

## Real-Time Text and TTY interworking in IMS/LTE and various technical environments.

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This is an overview of cases and technologies where real-time text (RTT) is considered for use in conversational services. One focus area is the IMS/LTE networks, expected to be a major environment after the PSTN/IP transition. The document focuses especially on how interworking with RTT in IMS/LTE environment and with TTY can be achieved. It is intended as a base for discussion and decisions on functionality and architecture of service deployments during the PSTN/IP transition.

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## 1 Background

The transition to an all-IP electronic communication network is going on. Support of the TTY, used by deaf, hard-of-hearing and speech-disabled persons for communication in text alternating with voice in the PSTN in USA, is regulated in USA and specified in TIA 825A[23] and ITU-T V.18 Annex A[5]. The functionality of the TTY is low, but it has some functionality that is unsurpassed among services deployed in USA. The TTY also, as many low-speed data devices, has severe problems in being transported with sufficient quality over IP. An IP based replacement with higher functionality and good transport quality needs to be established for use in IP networks. During the transition period, it seems inevitable to provide interoperability between the TTYs in PSTN and its replacement in IP.

The IP Multimedia Subsystem, specified by 3GPP is often referred to as the base for future all-IP networks. IMS is specified for both the Wireless LTE environment and for fixed and WiFi networks.

Gunnar Hellström from Omnicor has been involved in standardization, design and implementation of accessible conversational services since 1993, authored this document as a guide for specification of deployments.

The goals are to create descriptions of services that have the opportunity to satisfy user needs and likely future regulatory requirements in USA.

Some conclusions might not be compatible with current regulation or interpretation of current regulation. In such cases, a discussion with the regulating authorities is needed to see if the proposed solutions can be enabled.

A goal for TTY communication is that not more than 1% of the transmitted characters are lost or distorted during transmission. Because of the nature of the audio coding of TTY this would imply a packet loss rate of not more than 0.15%, and other conditions in the packet transmission chain not introducing other quality degradations on the TTY tones. This fact and the functional limitations of the TTY make it desirable to move to the more modern real-time text communication in IP networks.

## 2 Architecture

A very schematic architecture picture is shown here to support the discussion.

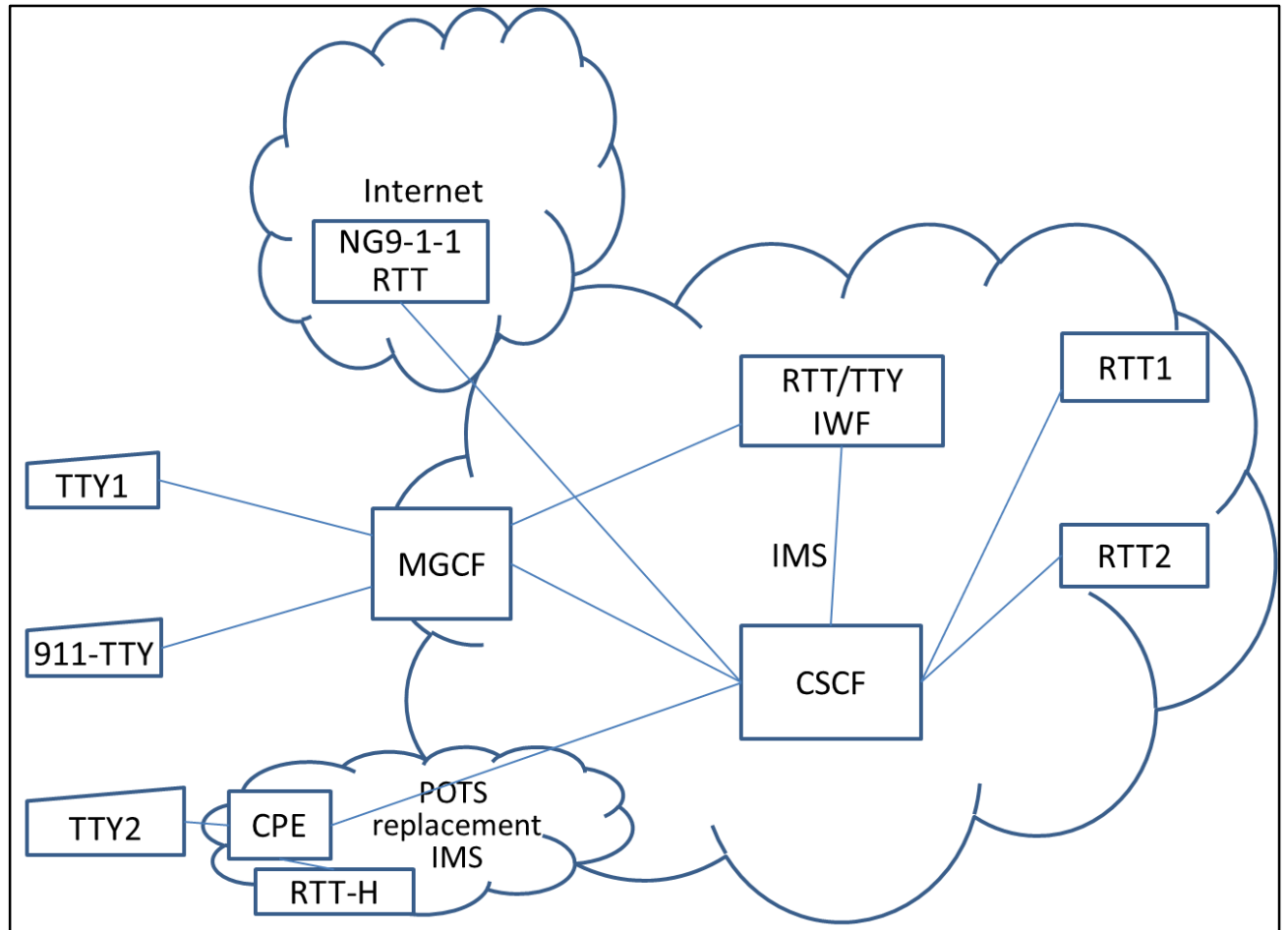


Figure 1: Schematic architecture

Functional elements in the picture:

TTY1. A POTS connected TTY.

