ATIS STANDARD

ATIS-0700029

ATIS Standard on -

Real Time Text Mobile Device Behavior

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Real Time Text Mobile Device Behavior

Alliance for Telecommunications Industry Solutions

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Abstract

This Standard specifies certain aspects of the mobile device behavior for handling Real Time Text (RTT) to facilitate communication between mobile devices (including emergency services) across multiple Commercial Mobile Service Providers (CMSPs).

Foreword

As a leading technology and solutions development organization, the Alliance for Telecommunications Industry Solutions (ATIS) brings together the top global information and communications technology (ICT) companies to advance the industry's most-pressing business priorities. ATIS serves the public through improved understanding between carriers, customers, and manufacturers.

This standard was developed jointly between ESIF, PTSC, and WTSC.

The Emergency Services Interconnection Forum (ESIF) provides a forum to facilitate the identification and resolution of technical and/or operational issues related to the interconnection of wireline, wireless, cable, satellites, Internet, and emergency services networks.

The Packet Technologies and Systems Committee (PTSC) develops and recommends standards and technical reports related to services, architectures, and signaling, in addition to related subjects under consideration in other North American and international standards bodies. PTSC coordinates and develops standards and technical reports relevant to telecommunications networks in the U.S., reviews and prepares contributions on such matters for submission to U.S. ITU-T and U.S. ITU-R Study Groups or other standards organizations, and reviews for acceptability or per contra the positions of other countries in related standards development and takes or recommends appropriate actions.

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.

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 initiation or issuance of the letter ballot for this document, the committees responsible for its development had the following leadership:

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The IP Multimedia Subsystem (IMS) Emergency Procedures for IMS Origination and Emergency Services IP Network (ESInet) (IMSESINET) joint project group was responsible for the development of this document.

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1 Scope, Purpose, & Application

1.1 Scope

The scope of this Standard is to specify a base level of Mobile Device Behavior for supporting Real Time Text (RTT) capabilities when accessing Voice over Long Term Evolution (VoLTE) networks[1]. The Standard includes the minimum behavior of the mobile device needed for performing RTT user-to-user communication within and between VoLTE Commercial Mobile Service Providers (CMSPs).

The scope of this Standard covers RTT implementations, as specified in IETF RFC 4103 [Ref 5] and as described in ATIS-1000068 [Ref 1] and ATIS-0700030 [Ref 6], for both emergency and non-emergency communications via CMSP networks.

The mobile devices that support real time text capabilities as defined in this standard are referred to as RTT-capable mobile devices.

1.2 Purpose

The purpose of this Standard is to specify certain aspects of mobile device behavior to facilitate RTT conversations between mobile devices across CMSPs.

1.3 Application

This Standard is applicable to CMSPs, CMSP network infrastructure vendors, and mobile device manufacturers. It may be applicable for RTT support by Public Safety Answering Points (PSAPs) and Telecommunications Relay Services (TRS) providers as well.

2 References

2.1 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] ATIS-1000068, *Technical Report on Support of TTY Service over IP using Global Text Telephony*; October, 2015.[2]

[Ref 2] GSMA IR.92, *GSM Association IMS Profile for Voice and SMS*; Version 9.0; 08 April 2015.[3]

[Ref 3] 3GPP TS 23.226, *Global Text Telephony (GTT); Stage 2*.[4]

[Ref 4] ITU-T T.140, *Protocol for Multimedia Application Text Conversion*, February 1998; including Addendum 1, *Marking of missing characters*, February 2000.[5]

[1] The support of RTT over Wi-Fi is for further study.

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

[3] This document is available from the GSM Association (GSMA) at: < http://www.gsma.com >.

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

[Ref 5] IETF RFC 4103, *RTP Payload for Text Conversion*; June 2005.[6]

[Ref 6] ATIS-0700030, *Real Time Text End-to-End Service Description Specification*; publication anticipated 2017.[2]

[Ref 7] 3GPP TS 26.114, *IP Multimedia Subsystem (IMS); Multimedia Telephony; Media handling and interaction*.[4]

[Ref 8] *The Unicode® Standard Version 8.0 – Core Specification*, The Unicode Consortium, June 17, 2015.[7]

[Ref 9] 3GPP TS 26.226, *Cellular text telephone modem; General description*.[4]

[Ref 10] 3GPP TS 22.173, *IP Multimedia Core Network Subsystem (IMS) Multimedia Telephony Service and supplementary services; Stage 1*.[4]

[Ref 11] 3GPP TS 24.229, *IP multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP); Stage 3*.[4]

[Ref 12] 3GPP TR 21.905, *Vocabulary for 3GPP Specifications.*[4]

[Ref 13] TIA 825-A, *A Frequency Shift Keyed Modem for Use on the Public Switched Telephone Network*.[8]

2.2 Informative References

[Ref 100] 3GPP TS 22.226, *Global Text Telephony (GTT); Stage 1*.[4]

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” means IETF RFC 4103-based implementation [Ref 5].

