



ATIS-0700030

ATIS Standard on -

Real Time Text End-to-End Service Description Specification



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Real Time Text End-to-End Service Description Specification

Alliance for Telecommunications Industry Solutions

Approved July 12, 2018

Abstract

This Standard defines the RTT end-to-end service behavior for the handling of Real Time Text (RTT) in support of the IP transition in order to facilitate a consistent use of RTT across multiple Commercial Mobile Service Providers (CMSPs).

Foreword

The Alliance for Telecommunication Industry Solutions (ATIS) serves the public through improved understanding between carriers, customers, and manufacturers. The Wireless Technologies and Systems Committee (WTSC) develops and recommends standards and technical reports related to wireless and/or mobile services and systems, including service descriptions and wireless technologies. WTSC develops and recommends positions on related subjects under consideration in other North American, regional, and international standards bodies.

The mandatory requirements are designated by the word shall and recommendations by the word should. Where both a mandatory requirement and a recommendation are specified for the same criterion, the recommendation represents a goal currently identifiable as having distinct compatibility or performance advantages. The word may denotes an optional capability that could augment the standard. The standard is fully functional without the incorporation of this optional capability.

Suggestions for improvement of this document are welcome. They should be sent to the Alliance for Telecommunications Industry Solutions, WTSC 1200 G Street NW, Suite 500, Washington, DC 20005.

At the time of consensus on this document, WTSC, which was responsible for its development, had the following leadership:

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- M. Younge, WTSC Vice Chair (T-Mobile)
- P. Musgrove, WTSC SN Chair (AT&T)
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1 Scope, Purpose, & Application

1.1 Scope

The scope of this Standard is limited to Real Time Text (RTT) service, implemented as specified in IETF RFC 4103 [Ref 4] and as described in ATIS-1000068 [Ref 2], for both emergency and non-emergency communications via Commercial Mobile Service Provider (CMSP) networks.

IP Multimedia Subsystem (IMS) service continuity of RTT (3GPP TS 23.237 [Ref 17]) is outside the scope of this Standard.

1.2 Purpose

The purpose of this Standard is to define the RTT end-to-end service behavior for RTT communication in an all Internet Protocol (IP) environment to facilitate a consistent use of RTT across multiple CMSPs.

1.3 Application

This Standard is applicable to CMSPs, CMSP network infrastructure vendors, mobile device manufacturers, emergency services vendors, and Public Safety Answering Point (PSAP) vendors.

2 Normative References

The following standards contain provisions which, through reference in this text, constitute provisions of this Standard. At the time of publication, the editions indicated were valid. All standards are subject to revision, and parties to agreements based on this Standard are encouraged to investigate the possibility of applying the most recent editions of the standards indicated below.

[Ref 1] FCC Doc-319386A, *Federal Communications Commission (FCC) Emergency Access Advisory Committee (EAAC) Report on TTY Transition*; March 2013.¹

[Ref 2] ATIS-1000068, *Technical Report on Support of TTY Service over IP using Global Text Telephony*.²

[Ref 3] TIA-825-A, *A Frequency Shift Keyed Modem for Use on the Public Switched Telephone Network*.³

[Ref 4] IETF RFC 4103, *RTP Payload for Text Conversion*.⁴

[Ref 5] FCC Doc-321705, *Federal Communications Commission (FCC) Emergency Access Advisory Committee (EAAC) Report on Procedures for Calls between TTY Users and NG9-1-1 PSAPs*; June 14, 2013.¹

[Ref 6] FCC Doc-321704, *Federal Communications Commission (FCC) Emergency Access Advisory Committee (EAAC) Report on Proposed Procedures for TTY as Text Terminal in Legacy PSAPs*; June 14, 2013.¹

¹ This document is available from the Federal Communications Commission. < <http://www.fcc.gov/> >

² This document is available from the Alliance for Telecommunications Industry Solutions (ATIS). < <http://www.atis.org> >

³ This document is available from the Telecommunications Industry Association. < <http://www.tiaonline.org> >

⁴ This document is available from the Internet Engineering Task Force (IETF). < <http://www.ietf.org> >

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[Ref 7] 3GPP TS 22.173, *IP Multimedia Core Network Subsystem (IMS) Multimedia Telephony Service and supplementary services; Stage 1*.⁵

[Ref 8] TIA/EIA J-STD-034, *Wireless Enhanced Emergency Services*.²

[Ref 9] J-STD-036-C, *Enhanced Wireless 9-1-1 Phase II*, June 2011 including the addendum in J-STD-036-C-1, *Addendum to J-STD-036-C, Enhanced Wireless 9-1-1 Phase II*.²

[Ref 10] NENA 05-001, *NENA Standard for the Implementation of the Wireless Emergency Service Protocol E2 Interface*.⁶

[Ref 11] NENA 08-502, *NENA Generic E9-1-1 Requirements Technical Information Document (TID)*.⁶

[Ref 12] Americans with Disabilities Act (ADA).⁷

[Ref 13] NENA-STA-010.2, *NENA Detailed Functional and Interface Standards for the NENA i3 Solution*.⁶

[Ref 14] IETF RFC 5369, *Framework for Transcoding with the Session Initiation Protocol (SIP)*.⁴

[Ref 15] ATIS-0700015.v003, *ATIS Standard for Implementation of 3GPP Common IMS Emergency Procedures for IMS Origination and ESInet/Legacy Selective Router Termination, Version 3*.²

[Ref 16] 3GPP TS 29.163, *Interworking between the IP Multimedia (IM) Core Network (CN) subsystem and Circuit Switched (CS) networks*.⁵

[Ref 17] 3GPP TS 23.237, *IP Multimedia Subsystem (IMS) Service Continuity; Stage 2*.⁵

[Ref 18] ATIS-0700029, *Real Time Text Mobile Device Behavior*.²

[Ref 19] FCC 16-169, *FCC Report and Order and Further Notice of Proposed Rulemaking in the Matter of Transition from TTY to Real-Time Text Technology*, December 15, 2016.¹

3 Definitions, Acronyms, & Abbreviations

For a list of common communications terms and definitions, please visit the *ATIS Telecom Glossary*, which is located at < <http://www.atis.org/glossary> >.

3.1 Definitions

Real-time text (RTT): Characters entered via keyboard, drawing, speech recognition, or other text creation method on terminal A and rendered in real time on the display of terminal B. The flow is time-sampled so that no specific action is needed from the user to request transmission resulting in text communication that is perceived by the user as being a real-time view of the text entry.

NOTE: In this standard, RTT is restricted to IETF RFC 4103-based implementation [Ref 4].

RTT call: Call with RTT and voice media.

RTT media: Real-time text.

RTT service: Conversational service providing communication with RTT and voice media.

TTY: The term for the text telephones that are based on Baudot and TIA-825-A [Ref 3].

NOTE: TTY was originally an abbreviation for Teletype.

Interworking Function (IWF): IWF provides conversion between TTY and RTT.

⁵ This document is available from the 3rd Generation Partnership Project (3GPP) < <http://www.3gpp.org/> >

⁶ This document is available from the National Emergency Number Association. < <https://www.nena.org/> >

⁷ Titles I, II, III, and V of the original law are codified in Title 42, chapter 126, of the United States Code beginning at section 12101. Title IV of the original law is codified in Title 47, chapter 5, of the United States Code. Information on ADA is available from the United States Department of Justice < https://www.ada.gov/ada_title_II.htm >

3.2 Acronyms & Abbreviations

ADA	Americans with Disabilities Act
ALI	Automatic Location Identification
ANI	Automatic Number Identification
AOA	Angle of Arrival
ATIS	Alliance for Telecommunications Industry Solutions
CA	Communications Agent
CAMA	Centralized Automatic Message Accounting
CDR	Call Detail Record
CMSP	Commercial Mobile Service Provider
CPE	Customer Premises Equipment
CS	Circuit Switched
E9-1-1	Enhanced 9-1-1
EAAC	Emergency Access Advisory Committee
E-MF	Enhanced MF
EMS	Emergency Medical Services
ESN	Emergency Service Number
ESRD	Emergency Services Routing Digits
ESRK	Emergency Services Routing Key
ESZ	Emergency Service Zone
FCC	Federal Communications Commission
FSK	Frequency Shift Keying
GMLC	Gateway Mobile Location Center
GPS	Global Positioning System
GTT	Global Text Telephony
IETF	Internet Engineering Task Force
IMS	IP Multimedia Subsystem
IP	Internet Protocol
ISDN	Integrated Services Digital Network
ISUP	ISDN User Part
IWF	Interworking Function
LTE	Long Term Evolution
MF	Multi-Frequency
MLP	Mobile Location Protocol
MSC	Mobile Switching Center
NCAS	Non-Call Associated Signaling
NENA	National Emergency Number Association
NG	Next Generation
NG9-1-1	Next Generation 9-1-1

NPD	Numbering Plan Digit
PIDF-LO	Presence Information Data Format Location Object
PSAP	Public Safety Answering Point
PST	Public Safety Telecommunicator
PSTN	Public Switched Telephone Network
PTSC	Packet Technologies and Systems Committee
RFC	Request for Comment
RTT	Real Time Text
SIP	Session Initiation Protocol
SR	Selective Router
SRDB	Selective Routing Database
SS7	Signaling System No. 7
TDOA	Time Difference of Arrival
TS	Technical Specification
TTY	Teletypewriter or Text Telephony
URI	Uniform Resource Identifier
VoIP	Voice over IP
VoLTE	Voice over LTE
WCM	Wireline Compatibility Mode
WSP	Wireless Service Provider
WTSC	Wireless Telecommunications Systems Committee

4 Background Information

TTY, in the sense of a text telephone operating in the USA, is supported by regulatory requirements and was originally created for a circuit-switched environment.

In 2013, the FCC Emergency Access Advisory Committee (EAAC) recommended the use of a standardized Real Time Text (RTT) service as CMSP networks transition toward an all-IP network [Ref 1]. The community for individuals with disabilities (including deaf and hearing impaired) was overwhelmingly in favor of RTT as a service versus continuing to perpetuate TTY. However, the community for individuals with disabilities stressed that TTY devices must continue to be supported as long as circuit-switched networks exist, since the legacy TTY devices may be the only method of communication for certain individuals.