TTY2. A TTY connected to customer premises equipment for POTS replacement

911-TTY. A TTY in a legacy connected legacy 911 PSAP.

RTT-H. A RTT terminal connected in customer premises to the IMS.

RTT1. A RTT terminal connected to the IMS/LTE network.

RTT2. Another RTT terminal connected to the IMS/LTE network

NG9-1-1 RTT. An RTT terminal in a NG9-1-1 IP connected PSAP.

CSCF. IMS Core network switching functions

MGCF. IMS/POTS Interworking function

RTT/TTY IWF. Interworking function for RTT/TTY interoperability

## 3 Functionality

### 3.1 Call functionality

The basic RTT terminal can call and be called by number in a national number plan and handle real-time text and audio simultaneously in the call.

The terminal can also use just audio, if the other party does not support RTT.

It is possible to create an extended terminal type that can handle video as well in the same calls.

### 3.2 RTT characteristics

RTT is two way simultaneous text communication. It provides smooth presentation of entered text characters and not more than one second latency from keypress to remote display. Character coding is UTF-8, suitable for international use. Less than 0.2 % of the characters are accepted to be lost and the place in presented text where loss may have occurred shall be marked with a specific character. See ITU-T F.700[2] and F.703[3].

RTT editing features includes at least new line and erase last character with remote effect as specified in ITU-T T.140[4]. The erasure has no specific limit other than the terminal display buffer. It can go into earlier "messages" previously ended by a line separator.

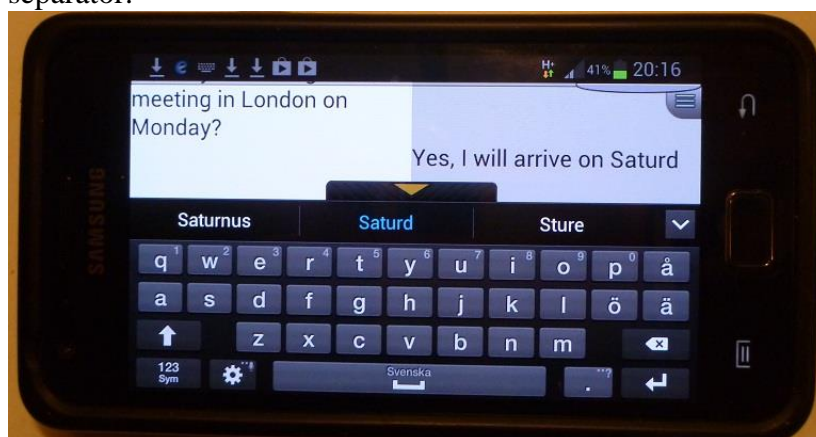


Figure 2: Example of total conversation app in RTT mode: Omnitor eCmobile.

### **3.3 Interoperability**

The RTT terminal can interoperate with TTY through an interworking unit carrying audio through when no text is transmitted, but converting between the audio carried TTY characters on the TTY side and the selected RTT text coding on the RTT side of the interworking unit while there is text to transmit.

It can be expected that voice communication quality is sufficient for carrying TTY audio-coded within the IMS core network.

It can be expected that voice communication quality may be insufficient to carry TTY audio-coded between customer located equipment for POTS replacement and the core network. Special consideration must therefore be taken to plan for that kind of connection of TTY or RTT terminals.

Note: Interworking between RTT based on 3GPP TS 26.114[11] and TTY is specified in 3GPP TS 23.226[13] and 3GPP TS 29.163 Annex I[15].

The interworking unit may also implement a feasible algorithm to manage the one-way-at-a-time nature of TTY text, and other limitations in TTY functionality. One goal should be that in most cases when the RTT user happens to type simultaneously with the TTY user, the RTT characters are stored until the TTY user gives turn. Alternatively, without this function, the RTT user needs to understand when there is a TTY in the other end and then use the old strict turn-taking etiquette for TTY communication. This issue and others related to avoiding problems inferred by the limitations of the TTY are described in chapter 9.

### **3.4 RTT terminal accessibility requirements**

The RTT terminal has settable audible and visible and tactile alerting on incoming calls. The RTT service also provides some external alerting means that can activate existing types of flashing lights, strong bells or other separate alerting devices usually used by deaf and hard-of-hearing persons. This may be a local function in the terminal, interacting wirelessly with alerting devices.

Alternatively this can be based on network signalling e.g. forked calls or BLF field notifications.

### **3.5 TTY RTT interworking function invocation**

TTY has no specific call signalling saying that a call might contain TTY. When activated in a call, the TTY interworking unit must therefore actively "listen" to the audio in the PSTN call and detect when there is TTY tones there and then perform decoding.

Assuming that it is not feasible to listen in on all audio calls for TTY detection/decoding, there should be a way to activate the TTY Interworking function

only for calls where TTY tones might appear and need to be converted, otherwise it would be incumbent to the RTT user to do so. This might be done through device software that will recognize an incoming TTY signal and invoke RTT and interworking functions, by specific routing of these calls, or by activating an interworking function in the call path.

The interworking function needs to be placed where the TTY tones can be carried with sufficient quality between the TTY in PSTN and the interworking function. That means no destructive audio coding, no misbehaving line echo cancelers in presence of even tones and less than 0.15% packet loss if the function is placed in IP network.

The main architecture problem here is where to place the interworking function and on what indications it shall be invoked.

3GPP TS 29.163 Annex I[15] specify feasible methods to invoke an interworking function in IMS controlled by the IM-MGW.

### **3.6 Interoperability cases**

Assuming that a service provider has selected a method for implementation of RTT and audio in calls, at least the following interoperability cases need to be investigated and solved.

- Other RTT terminals of the same service provider.
- TTY users in PSTN
- 711 text relay services in the PSTN.
- Legacy 911 TTY in PSAP.
- NG9-1-1, with SIP, audio and RTT coded as RFC 4103.
- RTT terminals in other service providers' networks using the same protocols.
- RTT terminals in other service providers' networks using other RTT protocols.
- 3G connected wireless terminals using the 3GPP defined CTM solution for TTY interoperability, deployed as wireless home POTS replacements and directly in wireless terminals.

