- *IP Multimedia Sub-System (IMS) protocols specified in TS 126 114, TS 122 173, and TS 134 229 (see EN 301 549 6.3.3(c)). ETSI TS 126 114, Universal Mobile Telecommunications*

System (which was referenced in the EAAC Report and Recommendation noted previously in Section IV.F.2) supports a "total communication" approach by establishing a minimum set of codecs and transport protocols that must be supported by all elements in the IMS system for video, real-time text, audio, and high definition (HD) audio. As noted previously, the Board decided not to require standards for video, audio, or HD audio in this proposed rule beyond the technical requirements set forth in proposed 410 (ICT with Two-Way Voice Communication). Both the ETSI TS 122 173 and ETSI TS 134 229 standards are still under development, and, therefore, cannot be referenced at this time.

[AvW] It is Total Conversation and not Total Communication.

Question 9. Are there sufficient net benefits to be derived from requiring ITU-T v.18 that the Board should reference it in addition to TIA 825-A (2003)? We are requesting that telecommunication equipment manufacturers, in particular, provide any data regarding potential costs related to complying with this standard. Are there suggestions for other standards which would result in the same level of accessibility?

Question 10. Are there net benefits to be derived from requiring more standards addressing multimedia than what we propose? The Board is requesting that telecommunication equipment manufacturers, in particular, provide any data regarding potential costs related to complying with the standards in EN 301 549 6.3.3(c). Are there suggestions for other standards which would result in the same level of accessibility?

I answer both 9&10 here: EN301549 does cover the Interoperability of Real-Time Text, but sadly it is as a result of the industry a too vague for manufacturers and developers. When you allow different companies to use different standards or specifying that the products and services using these standards (just) should interoperate with each other does NOT give the developers a clear statement of with what is needed to interoperate with.

An excerpt of EN301549 with my comments inline [AvW]:

6.2.3 Interoperability

Where ICT with RTT functionality interoperates with other ICT with RTT functionality (as required by 6.2.1.1) they shall support at least one of the four RTT interoperability mechanisms described below:

[AvW] this will confuse developers and manufacturers and leave too much room for alternative forms of RTT that breaks interoperability.

a) ICT interoperating over the Public Switched Telephone Network (PSTN), with other ICT that directly connects to the PSTN as described in Recommendation ITU-T V.18 [i.23] or any of its annexes for text telephony signals at the PSTN interface;

[AvW] In Europe, the PSTN text telephony is being phased out. So requirement a) is only relevant when interconnecting your VoIP network/service with a PSTN network. The question remains, how relevant is this in a few years for the US when the TTY is phased out as well? Any TIA 825-A (2003) device should work with any ITU-T V.18 regarding the FSK/Baudot part, since ITU-T V.18 has been harmonized with TIA 825-A. So the Board should reference ITU-T V.18 in addition to TIA 825-A (2003). A solution could be to use a transcoding gateway and no need to support PSTN text telephony on the all IP devices anymore. **(question 9)**

b) ICT interoperating with other ICT using VOIP with Session Initiation Protocol (SIP) and using real-time text that conforms to IETF RFC 4103 [i.13];

[AvW] This requirement b) is actually the MOST important requirement. RFC4103 must be considered as the facto RTT to interoperate with.

c) ICT interoperating with other ICT using RTT that conforms with the IP Multimedia Sub-System (IMS) set of protocols specified in ETSI TS 126 114 [i.10], ETSI TS 122 173 [i.11] and ETSI TS 134 229 [i.12];

[AvW] c) actually also included RFC4103, since RFC4103 is part of the IMS specifications (see ETSI TS 126 114).

d) ICT interoperating with other ICT using a relevant and applicable common specification for RTT exchange that is published and available. This common specification shall include a method for indicating loss or corruption of characters.

[AvW] This requirement d) will break interoperability, since it will allow manufacturers to implement any proprietary form of RTT and it will not be possible to interoperate with RFC4103 RTT.

This will also result in that when a platform or service provider wants to interconnect with a network using this proprietary form of RTT that they have to build a new transcoding gateway for <u>every type of RTT used by the network connecting with</u>. This will lead to a jungle of many different forms of RTT that are completely different and cannot interoperate with each other! This is a disaster for interoperability and will lead to unnecessary costs.
I do not say it is forbidden to use or build any proprietary or different standard of RTT (like XEP-301 for XMPP systems). But when a manufacturer or developer implements any kind of RTT, require that this kind of RTT shall always be able to interoperate with RFC4103 RTT when interconnecting with other networks and services. In this way, the other network will only need to enable that any RFC4103 RTT that enters will interoperate with the alternative form of RTT.

In short:

RTT type A -- network A-- transcode RTT A to RFC4103 RTT at border network A-- RFC4103/SIP---- interconnect between network A and B -- RFC4103/SIP -- transcode from RFC4103 RTT to RTT B -- network B -- RTT type B.

This allows network A to interconnect with all networks that have the same transcoding from RFC4103 to the proprietary form of RTT without having to worry about the other RTT forms.

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While it is a good idea to harmonize with EN301549, it is advised that the Access Board does maintain a clearer and stricter level of requirements, especially on real-time text. As described above.

The developers and manufacturers are urged to go for RFC4103 RTT as first choice when operating a full IP device and network. RFC4103 is already part of IMS, also used by the Dutch & Swedish Text and video relay and the Dutch emergency center (112/911), used by an increasing number of telecommunication companies as part of Total Conversation. Also RFC4103 and Total Conversation will be part of Europe wide next generation 112.

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Thus the text:
"*Section 410.6 of the proposed rule would require ICT with real-time voice communication features to also support communication through real-time text. Such ICT would be required to support RTT either within its own closed system or outside a network. For example, a closed communication system, such as within a federal agency, would be required to interoperate with either the publicly switched telephone network (PSTN) or Voice over Internet Protocol (VoIP) products or systems to support the transmission of real-time text. When ICT interoperates with VoIP products or systems using Session Initiation Protocol (SIP), the Board proposes to require the transmission of real-time text to conform to the Internet Engineering Task Force's RFC 4103 standard for RTP Payload for Text Conversation.*"

Is exactly what is must be. Correct.

Question 11. Is ETSI TS 122 173 or ETSI TS 134 229 sufficiently significant that the Board should consider referencing either standard when it becomes final?

Answer: I will leave that to the IMS experts.

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