To support the transition to an all-IP network, the ATIS Packet Technologies and Systems Committee (PTSC) created a Technical Report (TR) for "Support of TTY Service over IP using Global Text Telephony" [Ref 2] which is based upon existing 3GPP Global Text Telephony (GTT) specifications as well as existing IETF specifications supporting RTT.

ATIS technical report ATIS-1000068 [Ref 2] provides network support for TTY, RTT, and GTT. The mobile device behavior for handling RTT in support of the IP transition is standardized in ATIS-0700029 [Ref 18] and supports consistent use of RTT across multiple CMSPs. The standardization of the Real Time Text end-to-end service description supports consistent use of RTT communication between CMSPs.

Annex A of this Standard provides example Use Cases for the usage of RTT service. Annex B provides information on RTT-TTY interworking enhanced capabilities. Annex C provides information on interaction for RTT with emergency services.

5 Assumptions

1. The CMSP IP network provides Voice over LTE (VoLTE) services and supports IETF RFC 4103 [Ref 4].
2. The RTT service user will access the CMSP IP network with an RTT-enabled mobile device as specified in ATIS-0700029 RTT Mobile Device Behavior (MDB) Standard [Ref 18].
3. RTT users will be able to call and be called by Next Generation and legacy Enhanced 9-1-1 (E9-1-1) PSAPs to the extent that this is supported for voice calls and will be able to use RTT media and voice media in the calls.
4. RTT service users will be able to call and be called by other RTT service users and use voice media and RTT media in the calls.
5. RTT service users will be able to call and be called by TTY device users in Circuit Switched (CS) networks outside of the CMSP's IP network.
6. Multimedia Telephony Service and supplementary services; Stage 1, as specified in 3GPP TS 22.173 [Ref 7], is applicable to RTT services. See clauses 4.1 and 4.2 of 3GPP TS 22.173 [Ref 7]. The interaction of Multimedia Telephony supplementary services as applied to the RTT service and Enterprise IP-PBX Supplementary Services as described in Annex D of 3GPP TS 22.173 [Ref 7] is for further study.

6 End-to-End RTT Service Requirements

6.1 RTT Service Requirements

6.1.1 RTT Call Requirements Between RTT Users

1. The RTT service shall support call initiation, routing, and call completion with real-time text and voice.
2. The RTT service shall utilize IMS call control support for RTT calls.

6.1.2 Emergency Service Call Requirements

1. RTT communications with 9-1-1 emergency services shall be accomplished by initiating a voice emergency call with an RTT media component (i.e., both audio and text media shall be offered in the establishment of the initial emergency call).
2. The dialed number for an RTT emergency call shall be the same number as a voice emergency call (e.g., 911).
3. The RTT equivalent of voice supplemental services that are disabled for voice emergency calls shall also be disabled for the RTT emergency calls.
4. The location of the RTT-enabled mobile device shall be determined by emergency call location methodologies.
5. The same callback number shall be used independent of the callback call media.

An overview of RTT emergency services is provided in Annex C.

6.1.3 General

The RTT service shall support voice and RTT media operation in both directions of a conversation with all four media streams occurring simultaneously. RTT service shall consist of RTT and voice media.

6.1.4 Inclusion of the RTT Media

It shall be possible to request inclusion of the RTT media by any party in the call initially in a call as well as during the call as defined in the RTT Mobile Device Behavior (MDB) specification [Ref 16].

6.1.5 Removal of the RTT media

It shall be possible to delete the RTT media from the call at any point in the call after it has been included, when requested by the RTT-enabled mobile device, as defined in the RTT Mobile Device Behavior (MDB) specification [Ref 18].

6.2 RTT Interworking with TTY

6.2.1 Use of Text and Audio Media

RTT/TTY transcoding functions shall be activated in RTT calls. For further detail, see ATIS-1000068 [Ref 2]. The IWF shall provide transcoding of signals between RTT and TTY devices and handle control of audio transmission between them.

Transcoding TTY to RTT:

1. Baudot Frequency Shift Keying (FSK) received from the CS network shall be intercepted and decoded. The resulting TTY text is translated to RTT-encoded text and transmitted to the RTT user in the text media. The original Baudot FSK signal shall not be transmitted to the RTT user in the audio media. Silence or comfort noise may be transmitted to the RTT user in the audio media.
2. When Baudot FSK is not being received from the CS network, the audio received shall be transmitted to the RTT user in the audio media.

Transcoding RTT to TTY:

1. When FSK is detected from the TTY user, no text or audio transmission should be sent to the TTY user.
2. When no FSK is detected from the TTY device, and RTT media is received or available (buffered as described in clause 6.2.2), the received or available RTT media shall be transcribed and transmitted to the TTY device user. No voice media from the RTT call shall be transmitted to the TTY device during this period.
3. When no FSK is detected from the TTY device, and no RTT text is available from the RTT service user for transcoding and transmission, the voice media received from the RTT service user shall be transmitted to the TTY device user.

6.2.2 Buffering RTT Text for TTY Transmission

Text media received from the RTT user for which immediate transcoding and transmission to the TTY device user is not possible (see clause 6.2.1) should be buffered. Buffer behavior and performance characteristics (e.g., buffer size) are based on operator policy.

6.2.3 Interworking Character Error Rate

The IWF shall accurately decode and encode all Baudot FSK characters and RTT text that are received without errors or excessive noise and are supported by TTY.

6.3 Transmission Rate

Transmission rate supported shall be as defined in the RTT Mobile Device Behavior (MDB) specification [Ref 18].

Annex A
(informative)

Annex A: Use Cases

This informative annex provides example use cases for RTT service usage on an end-to-end basis. These use cases are for illustrative purposes only to explain various potential scenarios when using RTT services. The RTT service implementations implied by the use case descriptions are illustrative only for the purpose of describing the associated scenario and these implied implementations should not be interpreted as the only or the preferred implementation option.

This informative annex describes the following use cases:

- Use Case 1 – RTT Emergency Call and Call-Back
- Use Case 2 – Establish Non-Emergency RTT Call
- Use Case 3 – Add RTT Media During Voice Emergency Call
- Use Case 4 – Add RTT Media During Voice Call
- Use Case 5 – RTT Call to TTY Device User in Public Switched Telephone Network (PSTN)
- Use Case 6 – Receive RTT Call from TTY Device User in PSTN
- Use Case 7 – RTT Call To 711 For Legacy Text Relay Call to Voice Service User
- Use Case 8 – Receive RTT Call From 711 Legacy Text Relay Service for Voice Service User
- Use Case 9 – RTT Call To 711 For RTT Based Text Relay Call to Voice Service User
- Use Case 10 – Initiation of A Three-Party RTT Call
- Use Case 11 – RTT Call with Deaf-Blind RTT Service User

A.1 Use Case 1 –RTT Emergency Call & Call-Back

This is a Use Case for emergency communications with a PSAP when the emergency communications is an RTT emergency call.

Short Description

Steve is a wireless service user with a mobile device. In an emergency situation, Steve invokes an RTT emergency call.

Actors

- Steve – Mobile device user.
- Carol – Public Safety Telecommunicator (PST).

Pre-Conditions

- Steve's Service Provider's network supports RTT capabilities.
- Steve's mobile device supports RTT emergency service functionality.
- Steve is subscribed to his Service Provider.
- Steve is connected to his Service Provider's network.
- Steve's mobile device does not have any active sessions (e.g., RTT call, voice call, messaging) prior to Steve initiating an RTT emergency call to 9-1-1.

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- The PSAP covering a region within Steve's Service Provider network is capable of supporting RTT emergency calls
- Coarse location information (e.g., cell site location) is available for the determination of the appropriate PSAP for routing of the RTT emergency call request to emergency services.

Post-Condition

Steve was able to communicate to the PSAP successfully via an RTT emergency call using his mobile device.

Normal Flow

1. Steve already has his mobile device turned on.
2. Steve encounters an emergency situation and decides to communicate with 9-1-1 emergency services.
3. Steve accesses the RTT capability on his mobile device, enters the three digits "911" as the number to be called using the mobile device internal keyboard, and selects the "Initiate RTT" button on the internal keyboard to request an RTT emergency call.
4. The appropriate PSAP is selected based upon the cell site serving Steve, and the RTT emergency call request is delivered to the specified PSAP.
5. Carol is a PST and receives Steve's RTT call.
6. Carol accepts Steve's RTT call, and a communication session with both voice and RTT text is established between Steve and Carol.
7. Carol and Steve may exchange text communications via RTT media in order for Steve to describe the overall emergency situation.
 - a. If requested by Carol, Steve may describe his location in the RTT media via the established RTT emergency call.
 - b. Since this communication session also includes a voice component, Carol and Steve can also use voice.
 - c. Because there is an "always on" audio component in this call, Carol can listen to background noises in Steve's environment in accordance with the PSAP's policies and procedures and Steve has not muted his microphone.
8. Carol determines that dispatching emergency services is warranted and creates a request for service in her PSAP's dispatch system.
9. Carol ends the communication session according to the PSAP's policies and procedures.

Alternative Flows

It is possible that the RTT emergency call could be ended prematurely due to conditions such as communication errors or the accidental selection of the "Disconnect RTT" button by Steve in the mobile device RTT service capability. The usual PSAP procedures would be for the PST to initiate a call-back function. This alternative flow describes only one of the possible scenarios for the PSAP to perform a call-back.

If this premature disconnect condition occurs, the following additional steps could occur between steps 7 and step 8 of the normal flow:

- 7.1 The RTT call between Carol and Steve has ended prematurely, and Carol recognizes that the RTT emergency call has ended prematurely.
- 7.2 Per PSAP policies and procedures, Carol would initiate a request for an RTT call from the PSAP to Steve's mobile device using the telephone number of Steve's mobile device that was received during Steve's initial call request.
- 7.3 Steve's mobile device displays that an RTT call request has been received from "emergency services".