### **3.7 Network connection**

The main network for RTT terminal operation is IMS/LTE.

The RTT terminal shall also work in WiFi networks connected to the Internet, by connecting to the IMS.

Wireline IMS is also used for POTS replacement. RTT shall work for such connections.

In some networks, SIP based RTT is not expected to work when the terminal is using 3G network. Then, instead legacy CTM is available for TTY interoperability.

## **4 Main assumed technology base**

3GPP Multimedia Telephony (TS 26.114[11]) and GSMA IR.92[1] are the assumed bases for RTT implementation in the IMS/LTE terminals and the RTT service.

RTT for this environment is specified in the real-time text parts of TS 26.114[11], referring to ITU-T T.140[4] including its Addendum 1 for presentation, and IETF RFC 4103[9] for transport.

Interoperability with TTY is specified in TS 23.226[13] with further detail in TS 29.163 Annex I[15].

In many 3GPP specifications, RTT and ways to provide text telephony functionality through various networks are named GTT. CS-GTT is transported by CTM modem technology, and IP-GTT is transported by RFC 4103[9].

The requirements for use in emergency calls are briefly specified in TS 22.101[18] chapter 9, and in more detail in TS 22.173. Specific considerations for use of RTT, audio and optionally video in NG9-1-1 emergency calls is specified in ETSI TS 101 470[17].

## **5 Technologies to look at as alternatives for implementation base**

### **IMS Multimedia Telephony**

The main alternative is as discussed above: IMS Multimedia Telephony with SIP and RTP based RTT with RFC 4103[9] as in TS 26.114[11]. Standards and specifications from 3GPP and IETF are in place for this implementation, and adjacent systems, such as NG9-1-1 and relay services refer to the same basic technology, and will therefore be relatively easy to arrange interoperability with.

### **WebRTC**

A recent upcoming alternative making it possible to create terminals in web pages is WebRTC with WebRTC data channel coded real-time-text.

However, this is mainly a technology for occasional terminals that may be of interest as a complement when the main structure for RTT and TTY interoperability is in place.



### **XMPP Instant Messaging with real-time text extension**

IP Multimedia Telephony audio combined with XMPP XEP-0301[25] Real-time text as enhancement of an XMPP text chat system.

There are very clear research results indicating that Instant Messaging users would appreciate a change to real-time text characteristics in these messaging services. Especially the reading party feel stress when waiting for the sending party to complete next message, and that feeling disappears by using real-time text instead.

One often used standard for Instant Messaging is XMPP. It has an extension, XEP-0301, which makes it possible to use it in real-time text mode, with other features sustained.

There are also new IETF RFC documents specifying ways to combine SIP calls with XMPP messaging. Theoretically this combination could be used to achieve the functionality required here. But there are apparent obstacles. This type of text communication requires its own server. It has its own addressing concept. The transmission requires many times more bandwidth than the RTP carried real-time text and has therefore a risk to not meet the latency requirements of real-time text. There are also no specifications for how to interoperate with NG9-1-1 and TTY. All these drawbacks lead to the conclusion that it is better to use this extension of XMPP for enhancement of messaging services than as a medium in a real-time multimedia call.

### **MSRP messaging used in real-time text mode**

IP Multimedia Telephony audio combined with the messaging standard MSRP used in time sampled mode as a Real-time text as enhancement of some 3GPP text chat system

There was once an activity in IETF for specifying this solution, but it was stopped because RFC 4103[9] was already available solving the problem, and an MSRP based solution was regarded to cause too much bandwidth and packet load in some networks. Thus no specification is available for this method.

### **Conclusion**

IMS Multimedia Telephony with SIP and RTP based RTT with RFC 4103[9] as in 3GPP TS 26.114[11] is the preferred solution, because it has the best opportunities to meet performance requirements, and its use is well standardized and specified, and well proven implementations exist in the plain SIP environment. The rest of the document builds on this conclusion.

## **6 Actions for interoperability PSTN-TTY/IMS-RTT**

This chapter describes how interoperability is achieved for different call cases between PSTN located TTY and IMS located RTT terminal.

3GPP has specified the actions to invoke TTY/RTT interworking functions in TS 29.163 Annex I[15]. The procedures require analysis and explanation of possible ways to implement them in connection with RTT terminal designs.

The invocation of the interworking function is described briefly in the following words from TS 29.163 Annex I[15]:

-----Extract from TS 29.163 Annex I.-----

It is assumed that IMS terminals supporting text media will not automatically offer text media, but that this will be instead governed by terminal configuration options and user interactions to suit the communication preferences and abilities of the user. However, an IMS terminal desiring to set up a GTT call will offer Real-Time Text media, possibly in parallel to voice media. The IMS-MGW shall then provide the conversion between Real-Time Text over RTP and text/modem signals.

On the contrary, if the mobile does not request Real-Time Text support, no Interworking function is necessary. An IMS Multimedia terminal configured to use Real-Time Text Telephony but receiving an SDP offer for voice-only media will accept this offer and then send an own subsequent SDP offer adding text media.

When receiving such a subsequent offer for text media, the IMS-MGW shall provide the conversion between Real-Time Text over RTP and text/modem signals at the CS interface. On the contrary, if the mobile does not offer Real-Time Text, no Interworking function is necessary.

The IMS user may request a text connection from the beginning of a call, or add request for Real-Time Text media at a later stage in a call that was originally established with audio only.

-----End of excerpt-----

This means that invocation of the interworking function is depending on the IMS terminal user activating the real-time text function.

At first glance this seems risky. There might be many scenarios when a TTY call would not be passing any interworking function, and therefore situations without text communication between text users would be the result.

However, with knowledge from how TTY and RTT users handle calls, it can be seen that most scenarios are well handled. The scenarios which are not handled well can be amended by extra functionality in the terminal.

This is an analysis of how the approach from TS 29.163 Annex I[15] will perform in various scenarios.

### **6.1 TTY user in PSTN calling - RTT user depending on text answers with RTT enabled.**

- A TTY user in PSTN calls with a TTY.
- An RTT user answers and requests RTT from beginning of call.
- The interworking function will be invoked and ready from the beginning.
- The RTT user types a greeting.
- The interworking function translates to TTY tones.
- The greeting is received and displayed on the TTY.
- The TTY caller responds.
- The call continues in text, optionally alternating with voice during the call.

Conclusion: successful

### **6.2 TTY user in PSTN calling - Hearing but experienced RTT user answers.**

- TTY user calls.
- Hearing user (experienced RTT and TTY user) answers by voice. Just silence is heard from the calling party.
- The hearing user has friends who are TTY users, so the hearing user assumes that this can be a text call and presses "start text" button.
- RTT is added to the call and interworking function invoked.
- Hearing user types "Hi ...GA". and waits.
- TTY user gets the text and types back.
- The call can continue in text mode alternating with voice.

Conclusion: Successful.

Optionally, when the TTY user sees the line signal on the TTY flicker by the voice user's voice answer, the TTY user understands that a hearing person answered, and taps the space bar to produce a short tone sequence. The hearing person hears that tone sequence and is then sure that it is a TTY calling, and follows the scenario above a bit more assured. This is a habit already established in the PSTN, and it can continue to work in IMS as long as there are users who recognize the sound of the TTY.

### **6.3 TTY user in PSTN calling - hearing IMS user answering, not knowing about TTY**

- TTY user calls.
- Hearing user (not experienced with TTY) answers by voice. Just silence is heard from the calling party.
- The hearing user does not know what to do with the call and is near to hang up.
- When the TTY user sees the line signal on the TTY flicker by the voice user's voice answer, the TTY user understands that a hearing person answered, and taps the space bar to produce a short tone sequence.
- The hearing person hears that tone sequence, but does not know that it means TTY call, and therefore does not activate RTT mode.
- No text contact is achieved and the users hang up

Conclusion: Unsuccessful. But the scenario with a TTY user making unsolicited TTY call to hearing people without TTY experience will be very rare.

Possible amendment: The phone app can be extended with a function that detects the typical TTY tone pattern with 1400 Hz and 1800 Hz tones with lengths in multiples of 22 milliseconds. When such pattern is detected, a best effort decoding and presentation of the text is made, and a visible and maybe audible and tactile indication can be provided saying "Possible TTY call, activate RTT function and respond by text." Even if the quality of the TTY tones are not good enough for reliable decoding, they should be good enough to be deemed to be likely TTY tones, and some characters might be possible to decode and display.

- Then the hearing user assumes that this can be a text call and presses "start text" button.
- RTT is added to the call and interworking function invoked.
- Hearing user types "Hi" and waits.
- TTY user gets the text and types back.

The call can continue in text mode alternating with voice, with the TTY user needing to inform the TTY user about turn-taking etiquette etc., if the system has no specific protection against typing against turn.

There are risks for false detection of TTY in presence of music on hold etc. It is therefore important that the message is not destructive on the call, and that it is vaguely expressed.

Conclusion after amendment: Successful, but the amendment will be very rarely needed.

#### **6.4 Hearing PSTN phone user calling, knowing about TTY and having a TTY close - calling RTT user depending on text answers with RTT enabled.**

- A hearing user makes a voice call. The user knows about TTY and has a TTY close.
- An RTT user answers and requests RTT from beginning of call.
- The interworking function will be invoked and ready from the beginning.
- The RTT user types a greeting and the text will be heard as TTY tones at the POTS end.
- The POTS caller understands that the TTY needs to be connected and types something like "Hi, Gary here, sorry I had not my TTY connected so I missed your answer. GA"
- The RTT user gets the greeting and repeats the initial answer and the call continues by text.

Conclusion: Successful. Here we got a bit of delay in the beginning, and a lost greeting phrase, but that is exactly what happens in POTS calls in a corresponding scenario, so users are used to it and it will be accepted.

#### **6.5 Hearing PSTN phone user, knowing about TTY and having a TTY close - calling hearing but experienced RTT user.**

- A hearing PSTN user makes a voice call. The user knows about TTY and has a TTY close.
- A hearing user answers by IMS voice. The answering part is used to RTT but does not activate it.
- They start the call in voice mode.
- During the call, they talk about an address in another country that has complicated spelling. The IMS user detects that the PSTN user does not get the name right. He says: "Connect your TTY".
- The IMS user presses "Start RTT" on the phone, and types the name in the RTT user interface.
- The PSTN user types back "Got it, thanks, over to voice GA"

- The call continues in voice with RTT and TTY ready.

Conclusion: Successful enhancement of a voice call.

### **6.6 RTT user depending on text calls with RTT enabled - calls to TTY user.**

- RTT user activates RTT and calls a TTY user.
- RTT is enabled from the beginning and one leg of the call is in PSTN, thus the interworking function is invoked.
- The TTY user answers and types a greeting.
- The greeting is converted to RTT and displayed to the RTT user.
- The call continues in real-time text mode and optionally alternating with voice.

Conclusion: successful.

### **6.7 Hearing but experienced RTT user calls without RTT enabled - TTY user answers**

- Hearing IMS user with RTT experience calls a PSTN user.
- The PSTN user needs text and answers with TTY and types a greeting.
- The hearing IMS user hears the TTY tones, and knows that that means TTY.
- The IMS user presses "Start RTT"
- The interworking function is invoked.
- The IMS user types e.g. "Hi it is Gary, sorry I missed your answer, GA".
- The call continues in real-time text mode and optionally alternating with voice.

Conclusion: successful after a small delay that would have been the same in PSTN and is handled well by TTY users.

### **6.8 Hearing but experienced RTT user calls without RTT enabled - hard-of-hearing user answers first with voice but really need TTY to communicate well**

- Hearing IMS user with RTT experience calls a PSTN user.
- The PSTN user is hard-of-hearing but hopes to manage with voice and hearing and answers with voice.