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NOTE: the actual value of the incoming call identification displayed for “emergency services” is determined by the PSAP practices and policies and may be the same incoming call identification that the PSAP uses for voice only call-back operations.

7.4. Steve accepts Carol's RTT call request, and an RTT call is re-established between Steve and Carol.

At this point, the normal flow continues with step 8.

A.2 Use Case 2 – Establish Non-Emergency RTT Call

This is a use case for text communications between two users in an RTT call

Short Description

Steve is a wireless service user with a mobile device. Steve places an RTT call to his friend Anne.

Actors

- Steve – Mobile device user.
- Anne – Another mobile device user.

Pre-Conditions

- Steve's and Anne's Service Provider's networks support RTT service capabilities.
- Steve's mobile device supports RTT functionality.
- Steve needs RTT media in his calls, so therefore he has made a setting that shows RTT (RTT and voice media) as a type of call that can be made.
- Anne's mobile device supports RTT functionality.
- Steve and Anne are subscribed to their Service Providers.
- Steve and Anne are connected to their Service Provider's networks.
- Steve's and Anne's mobile devices do not have any active sessions (e.g., RTT calls, voice calls, messaging) prior to Steve initiating an RTT call to Anne's.

Post-Condition

Steve was able to communicate with Anne successfully via an RTT call using his mobile device.

Normal Flow

1. Steve already has his mobile device turned on.
2. Steve decides to communicate with Anne by the RTT service.
3. Steve accesses the RTT service capability on his mobile device, looks up Anne's entry in the phonebook in the device, selects the entry, and clicks the button for an RTT call.
4. Because of the RTT call type selection, the call will be made with both voice media and RTT media from the beginning of the RTT call.
5. The request for an RTT call is delivered to Anne's service provider and further on to Anne's device.
6. Anne's device shows that there is an incoming RTT call so that Anne is made aware that there might be a need to communicate with RTT text.

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7. Anne accepts Steve's RTT call, and a communication session with both voice and RTT media is established between Steve and Anne. Anne knows Steve and that he prefers to communicate in RTT text, so she answers, "Hi Steve" on the on-screen keyboard.
8. Steve knows that Anne hears well, so he starts his communication with voice.
9. Anne and Steve may continue exchange text communications via RTT media and voice media according to what is suitable for the current situation. RTT media is immediate, so Steve never needs to wait for completed sentences or lines of text. Steve is able to read Anne's communication concurrently with Anne's typing.
10. After a couple of minutes of typing and talking, Steve and Anne are ready to end the call and say/type "bye".
11. Steve ends the RTT call by pressing the end of call button.
12. Anne's phone detects the end of call and closes the RTT call.

Alternative Flows

None.

A.3 Use Case 3 – Add RTT Media During Voice Emergency Call

This is a use case for emergency communications with a PSAP when the emergency communications are with voice initially, and RTT media is added during the voice emergency call.

Short Description

Steve is a wireless service user with a mobile device. In an emergency situation, Steve invokes communication with the PSAP, initially with voice media only, but with RTT media added later to the voice emergency call.

Actors

- Steve – Mobile device user.
- Carol – Public Safety Telecommunicator (PST).

Pre-Conditions

- Steve's Service Provider's network supports RTT service capabilities.
- Steve's mobile device supports RTT emergency service functionality via the three-digit code 911.
- Steve's mobile device does not have any active sessions (e.g., RTT call, voice call, messaging) prior to Steve initiating a voice emergency call to 9-1-1.
- The PSAP covering a region within Steve's Service Provider network is capable of supporting end-users' RTT emergency service requests.

Post-Condition

Steve was able to communicate to the PSAP successfully via voice media using his mobile device and add and use RTT media later in the RTT emergency call.

Normal Flow

1. Steve already has his mobile device turned on.
2. Steve encounters an emergency situation and decides to communicate with 9-1-1 emergency services.
3. Steve accesses the voice emergency call capability on his mobile device, enters the three digits 911 as the number to be called using the mobile device internal keyboard, and selects the "Call" button on the

- device to request a voice emergency call.
4. The appropriate PSAP is selected based upon the cell site serving Steve, and the voice emergency call request is delivered to the specified PSAP.
 5. Carol is a PST and receives Steve's voice emergency call request.
 6. Carol accepts Steve's voice emergency call request and an emergency communication session with voice media is established between Steve and Carol.
 7. Carol asks Steve about a medicine name. Steve knows how time consuming it is to try to spell its complicated and long name by voice and decides to add an RTT media component to the call for typing the medicine name instead. Steve touches a text input area for the voice emergency call.
 8. Steve's device requests the addition of the RTT media component to the emergency call. When this is acknowledged by Carol's device, an on-screen-keyboard appears to Steve.
 9. Steve types the medicine name to Carol. Carol gets it character by character as Steve types.
 10. Before he has completed typing the medicine name, Steve feels worse and cannot continue. He just can say. "I am worse, come!"
 11. Carol got enough of the medicine name to guess the complete name and can thereby plan a rapid and efficient emergency response action.
 12. Carol can inform Steve about the action and end the emergency call, according to the PSAP's policies and procedures.

Alternative Flows

None.

A.4 Use Case 4 – Add RTT Media During Voice Call

This is a use case for communications between two mobile device users of voice services who add RTT media during the call.

Short Description

Steve is a wireless service user with a mobile device. Steve initiates an audio call to his friend Anne. During the voice call, Anne wants Steve to convey something in text to Anne in real-time, so she adds RTT media to the voice call and Steve types the text to send to Anne.

Actors

- Steve – Mobile device user.
- Anne – Another mobile device user.

Pre-Conditions

- Steve's and Anne's Service Provider's networks support RTT services.
- Steve's mobile device supports RTT service functionality.
- Anne's mobile device supports RTT service functionality.
- Steve and Anne are subscribed to their Service Providers.
- Steve and Anne are connected to their Service Provider's networks.
- Steve's and Anne's mobile devices do not have any active sessions (e.g., RTT call, voice call, messaging) prior to Steve initiating a voice call to Anne.

Post-Condition

Steve was able to communicate with Anne successfully via voice media using his mobile device and also via RTT media after adding the RTT media to the voice call.

Normal Flow

1. Steve already has his mobile device turned on.
2. Steve decides to communicate with Anne by voice call.
3. Steve accesses the voice call capability on his mobile device, looks up Anne in the phonebook in the device, selects the entry, and clicks the voice call button.
4. The request for a voice call is delivered to Anne's service provider and further on to Anne's device.
5. Anne's device shows that there is an incoming voice call, so that Anne is made aware that answering by voice is sufficient.
6. Anne accepts Steve's voice call, and a communication session with voice media is established between Steve and Anne.
7. Anne and Steve continue talking to each other.
8. Anne ends up in a noisy environment where it is hard to hear what Steve says. Anne touches a text input area on the screen. That causes a request to Steve's device to add an RTT media component to the voice call.
9. Steve's device acknowledges the RTT media addition.
10. Steve can type, and the RTT media is sent to Anne as he is typing.
11. After a couple of minutes of typing and talking, Steve and Anne are ready to terminate the call and say/type "bye".
12. Steve ends the communication session by pressing the end of call button.
13. Anne's phone detects the end of call and closes the RTT call.

Alternative Flows

None.

A.5 Use Case 5 – RTT Call to TTY in PSTN

This is a use case for communications between a mobile RTT service user and a TTY user in PSTN, using RTT media alternating with voice media during the RTT call.

Short Description

Anne is a wireless service user with a mobile device with RTT service functionality. Anne initiates a text and voice call to an elder relative Beatrice who uses a TTY device.

In a variant, Anne is deaf and only uses text media in the RTT call.

Actors

- Anne – Mobile device user.
- Beatrice – An elder user of TTY through the PSTN.

Pre-Conditions

- Anne's Service Provider's networks support RTT service capabilities.
- Anne's mobile device supports RTT service functionality.
- Beatrice's TTY device is connected to a PSTN service provider, together with a voice device.
- Anne is subscribed to her wireless Service Provider.
- Anne's mobile device do not have any active sessions (e.g., RTT call, voice call, messaging) prior to Anne initiating an RTT call with RTT to Beatrice.

Post-Condition

Anne was able to communicate with Beatrice successfully in an RTT call using her mobile device.

Normal Flow

1. Anne already has her mobile device turned on.
2. Anne decides to communicate by an RTT call with Beatrice on her TTY device.
3. Anne accesses the RTT service capability on her mobile device, enters Beatrice's phone number, and clicks the RTT call button, because she knows that Beatrice needs to use her TTY device.
4. The request for an RTT call to a PSTN destination activates an RTT / TTY interworking function.
5. The interworking function sends a short RTT text message to Anne, indicating that the call is going to the PSTN and if a TTY device answers, the user need to obey the TTY conventions waiting for the turn indicator 'GA' and ending turns with sending 'GA'. This information ends with '...' indicating that Anne should wait for an answer.
6. Beatrice's phone and the TTY device shows that there is an incoming call.
7. Beatrice answers Anne's call with the TTY device and types a greeting phrase followed by 'GA'.
8. Anne knows that Beatrice does not hear well, but wants to speak, so Anne types a greeting and 'Over to voice GA'.
9. Then Anne and Beatrice can continue the conversation taking turns with Anne typing and Beatrice talking. Anne remembers to take proper time when changing turn so that Beatrice has time to rearrange her phone and TTY device for text reception for each turn.
10. After some time of typing and talking, Anne and Beatrice are ready to terminate the call and say/type "bye".
11. Anne ends the RTT call by pressing the end of RTT call button.
12. Beatrice hangs up on the phone connected to the PSTN subscription, happy that she can have good direct contact with the younger ones who are always on the go.

Alternative Flows

An alternative is that Anne is deaf and does not use speech in voice calls. Points 7, 8, and 9 will be performed only with typing text back and forth between Anne and Beatrice. They respect the TTY turn-taking conventions giving turn by typing 'GA' and not typing against turn.