- They start talking.
- Soon the PSTN user experience that it is hard to hear the other part well, so the PSTN user says "Sorry, I cannot hear you well, can you type instead please? I activate my TTY"
- The IMS user activates RTT by pressing "Start RTT"
- The interworking function is invoked.
- The IMS user types the part of the conversation towards the PSTN user who reads it on the TTY.
- They alternate between text and voice, so that the PSTN user lifts the receiver and talks, and then returns to TTY mode for the responses.

Conclusion: Successful

### ***6.9 Hearing IMS user, not knowing about TTY calls - a TTY user happens to answer.***

- A hearing IMS user with no experience of TTY calls a PSTN number for a voice call.
- Unexpectedly only a TTY user was near the PSTN phone and answers and types a greeting.
- The IMS user hears the beeps, but does not know that it means TTY text.
- There is a risk that the IMS user hangs up believing that it was a fax number by mistake.

Conclusion: Unsuccessful, but this failing scenario is common and the failure should be accepted. When TTY users bother about taking calls in phones they share with hearing persons, they are used to the situation that they try to answer by TTY, but the calling party has no TTY and hangs up.

Possible but not required amendment: The phone app can be extended with a function that detects the typical TTY tone pattern with 1400 Hz and 1800 Hz tones with lengths in multiples of 22 milliseconds. When such pattern is detected, a best effort decoding and presentation of the text is made, and a visible and maybe audible and tactile indication can be provided saying "Possible TTY call, activate RTT function and respond by text." Even if the quality of the TTY tones are not good enough for reliable decoding, they should be good enough to be deemed to be likely TTY tones, and some characters might be possible to decode and display.

Conclusion after amendment: Successful, but rarely occurring and not expected that it should work.

### **6.10 RTT user depending on text calls with RTT enabled - to other IMS user not very experienced in RTT.**

- RTT user depending on text calls with RTT enabled.
- The call is answered by an IMS user. Since RTT is enabled by the caller, RTT will be enabled in the answering IMS terminal.
- The answering party is not experienced with RTT though and answers with a voice greeting.
- The calling party does not hear the answer but sees an indication of audio on the line.
- Therefore the calling party types a greeting that gets through to the called party.
- The called party sees the RTT text and understands that text must be used in this call.
- The call proceeds mainly in real-time text mode.

Conclusion: Successful after a bit of initial confusion.

This is an improvement over the situation with TTYs in the PSTN, where it is not expected that any other than those with reasons to have TTY calls have TTYs. In IMS it is possible to have RTT mode available in all or a large part of the terminals.

### **6.11 IMS user not very experienced in RTT calls by voice - RTT user depending on text answers.**

- A hearing IMS user with no experience of RTT calls an IMS call with voice enabled.
- Unexpectedly an RTT depending user answers and activates text
- The answering terminal will first come up in voice mode, but will immediately request addition of RTT.
- The called user types a greeting.
- The voice user detects that text is coming up in the RTT user interface and understands that the call must be held in text.
- The call continues in text.

Conclusion: Successful

### **6.12 IMS voice user calls - IMS voice user and during the call they get a reason to transfer text in the call.**

- A hearing IMS user makes a voice call. The user knows about RTT.
- A hearing user answers by IMS voice. The answering party is used to RTT but does not activate it.



- They start the call in voice mode.
- During the call, they talk about an address in another country that has complicated spelling. One IMS user detects that the other user does not get the name right. He says: "I will type it".
- The user who wants to type presses "Start RTT" on the phone.
- The terminals add real-time text to the call.
- The user who wanted to type types the name in the RTT user interface.
- The other user says "Got it, thanks."
- The call continues in voice with RTT ready.

Conclusion: Successful

### ***6.13 RTT user depending on text calls with RTT enabled - calls to legacy 9-1-1 for an emergency.***

- RTT user activates RTT and calls 9-1-1.
- RTT is enabled from the beginning and one leg of the call is in PSTN, thus the interworking function is invoked.
- A 9-1-1 PSAP operator answers with a voice greeting.
- No response is heard from the RTT caller.
- The RTT user gets a flickering indication of sound on the line from the 9-1-1 greeting phrase in voice.
- The RTT user types something in hope that the 9-1-1 operator shall understand that it is a text call. The text is translated to TTY tones by the interworking function and sent to the 9-1-1 PSAP.
- The 9-1-1 operator has instructions to try sending in TTY mode if no response is heard, so either that happens or the 9-1-1 operator hears the TTY tones.
- The 9-1-1 operator types a greeting on the TTY function built into their equipment.
- The greeting is converted to RTT and displayed to the RTT user.
- The call continues in real-time text mode and optionally alternating with voice.

Conclusion: successful.

Note: A callback from the legacy 9-1-1 PSAP would be done with TTY. The RTT user would activate RTT when answering and the call will work successfully in text mode through the interworking function.

### **6.14 RTT user depending on text calls with RTT enabled - calls to NG9-1-1 for an emergency.**

- RTT user activates RTT and calls 9-1-1.
- RTT is enabled from the beginning.
- Assuming that IMS has SIP interworking with NG9-1-1 according to NENA NG 9-1-1 08-003[21] specification, the call goes to an NG9-1-1 PSAP and RTT and audio is negotiated.
- A PSAP operator answers with a voice greeting and sees that a text window has opened indicating text has been activated in the call, so the operator also types a text greeting.
- The text greeting is seen by the RTT user in emergency.
- The call continues in real-time text mode and optionally complemented with voice or sound from the place of emergency.

Conclusion: successful.

Note: A callback from NG9-1-1 would have the same kind of text channel enabled when the call is placed, and therefore it would also be successful.

### **6.15 Conclusion**

The method for interoperability between TTY and RTT specified in TS 29.163 Annex I[15] is sufficient and good.

This analysis shows successful result for 12 of the 14 scenarios.

The two failing scenarios are failing also in similar situations in PSTN and could be left unsolved.

The failing scenarios are:

6.3. TTY user in PSTN calling - hearing IMS user answering, not knowing about TTY.

6.9. Hearing IMS user, not knowing about TTY calls - a TTY user happens to answer.

Possible amendments are mentioned and could be done in the IMS terminal. After the amendments the failing scenarios will also be successful, and therefore the result better than in PSTN.

## **7 Interoperability between 3G CS domain and IMS-RTT**

For the 3G Circuit Switched domain, there is a specification for TTY interoperability called GTT-CS, in TS 22.226[12], TS 23.226[13] and TS 26.226[14]. The core of the solution is the robust modem called CTM making it possible to send text in the audio channel alternating with voice.

Interoperability between GTT-CS and IMS-RTT can be done as briefly indicated but said to be out of scope of TS 29.292[16], section 5.4.4. Depending on how the architecture for connection between the

The implementations of GTT-CS has become widely used in wireless 3G POTS replacement equipment, enabling connection of TTY among other traditional PSTN telephony equipment.

It needs to be judged if a 3G based GTT-CS solution is needed also in wireless LTE/IMS terminals for cases when these terminals fallback to 3G in areas with no LTE coverage.

An alternative in some networks can be to use plain SIP based RTT when the terminal is outside LTE coverage, but that causes problems with two kinds of use of SIP accounts.

## **8 Actions for interoperability between POTS replacement equipment and IMS-RTT.**

This chapter discusses how interoperability can be achieved for different call cases between POTS replacement equipment and IMS located RTT terminal.

Three main cases need to be discussed.

### ***8.1 TTY connected to IMS wireline customer premises equipment for POTS replacement.***

An alternative to be investigated is when the POTS replacement equipment contains RJ-11 connector for connection of traditional POTS equipment, and the POTS replacement equipment converts to IMS.

Connection of TTY to such a connector without extra considerations for protecting or converting the TTY tones seems to have an opportunity to meet the requirements for good conditions for successful carrying TTY tones to the core IMS network.

The Customer Premises Equipment (CPE) will in this case work as a VoIP gateway, and the connection and functionality of TTY calls in this environment will be similar to the case with the wireline PSTN connected TTY.

The core network located TTY/RTT interworking function in TS 29.163 Annex I[15] is specified for working in the ISUP/SIP interface to the CS domain. That is different from this case, when the POTS replacement CPE converts the call to IMS/SIP with TTY in the audio stream. The interworking function is needed also for this case for calls between TTY and RTT terminals.

Additional specification work seems needed for a variant of TS 29.163 Annex I[15] for interworking in a SIP-TTY/SIP-RTT interface. It must then be considered if the RJ-11 port of the POTS replacement CPE needs to be assigned a specific device profile that allows the invocation of the interworking function for TTY-s connected to that port.

An alternative to invoking the central core network interworking function is to use the same customer premises located interworking function as described below for wireless POTS replacements.

In order to achieve the required quality for the TTY transmission, the following must apply:

- Audio is G.711 coded or a codec with equally good characteristics for TTY transmission is used.
- The line echo canceller in the CPE must be proven to behave well with TTY tones transmitted on the connection and in realistic network conditions. (= pass test 14 in ITU-T G.168)

Tests are needed to verify that acceptable TTY transmission results are achieved. If that is not the case, the solution selected for wireless POTS replacement need to be applied also for the unsuccessful wireline cases.

## ***8.2 TTY connected to IMS wireless customer premises equipment for POTS replacement.***

An alternative to be investigated is when wireless POTS replacement equipment contains RJ-11 connector for connection of traditional POTS equipment, and the POTS replacement equipment converts to IMS.

Connection of TTY to such a connector without extra considerations for protecting or converting the TTY tones has an apparent risk of introducing more than allowed destructive communication errors, resulting in character loss or corruption in the TTY communication.

The dominating sources of problems for TTY transmission in the audio path are packet loss, jitter and misbehaving line echo cancellers in VoIP gateways. See FCC EAAC TTY transition report[19], chapter 6.

Alternatives available to handle this situation are:

1. Introduce communication protection mechanisms to improve the robustness of audio packet transmission.
2. Convert between TTY and RTT in the Customer Premises Equipment.
3. Convert between TTY and RTT in a separate adapter and connect it to an IP interface of the CPE.

### *8.2.1 Communication protection mechanisms for packet transmission of TTY tones*

There are a few technologies available for protection against the destructive effect of packet loss on TTY transmission.

They are described in FCC EAAC TTY Transition report[19], chapter 8.

- TIA 1001[24]. A standard for transmission of TTY over an IP segment. Specifies detection on all calls and converting TTY characters to RFC 4351[10] text coded packets in the audio stream.
- ITU-T V.151[7]. A standard for transmission of text telephony including TTY over an IP segment. Specifies detection of tones during a call and converting TTY characters to RFC 4351[10] text coded packets in the audio stream, and converting back to TTY and PSTN at the other end.

Both these methods are mainly intended for cases when the text is converted back to PSTN based TTY at the other end. That case does not exist in the situation discussed here, so the application of these standards would require extra development.

Cisco has had implementation of V.151[7] in some VoIP gateways. V.150.1[6] is a related technology for modem over IP supported by Cisco. If there is any interest to transport TTY reliably over IP connections, it might be of interest to talk to Cisco about these solutions.

### *8.2.2 Convert between TTY and RTT in the Customer Premises Equipment.*

A function for RTT negotiation and TTY/RTT interworking function in a similar way as the TS 29.163 Annex I[15] specifies can be included in a CPE design. The interworking function for this case must be included in the CPE because the TTY tones will not survive transport to any central interworking function.

For this case, the CPE must have sufficient processing power to detect TTY in audio streams and invoke interworking function on such detection. This requirement is easy to fulfill for modern CPE designs. The result covers the communication scenarios slightly better than the core network located solution from TS 29.163 Annex I[15].

It is essential that the CPE performs the negotiation for addition of text in the call on detection of TTY in the audio stream. Otherwise no party would be able to request the core network located TTY/RTT interworking function in calls between PSTN located TTY and POTS replacement connected TTY.

It is recommended that the CPE can store a few characters of text to be transmitted while waiting for the request to open the text channel to be confirmed. Received tones

that can be detected to be from a TTY (even if slightly distorted and with gaps from lost packets) must cause the negotiation of a text channel. Such audio should then be carried through to the connected TTY. There is a risk for loss or corruption of initial received characters from called PSTN located TTYs before the core network located interworking function gets activated. Answering TTY users are used to cases when their answer is missed because the caller did not expect that a TTY would answer. But calling TTY users are used to getting the received answer text clear from the beginning. This risk for loss or corruption of a few initial characters from an answering PSTN located TTY may therefore cause slight user annoyance from calling users of TTYs connected to POTS replacements of this type .