A.6 Use Case 6 – Receive RTT Call from TTY in PSTN

This is a use case for communications between a mobile RTT service user and a TTY device user via the PSTN, using text alternating with voice during the call. The TTY device user originates the call.

Short Description

Beatrice is an elder person using a TTY device connected to her voice phone for communication via the PSTN. She does not hear well and prefers to talk in the phone and get text back on the TTY. Anne is a wireless service user with a mobile device with RTT service functionality. Beatrice calls Anne and they conduct a TTY / RTT call.

Actors

- Anne – Mobile device user.
- Beatrice – An elder user of a TTY device through the PSTN.

Pre-Conditions

- Anne's Service Provider's networks support RTT services.
- Anne's mobile device supports RTT service functionality.

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- Beatrice's TTY device is connected to a PSTN service provider, together with a voice device.
- Anne is subscribed to her wireless Service Provider.
- Anne's mobile device do not have any active sessions (e.g., RTT call, voice call, messaging) prior to Anne initiating an RTT call to Beatrice.

Post-Condition

- Anne was able to successfully communicate with Beatrice in text media and voice media using her mobile device.

Normal Flow

1. Anne already has her mobile device turned on.
2. Beatrice decides to communicate by TTY with Anne on her RTT-capable phone.
3. Beatrice enters Anne's mobile phone number on the phone connected to her TTY device and makes the call.
4. The call goes through to Anne's mobile phone.
5. Anne has Beatrice's number in her mobile device's phonebook, so when the voice call arrives it shows that Beatrice is calling. Anne knows about Beatrice's preference to use the TTY device, so she answers the voice call with the option to answer as an RTT call.
6. The call is initially placed as a voice call. Anne's answering as an RTT call causes Anne's handset to first answer the call with audio media, and then immediately request RTT media to be added to the voice call.
7. The addition of RTT media to a voice call initiated in the PSTN activates the interworking function between RTT calls and TTY. The interworking function sends a short RTT text message to Anne, indicating that the call is with a PSTN user and if a TTY user was calling, the RTT service user needs to obey the TTY conventions waiting for the turn indicator 'GA' and ending turns with sending 'GA'. This information ends with 'GA' indicating that Anne should enter her initial greeting.
8. Anne types a greeting ended by "over to voice GA" and listens to the phone.
9. Beatrice gets Anne's greeting phrase on the TTY device and talks back ended by "over".
10. Anne knows that Beatrice does not hear well, but wants to speak, so Anne types next phrase and "Over to voice GA".
11. Anne and Beatrice continue the conversation, taking turns with Anne typing and Beatrice talking. Anne remembers to take proper time when changing turn so that Beatrice has time to rearrange her phone and the TTY device for TTY reception for each turn.
12. After some time of typing and talking, Anne and Beatrice are ready to terminate the call and say "bye"/type "SKSK".
13. Anne ends the RTT call by pressing the end of RTT call button.
14. Beatrice hangs up on the phone connected to the PSTN subscription, happy that she can have good direct contact with the younger ones who are always on the go.

Alternative Flows

Alternative flow 1: The TTY user calls from an unknown phone number.

In an alternative flow, points 5 and 6 above have the following variation:

- 5 a. Beatrice calls from an unknown PSTN voice device from a number that is not in Anne's mobile device's phonebook. Anne has no idea that this is a TTY call, and therefore answers by speaking a greeting.
- 6 a. The voice causes a flickering light on Beatrice's TTY device, so she then understands that Anne answered by voice and is listening to the mobile phone. Beatrice therefore says, "Hi, it is Beatrice, activate your RTT, over".
- 6 b. Anne hears that and presses a button on her RTT service capable phone that requests RTT media to be added to the voice call.

The call continues as from point 7 above.

Alternative Flow 2: The receiving mobile user has no previous experience of RTT or TTY.

This is another alternative flow, with the following variation:

Beatrice is deaf and does not use voice. Anne is hearing and has no previous experience of TTY or RTT services.

5 a. Beatrice calls Anne's number from the PSTN phone attached to the TTY device. Anne has no idea that this is a TTY call, and therefore answers by speaking a greeting.

6 a. Anne's voice causes a flickering light on Beatrice's TTY device, so she then understands that Anne answered by voice media and is listening to the mobile phone. Beatrice therefore taps the spacebar on the TTY a number of times, causing TTY tones to reach Anne.

6 b. Anne hears and is aware that the tones she hears are from a person who wants to communicate through TTY. She sees the "add text" (RTT) button and takes a chance to press that button on her RTT service capable phone. That causes a request to add RTT media to the call.

The call continues as from point 7 above.

NOTE: The functionality is described according to how ATIS-1000068 [Ref 2] and 3GPP TS 29.163 [Ref 16] Annex I specify it.

A.7 Use Case 7 – RTT Call to 711 for Legacy Text Relay Call to Voice Service User

This use case is about using the legacy text relay service available via dialing "711" for a relayed call with a voice service user.

Short Description

Steve is a wireless service user with a mobile device with RTT service functionality. Steve is deaf. For a call to David, Steve calls "711" to get support with conversion between RTT media and voice media by a text relay service communications agent.

Actors

- Steve – Mobile device user, preferring text communication.
- Celia – Text relay communications agent (CA).
- David – Voice phone user unfamiliar with RTT services.

Pre-Conditions

- Steve's Service Provider's networks support RTT service capabilities.
- Steve's mobile device supports RTT service functionality.
- David's voice phone is ready to have calls.
- Steve's mobile device is subscribed to his Service Provider.
- Steve's mobile device does not have any active sessions (e.g., RTT call, voice call, messaging) prior to Steve initiating an RTT call to "711".

Post-Condition

Steve was able to successfully communicate with David; Steve via RTT media and David via voice media, with the text relay service CA converting between the media.

Normal Flow

1. Steve already has his mobile device turned on.
2. Steve decides to communicate with David using RTT service with support from the text relay service.
3. Steve accesses the RTT service capability on his mobile device, enters “711” for the legacy text relay service, and clicks the button for an RTT call.
4. The request for an RTT call to a PSTN destination (711) activates an RTT / TTY interworking function to assist in facilitating the communication.
5. The interworking function sends a short RTT text message to Steve, indicating that the call is going to the PSTN destination and if a TTY device answers, the user needs to obey the TTY conventions waiting for the turn indicator 'GA' and ending turns with sending 'GA'. This information ends with '...' indicating that Steve should wait for an answer.
6. The “711” text relay service receives an incoming voice call.
7. Celia, working as a CA in the text relay service answers Steve’s call with the TTY device integrated in the text relay system and types a greeting phrase asking who Steve wants to call, followed by 'GA'.
8. Steve answers with David's phone number followed by 'GA'.
9. Celia calls David's number and keeps Steve informed about the progress of the call. When David has answered via voice, Celia informs David by voice that it is a relayed call, and then she types David's greeting phrase to Steve followed by 'GA'.
10. Steve and David now take turn in typing/talking and Celia translates between them. They remember to type 'GA'/say “Go Ahead” to indicate giving turn.
11. After some time of typing and talking, Steve and David are ready to terminate the call and say/type "bye". Steve also types 'SKSK'.
12. Steve ends the RTT call by pressing the end of RTT call button.
13. David hangs up the phone.

Alternative Flows

None.

A.8 Use Case 8 – Receive RTT Call from 711 Legacy Text Relay Service for Voice Service User

This use case is about using the legacy text relay service available via dialing “711” for a relayed call with a voice service user calling and a mobile RTT service user answering.

Short Description

Steve is a wireless service user with a mobile device with RTT service functionality. Steve is deaf. To initiate a call to Steve, David, a hearing person, calls “711” to get support for a conversion between RTT media and voice media by a text relay service communications agent.

Actors

- Steve – Mobile device user, preferring RTT services.
- Celia – Text relay communications agent (CA).
- David – Voice service user unfamiliar with RTT services.

Pre-Conditions

- Steve's Service Provider's networks support RTT service capabilities.
- Steve's mobile device supports RTT service functionality.
- Steve's mobile device does not have any active sessions (e.g., RTT call, voice call, messaging) prior to David initiating a voice call to "711".

Post-Condition

Steve was able to successfully communicate with David; Steve via RTT media and David via voice media, with the text relay service converting between the media.

Normal Flow

1. Steve already has his mobile device turned on.
2. David decides to communicate with Steve by voice via the text relay service, which handles the conversion between text and speech.
3. David calls "711" for the legacy text relay.
4. The voice call arrives at the legacy text relay service, and Celia, working as a CA, answers it.
5. Celia asks David who he wants to call, and David responds verbally with Steve's mobile phone number.
6. Celia calls Steve's number from the relay service.
7. Steve answers with RTT media and voice media activated.
8. The call is first established with voice media only, but Steve's mobile device automatically adds RTT media when the initial RTT call is established.
9. The request by Steve's mobile device to add RTT media to a voice call from a PSTN destination activates an RTT / TTY interworking function to assist in facilitating the communication.
10. The interworking function sends a short RTT text message to Steve, indicating that the call is going to the PSTN and if a TTY answers, the user needs to obey the TTY conventions of waiting for the turn indicator 'GA' and ending turns with sending 'GA' This message ends with 'GA' indicating that Steve should provide an answer.
11. The "711" text relay service receives the answer.
12. Celia, working as a CA in the text relay service gets Steve's answer with the TTY device integrated in the text relay system and types a greeting phrase ending with '...' and continued with a greeting message from David.
13. Steve and David now take turn in typing / talking and Celia translates between them. They remember to type 'GA' or say "Go Ahead" to indicate giving turn.
14. After some time of typing and talking, Steve and David are ready to terminate the call and say/type "bye". Steve also types 'SKSK'.
15. Steve ends the RTT call by pressing the end of call button.
16. David hangs up the voice call.

Alternative Flows

None.

A.9 Use Case 9 – RTT Call to 711 for RTT based Text Relay Call to Voice Service User

This use case is about using a text relay service supporting RTT services for a relayed call with a voice service user.