### *8.2.3 Convert between TTY and RTT in a separate adapter and connect it to an IP interface of the CPE.*

A function for RTT negotiation and TTY/RTT interworking function in a similar way as the TS 29.163 Annex I[15] specifies can be included in a separate adapter placed between the TTY and the CPE. The connection between the adapter and the CPE would be IP based and can e.g. be Ethernet cabled or by WiFi.

The adapter must also have sufficient processing power to detect TTY in the audio streams in any direction and must invoke interworking function on such detection, covering the communication scenarios slightly better than the core network located solution from TS 29.163 Annex I[15].

It is essential that the adapter performs the negotiation for addition of text in the call. Otherwise no party would be able to request the core network located TTY/RTT interworking function in calls between PSTN located TTY and POTS replacement connected TTY.

The negotiation for text in the call can be done initially in all calls, or on detection of TTY tones in a call. TTYs are sometimes combined with regular POTS telephones on the same line, and the calling made from the POTS telephone. In that case, many calls may be done through the adapter without any use of the TTY. For such cases, it is a waste of resources to signal text use from the beginning of the call. Instead the early invocation prevents loss of initial TTY characters in the call.

If instead text inclusion is negotiated when it appears in a call, it is recommended that the adapter can store a few characters of text to be transmitted while waiting for the request to open the text channel to be confirmed. Received tones that can be detected to be from a TTY (even if slightly distorted and with gaps from lost packets) must cause the negotiation of a text channel. Such audio should then be carried through to the connected TTY. There is a risk for loss or corruption of initial received characters from called PSTN located TTYs before the core network located interworking function gets activated. Answering TTY users are used to cases when their answer is missed because the caller did not expect that a TTY would answer. But calling TTY users are used to getting the received answer text clear from the beginning. This risk for loss or corruption of a few initial characters from an answering PSTN located

TTY may therefore cause slight user annoyance from calling users of TTYs connected to POTS replacements of this type.

Omnitor has developed an adapter solution in the RERC TA project work together with Trace Center. It was running in a small form factor box with RJ-11 and RJ-45 connectors. It used plain SIP instead of the IMS encapsulated SIP required in the IMS solution. This was in 2008. Both available hardware and the base Asterisk system have changed since 2008, but it is certainly doable to refresh this implementation and make it work in new hardware and IMS.

### ***8.3 RTT terminal located in customer premises being used as IMS connected TTY replacement***

An alternative, for cases when TTY-like functionality is desired at a location where wireless IMS based POTS replacement is used, is to enable direct IMS connection of an RTT terminal provided as a TTY replacement. (RTT-H in the architecture picture)

Because of the better functionality of RTT compared to TTY, this approach should be available anyway for users who accept to change equipment and for new users in need of text, audio (and optionally video) communication.

The device should have a convenient keyboard, and a screen that allows convenient operation and reading the conversation. A 10-11" WiFi communicating tablet format with attached keyboard would likely be most popular. Some users would likely feel most comfortable with a simplistic design, where the only task of the device is to be a conversational terminal, while others would appreciate to have the communication function available among many other functions.

A few users will have problems to move from TTY to this type of device because of the need for re-learning how to handle a call.

The communication technology for this type of device would most naturally be the same as for other IMS connected RTT terminals, based on GSMA IR.92[1] including its Annex B.2, and the considerations for accessible alerting and TTY interoperability fulfilled.

## **9 Presentation and dialogue control in RTT-TTY interworking**

TTY communication has a number of serious shortcomings compared to modern real-time text communication.

Special care must be taken so that these limitations cause minimal confusion and quality degradations in contact with RTT users.

This topic is discussed and more elaborated in the FCC EAAC report "EAAC Report on Procedures for calls between TTY Users and NG911 PSAPs"[20].

Very briefly, the shortcomings of the TTY are:

- **One transmission direction at a time.**

While the TTY is transmitting, it cannot receive. If someone tries to transmit at the same time, the characters are garbled or lost. Therefore users are expected to take turns typing, and give turn by specific text tokens.

- **Use of audio only when not involved in text transmission.**

While the TTY handles text, voice cannot be transmitted. Alternating between audio and text can be done but must be formally signalled between the users.

- **Slowness.**

TTY can only transmit between 4 and 7 characters per second. That is slower than many typing persons can type, easily causing backlog in communication.

- **Only upper case or lower case characters - not both.**

The character set of the TTY is limited. One limitation is that it has only one case. The presentation as upper or lower case is usually a setting in the device.

- **Limited character set.**

Another character set limitation is that many special characters are missing from the TTY character set. The following characters cannot be presented:

@ # % & \ \* \_ < >.

- **Limited display area.**

Many TTYs have only one or two lines for display of text. That makes it inconvenient to use new lines in text to a TTY.

- **Risk for corruption or loss of characters.**

TTY transmission is very sensitive to packet loss and audio distortion from coding and choking or squelching echo cancellers.

- **Risk for corruption of long series of characters.**

TTY transmission has two case shifts: letter shift and figure shift. They are selected by a shift character. If a shift character is lost, then there may be a long series (up to 72) of characters presented in the wrong case shift and therefore unreadable.

The more RTT is used for RTT to RTT communication, the more they will forget or not be informed about the restrictions and conventions used in TTY communication. Then, if a RTT user has a call interworking with a TTY, it is an apparent risk that the RTT user will not obey the TTY turn-taking conventions, and send against turn potentially resulting in character loss or corruption of text.



The limitations in the TTY character set and speed can also cause confusion and risk in calls with ignorant RTT users.

Certain functions may be introduced to reduce the risks. Note that some of these functions act on user behaviour and that is not totally predictable. Some also modify text towards the TTY user without showing that to the RTT user. So precaution must be taken to not introduce more confusion or deadlocks by introducing these functions. Selection of which functions to introduce in real services should be preceded by usability evaluation.

#### **Functions that may selectively be introduced in the TTY/RTT interworking function:**

- **Include an agreed indication in the signalling path** to the RTT device indicating that TTY interworking is active in the call.
- **Store characters received from the RTT side** while reception from the TTY side is going on. Send them when TTY reception has finished.
- **Store characters received from the RTT side** while reception from the TTY side is going on. Send them when a turn-giving token has been received from the TTY ( with precautions against deadlock by timeout )
- **Send text information to the RTT user reminding on TTY conventions.**  
Short text information can be sent in the beginning of the call.
- **Add turn-giving token in text towards the TTY.**  
In some situations, logic in the interworking function can deduce that the RTT user likely forgot to end by giving turn. A turn-giving token ("GA") can then be inserted to reduce confusion.
- **Conversion of non-supported characters towards the TTY**  
Characters sent from the RTT device not supported by the TTY can be converted to supported characters reminding of the original ones.

#### **Functions that may selectively be introduced in the RTT communication user application:**

- **Detect signalling path indicator that TTY interworking is active**  
and then change the user interface to reduce risks by e.g.
  - Clearly indicating in the text input area that it is in "TTY mode"
  - Ignoring character input while reception is going on, or storing typed characters for later transmission.

- Ignoring unsupported characters typed by the RTT user and clearly showing to the user that that happened
- Converting unsupported characters typed by the RTT user to similar supported characters and clearly showing to the user that that happened.
- Not accepting large chunks of pasted text.
- Adding turn-giving token "GA" when the user indicated end-of-message.
- **Provide a shift change function on presentation of received text.**  
TTY products usually have a function for the user to change the interpretation of already received text to the opposite case. This is to compensate for possible loss of a case shift character causing a series of seemingly garbled characters. The same function could be provided in the RTT user interface.

## 10 Multi-party call aspects

Multi-party call support for RTT can be done in a number of ways.

- A multi-party bridge can mix multiple sources of text to one stream in a text chat style presentation in conference-unaware terminals. A specification of a multi-party real-time text mixer is available at [Realtimetext.org](http://Realtimetext.org)[26].
- Each source can set up its own session with a multi-session enabled terminal.
- A multi-party bridge can merge text from different sources and mark text with its source by the CSRC contributing source label concept of RTP, so that a conference-aware terminal can display text in a way that is locally decided in the terminal.

Because of the speed and simultaneity limitations in TTY, it is not feasible to participate in multi-party calls with TTY.

## 11 Network considerations

All network components involved in RTT calls must allow and be able to handle the signalling about the real-time text medium.

Keep-alive traffic may need to be sent along the real-time text media path at intervals depending on the involved network components.

If network address translation (NAT) is used in the network, the usual methods for traversing NAT must be applied on the real-time text stream as well as other media.

## **12 Security**

The RTT transport mainly discussed in this document is carried by RTP as specified in RFC 4103[9]. RFC 4103 specifies that it is possible to secure RTT by using SRTP (RFC 3711[8]).

It is recommended that SRTP use is enabled with a suitable key management standard and that it is made clearly visible in the user interface when RTT is secured.

The 3GPP specifications have not yet mentioned the use of SRTP for RTT.

## **13 Recommended actions**

In the chapters above, some items are described that lack previously written specifications. They are listed here as a summary of points needing resolution.

### ***13.1 RTT terminal functionality***

The scenarios in chapter 6 show that the following functionality is important for successful calls.

- An indicator for received audio, so that a user depending on text can see that a voice user has answered.
- Optionally a best effort decoding and presentation of received audio coded TTY text before RTT text has been negotiated, or an indication that TTY text is likely received.
- Include a setting in the terminal for requesting RTT initially in calls or not.
- Include in the user interface an easily operated way to request RTT at the moment initially when making or answering a call or during a call.
- With the setting for requesting RTT initially in calls, or when the user requests RTT when answering a call, let the terminal make a reINVITE with RTT if the session was initially offered and therefore answered without RTT.

### ***13.2 Core network TTY/RTT interworking function invocation for POTS replacement connected TTY***

As described in section 9.1, a method for invocation of core network located TTY/RTT interworking function needs to be specified for communication with TTYs attached to wireline IMS POTS replacement. The specification would be similar to the specification in TS 29.163 Annex I[15] of invocation of the TTY/RTT interworking function in the IMS/CS interface.

### ***13.3 Wireless IMS POTS replacement***

The placement of the TTY/RTT interworking function for wireless IMS POTS equipment needs to be decided and specified. One alternative is in the CPE, another is in a specific adapter between the CPE and the TTY. See chapter 8.2.

### **13.4 Presentation and dialogue control in RTT-TTY interworking**

The mechanisms described in chapter 9 for avoiding problems by the limitations in functionality of the TTY need to be reviewed, and the ones selected to be included in implementations further specified.

## **14 Abbreviations and definitions**

3GPP	3 <sup>rd</sup> Generation Partnership Program
BLF	Busy Lamp Field
CPE	Customer Premises Equipment
CS	Circuit Switched
CTM	Cellular Text Modem
ETSI	European Telecommunication Standards Institute
GSMA	GSM Association
GTT	Global Text Telephony
ITU	International Telecommunications Union
IETF	Internet Engineering Task Force
IMS	IP Multimedia Subsystem
IWF	Interworking Function
IP	Internet Protocols
LTE	Long Term Evolution (Mobile network for IMS)
MSRP	Message Session Relay Protocol
NENA	National Emergency Number Association
NG9-1-1	Next generation (IP based) emergency services in USA
PSTN	Public switched telephone network
POTS	Plain old telephone system
PSAP	Public Safety Answering Point
RJ-11	POTS connector standard
RTP	Real Time Protocol
RTT	Real-time text
	Real-time text Text transmitted instantly while it is being typed or created, to allow recipient(s) to immediately read the sender's text as it is written, without waiting.
SIP	Session Initiation Protocol ( base for VoIP )
TTY	Text telephone type used in USA
UTF-8	Uniform Transformation Format -8. (Unicode coding)
XMPP	Extensible Messaging and Presence Protocol

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