Short Description

Steve is a wireless service user with a mobile device with RTT service functionality. Steve is deaf. To initiate a call to Steve, David, a hearing person, calls “711” to get support for conversion between RTT media and voice media by a text relay service communications agent.

Actors

- Steve – Mobile device user, preferring RTT services.
- Celia – Text relay communications agent (CA).
- David – Voice service user unfamiliar with RTT services.

Pre-Conditions

- Steve’s Service Provider’s networks support RTT service capabilities.
- Steve’s mobile device supports RTT service functionality.
- Steve’s mobile device does not have any active sessions (e.g., RTT call, voice call, messaging) prior to Steve initiating an RTT call to 711.

Post-Condition

Steve was able to successfully communicate with David; Steve via RTT media and David via voice media, with the text relay service converting between the media.

Normal Flow

1. Steve already has his mobile device turned on.
2. Steve decides to communicate with David using RTT with support from the text relay service.
3. Steve accesses the RTT service capability on his mobile device, enters the number “711” for the text relay and clicks the button for an RTT call.
4. The request for an RTT call to an IP-based RTT-capable text relay service.
5. The 711 text relay service receives an incoming RTT call.
6. Celia, working as a CA in the text relay service, answers Steve’s call with the RTT service functionality integrated in the text relay system and types a greeting phrase asking who Steve wants to call.
7. Steve provides David’s phone number.
8. Celia calls David’s number and keeps Steve informed about the progress of the voice call. When David has answered, Celia informs David that it is a relayed call, and then she types David’s greeting phrase to Steve.
9. Steve and David now take turns typing and talking, and Celia translates between them. The flow is smooth without any turn-taking tokens.
10. After some time of typing and talking, Steve and David are ready to end the call and type/say “bye”.
11. Steve ends the communication session by pressing the end of RTT call button.
12. David hangs up the voice call.

Alternative Flows

None.

A.10 Use Case 10 – Initiation of a Three-Party RTT Call

This use case is about making a three-party RTT call.

Short Description

Steve decides to make an RTT call with both Anne and Robin simultaneously. He calls Anne and adds Robin as the third party in the RTT call. All can communicate by RTT media and voice media.

Actors

- Steve, Anne, and Robin, all mobile device users.

Pre-Conditions

- Steve, Anne, and Robin have their mobile phones registered and have RTT service functionality.
- Steve, Anne, and Robin have no other voice or RTT call going on when Steve wants to initiate the three-party RTT call.

Post-Condition

Steve, Anne, and Robin have had a three-party RTT call where all can read the other's RTT media.

Normal Flow

1. Steve makes an RTT call to Anne by clicking the RTT call button.
2. Anne answers and responds with RTT media.
3. Steve informs Anne via the RTT media that he will also invite Robin to the RTT call.
4. Steve sees an *add person* symbol in the dialer user interface and adds Robin.
5. The call is handled by an RTT capable 3 way calling service platform, and Robin is added to the 3 way call.
6. Steve, Anne, and Robin communicate in the three-party RTT call.
7. When they are finished, they type "bye" and hang up the three-party RTT call.

Alternative Flows

None.

A.11 Use Case 11 – RTT Call with Deaf-Blind RTT Service User

This is a use case for communications between two users of the RTT service with one of the users being deaf-blind.

Short Description

Robert is a deaf-blind wireless service user with a mobile phone. Robert is a braille user, so he also has an assistive technology, a braille device with braille keyboard and 16 position braille display, which connects to mobile devices via Bluetooth. He uses this braille device in combination with a screen reader function with braille support that is

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built into his mobile phone. Robert calls his friend Anne and makes use of the RTT service capability in his mobile device during the RTT call. Anne has a mobile device with RTT service capability as well.

NOTE: Deaf-blindness does not necessarily mean that a person is completely deaf and blind. Although some are, other people who are categorized as deaf-blind may still have some residual vision and/or hearing. Thus, some will use braille in both directions, while others may use visual enlargement, large print keyboards, amplifiers with regular phone audio and other devices in one or the other direction with or without braille. This example deals with braille support for those whose vision and hearing are such that it is required for them to be able to converse with others over the phone.

Actors

- Robert – Deaf-blind mobile device user with an RTT service capable mobile device and a Bluetooth-connected braille device with (braille) display and keyboard.
- Anne – Another mobile device user with an RTT service capable mobile device.

Pre-Conditions

- Robert's and Anne's Service Provider's networks support RTT service capabilities.
- Robert's mobile device supports RTT service functionality.
- Robert needs RTT media in his calls so therefore he has made a setting that makes RTT call the default type of call.
- Anne's mobile device supports RTT service functionality.
- The dialer application on Robert's mobile device is written to conform to the accessibility guidelines for the platform so that the phone dialer will work with the built-in screen reading functions of the phone.
- Robert and Anne are subscribed to their Service Providers.
- Robert and Anne are connected to their Service Provider's networks.
- Robert's and Anne's mobile devices do not have any active sessions (e.g., RTT call, voice call, messaging) prior to Robert initiating an RTT call to Anne.

Post-Condition

Robert was able to communicate with Anne successfully via RTT media using his mobile device.

Normal Flow

1. Robert already has his mobile device and the braille display turned on.
2. Robert decides to communicate with Anne by an RTT call.
3. Using the braille device compatible screen-reader functionality of the mobile device, and the keys on the braille device Robert, looks up Anne's entry in the phonebook in the mobile device, selects the entry, locates the logical call button for an RTT call, and activates the call button. All these actions are done through the braille keyboard/display and the screen reader of the mobile device.
4. Because of the RTT call type selection on Robert's mobile device, the RTT call will be made with RTT media requested from the beginning.
5. The request for an RTT call is delivered to Anne's mobile device.
6. Anne's mobile device shows that there is an incoming RTT call, so Anne is made aware that there might be a need to answer via text media.
7. Anne accepts Robert's RTT call, and an RTT communication session is established between Robert and Anne. Anne knows Robert and that he needs RTT media, so she answers, "Hi Robert" on the on-screen keyboard.

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8. On Robert's braille display he feels that new RTT media has been received, so he starts reading the incoming RTT media. His reading cursor is positioned to the latest incoming RTT media, so the display follows received RTT media. Newly received characters are appended to the display as soon as they are received. When Robert gets behind in reading the new text, he scrolls back in the RTT media history to where he wants to read, then the automatic scrolling and following new RTT media presentation is paused, and Robert can easily read what was received.

NOTE: Most or all of this is provided as a function of the screen reader and braille display and the platform-specific accessibility guidelines are followed.

9. Robert reads the greeting from Anne and types back. If Anne transmits anything while Robert is transmitting, Robert can decide if he want to continue typing or read what is being received in real time.
10. Anne and Robert continue to exchange RTT media communications via the RTT call.
11. After a while, Robert and Anne are ready to end the RTT call and one types "bye".
12. One of them ends the RTT call by using the RTT hang up button.
13. The other's phone detects the hang-up and closes the RTT call.

Alternative Flows

None.

Annex B: RTT/TTY Interworking Enhanced Capabilities

B.1 Summary of TTY Limitations

TTY communication technology has a number of significant limitations compared to RTT communication technology. However, during the transition from TTY to RTT communication, service interworking (transcoding) is an FCC-mandated capability [Ref 19] to deploy alongside RTT services. Many new RTT users, during this transition, are not likely to be familiar with these TTY workarounds and conventions. These new users may benefit from the automation of these workarounds and conventions when communicating with TTY users.

The topic of RTT-TTY interworking at the presentation or user level has been previously discussed in FCC EAAC reports [Refs 1, 5, & 6].

The limitations of TTY communications, which have generally accepted TTY user conventions and workarounds, is summarized as follows:

- **One transmission direction at a time (half duplex).**

While the TTY device is transmitting, it cannot receive. If someone tries to transmit at the same time, the characters may be garbled or lost. Therefore users are expected to take turns typing, and give turn by using specific text tokens. The common token during the call is the letters "GA". During ending of the call, combinations with "SK" are also used, such as "SK", "SKSK", "GASK", "SKGA". A space or new line may sometimes follow these tokens. A request to wait without giving the turn is usually indicated by "...".

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

While the TTY device handles text, voice cannot be transmitted. Alternating between audio and text can be done during the same call but must be formally signaled between the users. Going from text to speech is signaled by the text tokens for giving turn described above. Changing from a voice conversation to text conversation is signaled in the voice path by saying the same tokens.

- **Limited text throughput.**

TTY devices can only transmit between 3 and 6 characters per second, typically slower than many people can type, easily causing backlog in communication. The variation stems from occasional insertion of shift characters.

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

The alphabetic character set of TTY is limited to a single case.

- **Limited character set.**

Many special characters in common usage are missing from the TTY character set. The following special characters cannot be used in TTY communication:

- At sign character "@"
- Octothorpe or hash sign character "#"
- Percentage character "%"
- Ampersand character "&"
- Backslash character "\"
- Asterisk character "*"
- Underscore character "_"
- Less than sign character "<"
- Greater than sign character ">"

- **Limited display area.**

Many TTY devices have only one or two lines for display of TTY text making it inconvenient to use new lines in text to a TTY device. Many TTY devices therefore are designed to not transmit any new line characters. The TTY users rather use the turn token 'GA' as delimiter and indicator of end of message.

B.2 TTY Presentation Level Conventions & Workarounds

In addition to the TTY limitations summarized in clause B.1, the following RTT specific RTT/TTY interworking capabilities, if supported by the interworking function, may assist in enhancing clear and understandable communication between RTT service users and TTY users.

- **Include an indication to the RTT service user** that TTY interworking is active in the RTT call.
- **Buffer characters received from the RTT service user** while receiving text from the TTY user. See clause 6.2.2 for more information.
- **Conversion of RTT service special characters that are not supported by TTY.**

Special characters sent from the RTT service user not supported by TTY can be converted to TTY supported characters similar to the RTT service special character. This converted special character should be sent to the TTY user according to the TIA-825-A standard [Ref 3], for insertion of shift character and use of the 5-bit code.

Convert RTT service special characters to TTY-supported characters according to the following table:

Table B.1 – Conversion of Non-Supported RTT Characters

Original Characters	Character Conversions
At sign character "@"	Replace with "(at)"
Octothorpe or hash sign character "#"	Replace with dollar sign character "\$"
Percentage character "%"	Replace with slash character "/"
Ampersand character "&"	Replace with plus sign character "+"
Asterisk character "**"	Replace with a period character "."
Underscore character "_ "	Replace with space character " "
Less than sign character "<"	Replace with left parenthesis character "("
Greater than sign character ">"	Replace with right parenthesis character ")"
National character	Replace with their closest companion in the a-z character range – e.g., "ñ" => "n".
Unknown character	Replace with apostrophe character '

- **Announcing the intended use of TTY in a CS call.**

There is currently no facility to signal out-of-band in a CS network that a TTY device is or will be used on a call, and the other party should prepare to send and receive TTY communication. The TTY device workaround has been to pause about six seconds between the sequences and send a sequence of TTY characters (such as four space characters with short pauses in between) after CS call establishment until a TTY character is sent by either the TTY device user or the called party in the CS call. Detection of TTY communication is active during all pauses. The timing pattern is aimed at detection by TTY detection equipment used by legacy PSAPs and by TTY device users, for rapid detection, and not interrupting TTY text being received. A similar procedure may be implemented for the RTT-TTY interworking case. This feature is particularly important for 9-1-1 calls to PSAPs relying on TTY devices to reduce the call taker's time to recognize that the emergency call will include the use of TTY devices. Refer to Annex C for more details of E9-1-1 calls.

- **Conversion between text delimiter conventions in TTY vs RTT communication.**

Text from TTY devices often lack line delimiters (CRLF). Instead, a line delimiter is indicated by the TTY device user typing turn-taking token characters "_GA", "_GASK", or "_SKSK" (where the underscore indicates a space character), possibly followed by a space character. RTT service conversation presentation in text chunks is typically delimited by a new line coded as a Line Separator. To improve the readability of the TTY device originated text during RTT calls, a potential solution is to insert a Line Separator after each turn-taking-token that is not already followed by a line delimiter.

Annex C
(informative)

Annex C: RTT Emergency Services

One of the objectives for the support of RTT service communications is to provide direct communications with emergency services. This informative annex discusses the use of RTT services to support the ability for users to contact emergency service PSAPs via the same procedures used by voice service callers, e.g., by dialing the digits “9-1-1”. The use of RTT service communications to access emergency services will require the interconnection of IMS originating networks with legacy and Next Generation (NG) (i.e., IP-based) Emergency Services Networks to support communications with both legacy and NG PSAPs.

This informative annex includes an overview of emergency services and describes the use of RTT services to support emergency communications with PSAPs. The overview of emergency services provided in this annex is a high-level summary of the handling of RTT emergency calls. This overview does not cover all possible environments and scenarios. The full descriptions of the capabilities for emergency services are in the relevant ATIS, IETF, and National Emergency Number Association (NENA) standards, and these standards continue to be enhanced and expanded. If there are any inconsistencies between this high-level overview and the emergency services standards, the relevant ATIS, IETF, and NENA emergency services standards take precedence.

C.1 Overview of Emergency Services

In the United States, the entity which receives 9-1-1 calls is referred to as a PSAP. Each PSAP provides 9-1-1 emergency services to defined geographic areas (e.g., municipality, county). Each PSAP operates under the policies and processes defined by the associated state, county, and/or municipality.

The PSAPs receive 9-1-1 emergency calls from traditional wireline telephones, cellular mobile devices, and Voice over Internet Protocol (VoIP) telephones, and in the US, an estimated 240 million calls are made to 9-1-1 emergency services each year. The 9-1-1 emergency calls are estimated to be about 70% from mobile devices, about 30% from wireline telephones and about 1% from VoIP telephones. The number of US households which are wireless only is increasing each year, and as of the end of 2015, about 40% of the US households are wireless only. Consequently, the percentage of 9-1-1 emergency calls from mobile devices will probably continue to increase.

Version 2.229 of the FCC 911 Master PSAP Registry⁸ which was last updated on October 5th, 2017 lists 6,326 primary PSAPs⁹ and 627 secondary PSAPs¹⁰ in the US. Each PSAP can either be a next generation IP-based (NG9-1-1) PSAP or a legacy PSAP which is not IP-based and is a pre-NG9-1-1 PSAP. The 9-1-1 emergency call will be delivered to either the NG9-1-1 PSAP or the legacy PSAP associated with the calling party’s location. Interworking functions will be used to support interoperability between the technology of the calling party’s mobile device and the corresponding PSAP.

There is a large variance in the size of the PSAPs. The PSAPs serving large metropolitan areas may have many PSAP telecommunicator positions with associated call distribution and Computer Aided Dispatch (CAD) systems. The PSAPs serving small rural communities or counties could use the desk phones in the Sheriff’s office. It is estimated that at least 80% of the PSAPs have three or less PSAP telecommunicator positions.

This annex provides a high-level overview of Enhanced 9-1-1 and Next Generation 9-1-1 as background information for the support of RTT emergency services as a replacement for TTY device access to emergency services in the long term. This annex also discusses interactions between emergency services and other supplemental services.

C.1.1 Enhanced 9-1-1 (E9-1-1)

E9-1-1 Service is a public safety feature that allows emergency calls to be routed to a designated Emergency Service bureau when the 3-digit telephone number “9-1-1” is dialed. The Emergency Service bureau is a centralized

⁸ Available from the FCC at < <https://www.fcc.gov/general/9-1-1-master-psap-registry> >.

⁹ A primary PSAP is defined as a PSAP to which 9-1-1 calls are routed directly from the Emergency Services Network.

¹⁰ A secondary PSAP is defined as a PSAP to which 9-1-1 calls are transferred from a primary PSAP.

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agency or facility that a municipality operates to receive and respond to requests for emergency services such as police, fire, and Emergency Medical Services (EMS). Attendants at the Emergency Service bureau may be personnel from the police or fire department or any agency designated to receive emergency calls. In general, the centralized answering point for emergency calls is referred to as a PSAP.

Existing E9-1-1 Service architectures, and any evolution to the service architecture brought about by changes in technology and the regulatory environment, must support critical functionality to facilitate the delivery of emergency services to those in need. This functionality includes:

- The ability to detect an emergency call based on the dialed digits “911”.
- The selective routing of emergency calls via a specially-equipped Selective Router (SR)/E9-1-1 Tandem switch to an appropriate serving PSAP, based on the caller’s location (or approximate location). This includes the ability to default or alternate route emergency calls under certain failure scenarios.
- The delivery of a callback number to the PSAP with the call (allowing for some exceptions for certain wireless implementations and non-initialized handsets).
- The ability for the PSAP to use information received in call setup signaling to query an Automatic Location Identification (ALI) system to obtain location information for the caller.
- The ability for a PSAP call-taker to transfer an emergency call to another agency for further handling (e.g., for dispatch).

SRs are a critical element of legacy Emergency Services Networks that support E9-1-1 Service. SRs (also known as E9-1-1 Tandems) are specially-equipped central offices that provide the tandem switching of 9-1-1 calls. Selective routing is the process by which 9-1-1 calls are routed to the appropriate PSAP (or other designated destination) based on the caller’s location. For emergency calls that originate in legacy wireline networks, the caller’s location is represented by their 10-digit telephone number or Automatic Number Identification (ANI). For emergency calls that originate in legacy wireless networks, selective routing is done based on a 10-digit key that represents the caller’s location. In addition to providing selective routing functionality, SRs control the delivery of the voice call to the PSAP, as well as emergency call transfer and certain maintenance functions for each PSAP.

Emergency calls are typically delivered to SRs/E9-1-1 tandems in legacy Emergency Services Networks over dedicated Multi-Frequency (MF) or Signaling System No. 7 (SS7) trunk groups from wireline end offices and Mobile Switching Centers (MSCs). Selective Routing uses the calling number/ANI, or other location information that may be provided with the call to identify the target PSAP for the call. To support selective routing, an SR/E9-1-1 tandem will interact with a Selective Routing Database (SRDB). The SR provides the calling number/ANI or location key to the SRDB, and the SRDB returns an Emergency Service Number (ESN). An ESN is a three- to five-digit number representing a unique combination of emergency service agencies (law enforcement, fire, and EMS) designated to serve a specific range of addresses within a particular geographical area referred to as an Emergency Service Zone (ESZ). The SR uses the ESN to select the path to the destination PSAP for the emergency call. The ESN may also play a role in selecting the transfer-to PSAP if the primary PSAP requests selective transfer of the emergency call.

The SR delivers the emergency call to the PSAP, typically over traditional Centralized Automatic Message Accounting (CAMA)-like or Enhanced Multi-Frequency (E-MF) interfaces, with a key piece of information that will be used by the PSAP to query an ALI system for the caller’s location information. Having retrieved the location information, the PSAP can facilitate the dispatch of emergency personnel to the incident location. The introduction of the ANI feature was critical to supporting E9-1-1 because it allowed delivery of the 9-1-1 caller’s telephone number to the PSAP with the call. Using this information, the PSAP could identify the caller and, if necessary, call back the caller. A CAMA-like MF signaling scheme was initially used to support ANI delivery to the PSAP. This signaling scheme, referred to as Traditional MF, is still in use in certain areas today, and supports the delivery of a 7-digit number, along with a single Numbering Plan Digit (NPD) that can be used to derive the NPA. A Feature Group D-like signaling scheme, referred to as E-MF, is more commonly used between SRs and PSAPs, and supports the delivery of either one or two 10-digit numbers to the PSAP with the call, along with an ANI II value that tells the PSAP Customer Premises Equipment (CPE) whether to display the information using a steady or flashing display.

The delivery of the wireline 9-1-1 caller’s telephone number allows PSAPs to access the location information associated with the telephone number by querying the ALI database. In the case of wireline emergency callers, the ALI database contains static telephone number-to-street address mappings. The carrier that serves the PSAP typically operates the ALI databases.

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Enhanced wireless emergency services (i.e., wireless E9-1-1), requiring wireless carriers to provide the location of the wireless 9-1-1 callers to PSAPs, is mandated by the U.S. Federal Communications Commission (FCC). For the initial stage, Phase I, the FCC required Wireless Service Providers (WSPs) to upgrade their networks to support delivery of a callback number, and an identifier of the cell site or base station location where the 9-1-1 call originated, to the PSAP. Phase I of the FCC's mandate was required to be implemented by April 1998.

Phase II requires delivery of E9-1-1 service that includes the latitude and longitude of the 9-1-1 call within specific accuracy and reliability parameters, depending on the location technology that the carriers have chosen. For network-based technologies, location must be accurate to within 100 meters for 67 percent of calls, and 300 meters for 90 percent of calls. For handset-based technologies, location must be accurate to within 50 meters for 67 percent of calls, and 150 meters for 90 percent of calls. While the original order called for handset-based geographic location to be provided to the PSAPs by October 1, 2001, the FCC has since introduced a phased-in implementation schedule for handset- and network-based location that extends until January 2019.

The industry standard for E9-1-1 Phase I, J-STD-034 [Ref 8], uses the term Emergency Services Routing Digits (ESRD) for the cell site identifier that is associated with an emergency call. With E9-1-1 Phase I, the SR uses the ESRD to selectively route the wireless 9-1-1 call to the appropriate PSAP. Specifically, the SR queries the SRDB using the ESRD. As in the wireline architecture described above, the SR receives an ESN in the response from the SRDB and associates the ESN with the destination PSAP.

To fulfill Phase II requirements for the delivery of latitude and longitude associated with the 9-1-1 call, wireless carriers have had to deploy location determination technology in their networks. The location technology may be handset-based [e.g., Global Positioning System (GPS)] or network-based [e.g., Time Difference of Arrival (TDOA), Angle of Arrival (AOA), etc.]. Technically, if SS7 Integrated Services Digital Network (ISDN) User Part (ISUP) signaling is supported between the MSC and the SR/E9-1-1 tandem, Phase II location information could be delivered in the ISUP Initial Address Message (IAM), provided the wireless network is able to determine the location coordinates prior to call setup. However, due to limitations in today's location determination technology that result in delays in obtaining Phase II location, existing Phase II implementations only support the delivery of Phase I information or a location key in the call setup signaling. Phase II location information is delivered over a separate data link between the wireless network and the emergency services network. The E2 protocol defined in J-STD-036-C [Ref 9] and NENA-05-001 [Ref 10] is typically used over the data link between the wireless network and emergency services network to request/deliver initial caller location information to the PSAP via the ALI system, and to provide updated location information when requested. There are also some implementations that use the Mobile Location Protocol (MLP) between the ALI system and the Mobile Positioning Center (MPC)/Gateway Mobile Location Center (GMLC) in the wireless network to obtain the location associated with an emergency call.

The method in which Phase II location is delivered by the wireless network to the emergency services network via a separate data link is referred to as the Non-Call Associated Signaling (NCAS) approach in J-STD-036-C [Ref 9]. There are two variants of this approach: one which is just referred to as NCAS, and the other which is referred to as Wireline Compatibility Mode (WCM). Of the two variants, the WCM approach is more widely deployed.

With the WCM approach, all the FCC-mandated Phase I and Phase II location information and the callback number is sent over a separate data link to the ALI database. The MSC may support either an MF or an SS7 interface over which a single 10-digit number is delivered to the SR, and the SR supports an interface where only a single 7/10-digit number is delivered to the PSAP. The one piece of information sent by the MSC to the SR is referred to as the Emergency Services Routing Key (ESRK). The ESRK represents an ESZ in the jurisdiction of a PSAP, and also uniquely identifies the 9-1-1 call. The ESRK also uniquely identifies an MPC/GMLC in the wireless network. The MSC stores a set of ESRKs in its database and assigns them to wireless 9-1-1 calls. Upon completion of the 9-1-1 calls, the ESRKs are released to be assigned to other 9-1-1 calls. The PSAP receives only the ESRK from the SR and uses it to query the ALI database to receive the callback number, cell site identifier (ESRD), and the latitude and longitude information for the mobile caller.

With the wireless NCAS approach, the MSC sends a callback number and the cell site identifier (ESRD) to the SR using either MF or SS7 signaling. Call setup using the wireless NCAS approach is identical to the call setup in the E9-1-1 Phase I scenario documented in J-STD-034 [Ref 8]. The SR uses the ESRD to interact with the SRDB, and the SRDB returns an ESN that causes the SR to route the 9-1-1 call to the appropriate PSAP. The interface between the SR and the PSAP will typically be an E-MF interface that supports the delivery of both the callback number and the ESRD to the PSAP, although hybrid arrangements (involving the real-time update of the ALI data via the SR) may be used if the PSAP supports either a Traditional MF interface or the delivery of a single 10-digit number via an E-MF interface.

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Regulations implementing Title II of the American with Disabilities Act (ADA) [Ref 12] mandate direct access to emergency services for people who do not fully support voice communication because of a disability in hearing or speech and who use TTY technologies. Therefore, as specified in NENA 08-502 [Ref 11], every answering position is required to be equipped with TTY device capability, and when operating in TTY device mode, each position is required to retain all system features available from the existing 9-1-1 system. TTY technology provides a kind of limited real-time text feature with the goal of enabling conversations using text (communicated via Baudot tones) and voice. (See Annex B for further details related to TTY.)

C.1.2 Next Generation 9-1-1 (NG9-1-1)

The goal of NG9-1-1 is to provide at least E9-1-1-equivalent functionality in support of emergency call originations from fixed, nomadic, and mobile IP users, and to build on those capabilities to improve performance and extend feature functionality (e.g., to support delivery of text-based emergency services requests to PSAPs). There are a number of alternative NG9-1-1 Service architectures under discussion in various industry groups. NENA has defined a long-term solution for emergency calling, referred to as the i3 Solution, whose end state assumes end-to-end IP signaling from an IP-enabled endpoint to an IP-enabled PSAP, with callback and caller location information provided to the PSAP with the call. Similarly, ATIS is defining the architecture, protocol, and procedures to support the processing of emergency calls by an IMS-based NG Emergency Services Network. Regardless of the functional elements and interfaces that make up these architectures, NG9-1-1 Service architectures must, at a minimum, support the E9-1-1 capabilities identified in clause 6.1.2.

A fundamental capability required of any NG Emergency Services network is the ability to selectively route an emergency call to the appropriate PSAP based on the location from which the call was originated. This implies that information identifying the location of the caller must be available to any routing element in the call path. Emergency call setup in an NG9-1-1 environment is expected to be Session Initiation Protocol (SIP)-based. The SIP signaling associated with an emergency session request is expected to include location information, either “by value” [i.e., as a Presence Information Data Format Location Object (PIDF-LO)] in the body of the SIP message or “by reference” (where a location reference is included in the SIP signaling and can be de-referenced to obtain the location value/PIDF-LO). The routing element is expected to use a location value to query a call routing function to obtain routing information for the call. The location information used as input to the call routing function can either be in the form of a civic/street address or geo-coordinates. The output of the call routing function is expected to be in the form of a Uniform Resource Identifier (URI).

If location-based routing cannot be performed because the information received with the call is not sufficient to allow the location-based process to be successful (e.g., location information is not received with the call, or a route cannot be determined for the location value associated with the call), the NG Emergency Services Network must be able to route the call using a default location or default next hop URI (as appropriate for the abnormal condition encountered). Alternate/Overflow routing allows the NG Emergency Services Network to temporarily redirect emergency calls to/toward a pre-designated alternate PSAP/destination (e.g., a call center) when the primary PSAP or next hop element is not available to take calls (e.g., due to network/PSAP conditions or other policy).

When the NG Emergency Services Network delivers an emergency call to an NG PSAP, it is expected to generate SIP signaling that includes location information (by-value or by-reference), callback information, and Additional Data (by-value and/or by-reference). The location information that the NG Emergency Service Network signals to an NG PSAP will be the same as the location information that it received in incoming SIP signaling. For example, if a routing element within the NG Emergency Services Network receives location-by-reference in a SIP INVITE message associated with an incoming emergency call and it de-references that location reference to obtain a location-by-value with which to query a location-based routing functional element, it will still send the location-by-reference forward in outgoing SIP signaling to/toward the NG PSAP.

Likewise, routing elements in the NG Emergency Services Network may receive additional data associated with a call by reference and/or by value in an incoming SIP INVITE message associated with an emergency call. The routing element is expected to pass the additional data to/toward the NG PSAP in the same form as it was received. Today, PSAPs receive non-location information, such as class of service information and service provider contact information, associated with an emergency call, in the response from the ALI system. PSAPs that receive emergency calls from the NG Emergency Services Network must, at a minimum, have the same type of non-location information available to them as is available in ALI responses today.

NG9-1-1 anticipates that callers with hearing and/or speech disabilities will migrate from TTY devices to other forms of communication including RTT services. Although use of TTY devices is expected to decline, it cannot be assumed

that TTY devices will be completely gone by the time transition to NG9-1-1 is complete. Therefore, PSAPs that are served by NG Emergency Services Networks must be capable of receiving calls that are originated from TTY devices. Likewise, RTT service originations destined for legacy PSAPs that are served by NG Emergency Services Networks (i.e., via a Legacy PSAP Gateway) must be delivered as TTY calls.

Handling Baudot tones within an IP (VoIP) network can be very problematic, due to the sensitivity of Baudot to packet loss and other impairments. The NENA i3 Solution defined in NENA-STA-010.2 [Ref 13] recommends that transcoding from Baudot to RFC 4103 RTT [Ref 4] should occur as close to the TTY device as possible. The IMS-based NG9-1-1 Service Architecture being developed by ATIS supports the same philosophy. If an emergency call originates in a legacy originating network, the Legacy Network Gateway will be responsible for performing this transcoding.

Since it cannot be assured that all IP-based origination networks will perform the necessary transcoding, NG Emergency Services Networks will need to have the ability to transcode if an emergency call is presented to the NG Emergency Services Network containing Baudot tones. A transcoder must be placed as early in the media path as possible, and the IP network between the origination network and the transcoder must be engineered to have very low packet loss and minimize other impairments. The transcoder must be compliant with RFC 5369 [Ref 14] and must be transparent to audio that is not Baudot tones.

When used for communications to emergency services, it is important that RTT calls allow the simultaneous delivery of voice media and RTT media to the PSAP so that the PSAP can hear any background audio from the caller's location (while texting is going on) and use this valuable input in handling the emergency situation. Simultaneous delivery of voice media and RTT media will also be helpful in allowing people who can speak but not hear or hear clearly, or who cannot type well or at all, to speak to 9-1-1 but get text back from the PSAP. Mixing of voice media and RTT media may also be useful when the presence of excessive noise makes it hard to understand specific words or instructions. When Legacy Network Gateways convey TTY originations from legacy originating networks to NG Emergency Services Networks, the SIP call between the Legacy Network Gateway and the NG Emergency Services Network is required to include both voice media and RTT media from the point at which the presence of a TTY device is detected.

C.1.3 Interaction with Supplemental Services

People call 9-1-1 emergency services when they are in life-threatening or property-threatening situations and they need the immediate aid of emergency services. Consequently, communications with 9-1-1 emergency services are critical and distractions or interference from other services must be minimized. It is important that carriers consider interactions between originating and terminating features and 9-1-1 calls. In particular, 9-1-1 calls should not be interrupted by other originating features.

When 9-1-1 emergency services are requested from an IMS-based mobile device using emergency call dialed digits (e.g., 911, 112, 999), the mobile device is placed in emergency mode and restrictions are imposed for the duration of the emergency call. As described in 3GPP TS 22.173 [Ref 7], there are restrictions on supplemental services such as Communication Waiting (CW), Communication HOLD (HOLD), CONFERENCE (CONF), Explicit Communication Transfer (ECT), and Call DIVersion (CDIV) at the mobile device end of an emergency call and/or callback call. The temporary disabling of supplemental services is performed by the mobile device, by the CMSP network, or by a combination of both the mobile device and the CMSP network. The above restrictions apply equally to RTT supplementary services during an emergency call.

There are no such restrictions on the PSAP end of the emergency call. The PSAP can utilize conference calls for additional assistance such as language translation services. The PSAP can also transfer an emergency call to another PSAP or other destination (e.g., Poison Control).

C.2 RTT Emergency Services Overview

RTT service communications from an RTT service capable mobile device via an Emergency Services Network to a PSAP are viewed as an RTT emergency call. Consequently, the special rules, special processing, and temporary service overrides that occur on the mobile device end of the voice emergency call will also be applicable RTT emergency call and any associated RTT services which are equivalent to voice supplemental services.

RTT supplemental services (if any) that are equivalent to the voice supplemental services described in clause 6.1.2 would also need to be disabled for the mobile device end of the RTT emergency call for the duration of the RTT emergency call.

The processing of RTT emergency calls from mobile devices to 9-1-1 emergency services utilizes special signaling and dedicated resources. The networks and entities involved in these RTT emergency calls are shown in the following figure:

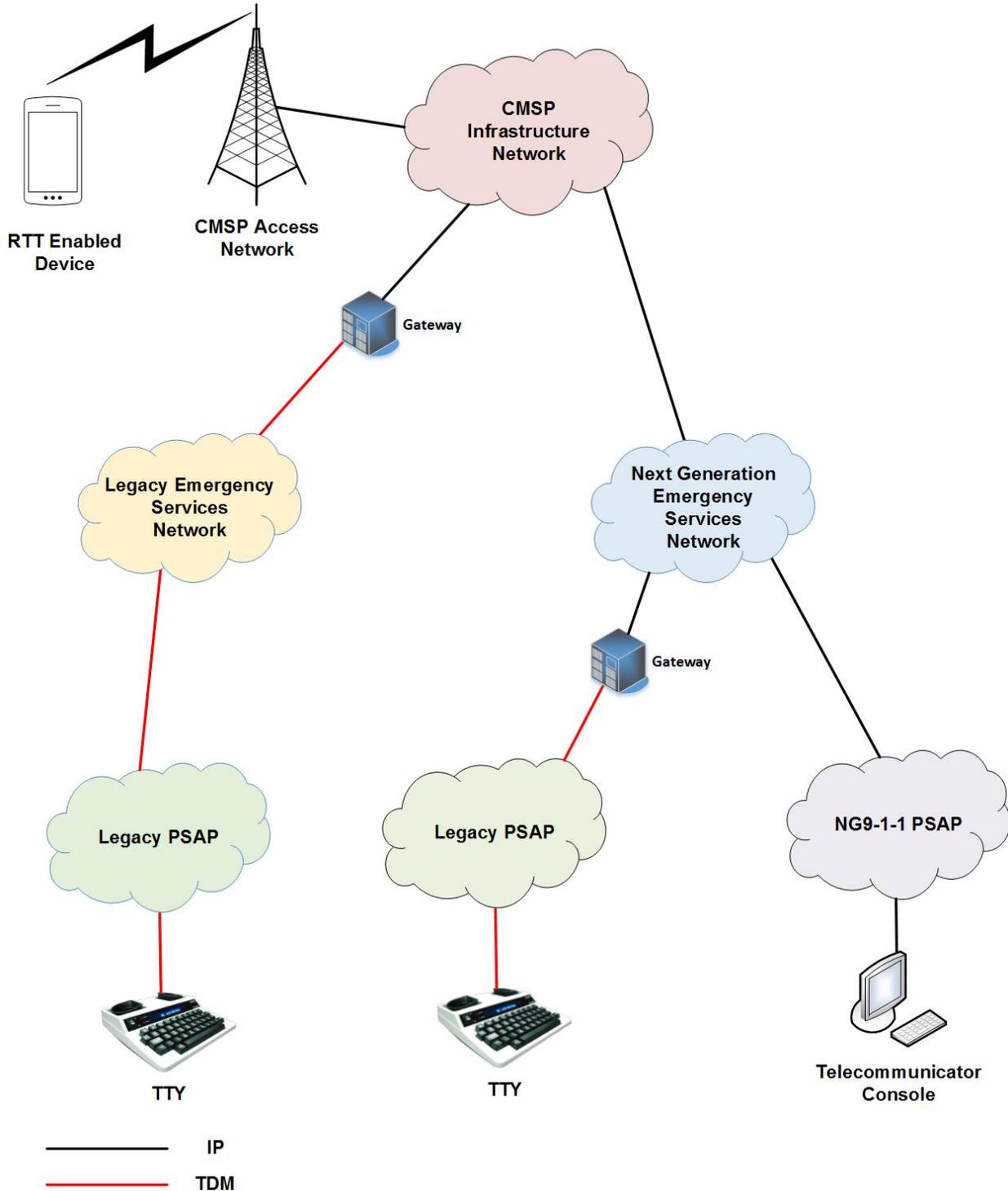


Figure C.1: RTT Emergency Services Network Diagram

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The following is a very high-level description¹¹ of how an emergency call is handled by these networks and entities:

1. The RTT emergency call is initiated with the same set of dialed digits that the mobile device would use to initiate a voice emergency call (e.g., 911, 112, 999).
2. The mobile device recognizes the dialed digits are an RTT emergency call request and sends signaling to the CMSP Access Network that an RTT emergency call is being established.
3. The CMSP Access Network establishes bearer channels between the mobile device and the CMSP Infrastructure Network and signals to the CMSP Infrastructure Network that this is an emergency call.
4. The CMSP Infrastructure Network delivers the incoming emergency call to entities in the CMSP Infrastructure Network which are dedicated to emergency call processing and these dedicated network entities will perform the following functions:
 - a. Identify the cell site serving the mobile device.
 - b. Initiate emergency service location determination procedures.
 - c. Select the appropriate Emergency Services Network based upon a routing location associated with the serving cell site. The emergency call may be directed to either a Legacy Emergency Services Network or a Next Generation Emergency Services Network.
 - d. Route the emergency call to the selected Emergency Services Network.
 - e. If the selected network is a Next Generation Emergency Services Network, the emergency call is delivered to the Next Generation Emergency Services Network as an RTT emergency call.
 - f. If the selected network is a Legacy Emergency Services Network, the emergency call is delivered to a legacy Selective Router via a gateway system where RTT-TTY interworking is performed.
 - g. Tag the associated Call Detail Records (CDRs) as an emergency call so that the back office CMSP systems will not charge the calling party's subscription for the RTT emergency calls.
5. When the Emergency Services Network receives the incoming RTT emergency call from the CMSP Infrastructure Network, the Emergency Services Network will perform the following functions:
 - a. Identification of the PSAP based on the routing location/key associated with the cell site serving the mobile device.
 - b. Determination of the callback number associated with the calling party's mobile device that can be used by the PSAP telecommunicator if the RTT emergency call is disconnected prematurely.
 - c. Determine if the identified PSAP is a legacy PSAP or an NG IP-enabled PSAP.
 - d. If the identified PSAP is a legacy PSAP, an RTT-TTY interworking function will be utilized to establish communications with the TTY device located at the PSAP.
 - e. If the identified PSAP is a next generation IP-enabled PSAP, the RTT emergency call is routed to the PSAP for presentation on a PSAP telecommunicator console.
 - f. The results of the enhanced mobile device location determination initiated in Step 4.b would either be delivered to the PSAP or will be stored for retrieval by the PSAP, depending on the capabilities of the PSAP.
6. Direct communications have now been established between the mobile device and the 9-1-1 emergency services.

¹¹ For additional detailed information on the processing of emergency calls, see NENA-STA-010.2 [Ref 13] and ATIS-0700015.v003 [Ref 15].