RTT Call: A call with RTT and voice components.

TTY: The term for the text telephones that are based on Baudot and TIA 825-A [Ref 13].

NOTE: TTY was originally an abbreviation for Teletype, a device that was later combined with a modem so that it could be used for text conversations over analog phone lines.

Voice over LTE: IP Multimedia Subsystem (IMS) Voice service over LTE.

3.2 Acronyms & Abbreviations

3GPP	Third Generation Partnership Project
ATIS	Alliance for Telecommunications Industry Solutions

[5] This document is available from the International Telecommunications Union Telecommunication Standardization Sector (ITU-T) at: < http://www.itu.int/en/ITU-T/publications/Pages/recs.aspx >.

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

[7] This document is available from the Unicode Consortium at: < http://unicode.org/versions/Unicode8.0.0/ >.

[8] This document is available from the Telecommunications Industry Association (TIA) at: < https://www.tiaonline.org/ >.

CMSP	Commercial Mobile Service Provider
CS	Circuit Switched
CTM	Cellular Text Telephone Modem
IETF	Internet Engineering Task Force
IMS	IP Multimedia Subsystem
IP	Internet Protocol
ITU-T	International Telecommunications Union Telecommunication Standardization Sector
LTE	Long Term Evolution
ME	Mobile Equipment
ms	millisecond
OEM	Original Equipment Manufacturer
PSAP	Public Safety Answering Point
PSTN	Public Switched Telephone Network
RTT	Real Time Text
TRS	Telecommunications Relay Services
TS	Technical Specification
TTY	Teletypewriter or Text Telephony
UE	User Equipment
UI	User Interface
VoLTE	Voice over LTE

4 RTT Background Information & End-to-End Service

Background information on RTT and the RTT end-to-end service description, including Use Cases and end-to-end service requirements, will be provided in the ATIS Real Time Text End-to-End Service Description Specification [Ref 6].

5 Assumptions

1. It is assumed that the user interface for the RTT capability will utilize accessibility settings and capabilities provided by the underlying mobile device platform.
2. It is assumed that the RTT-capable mobile device has implemented Voice over LTE (VoLTE) capabilities.
3. It is assumed that the RTT-capable mobile device may implement other 3GPP IMS non-voice multimedia capabilities as well.
4. The term “Mobile Device” only refers to the User Equipment (UE) component as defined in 3GPP TR 21.905 [Ref 12].

6 High Level Requirements

This section defines the high-level requirements for RTT that are applicable to the RTT-capable mobile device. These high level requirements were derived from [Ref 100]. These requirements are independent of the user's perception of the feature:

- **All Calls**
    1. The RTT-capable mobile device shall provide the capability to the user to have real-time text conversations in calls also capable of supporting voice communication.
- **Emergency Calls and Callbacks**
    2. The RTT-capable mobile device shall enable emergency RTT calls toward a PSAP and enable reception of RTT-enabled callbacks from a PSAP.

In addition to the high level RTT service requirements that apply to the RTT-capable mobile device, the following general RTT requirements also apply to the RTT-capable mobile device:

- **Session Control**
    3. User control of a combined RTT and voice session (call control) functions shall follow similar procedures as IMS voice-only sessions.
- **Invoking RTT Conversation**
    4. It shall be possible to initiate an RTT conversation at call establishment. It shall also be possible to add an RTT component to a voice call already in progress.
- **RTT Conversation Handling During Calls**
    5. Text transmission shall be done character by character as entered, or characters may be transmitted in small groups. No character should be buffered before transmission more than 300 milliseconds [Ref 4].
    6. The text transmission shall support maximum rates of at least 30 characters per second so that human typing speed as well as speech-to-text methods of generating RTT text can be supported.

7 RTT-Capable Mobile Device User Interface Requirements

This section defines the minimum set of requirements applicable to RTT-capable mobile device user interfaces. These requirements are grouped into the following topics:

- Integrated user interfaces.
- External interfaces.
- Text creation, transport, and presentation.
- Creating and editing input.
- Alternative presentation of audio and call progress information.

7.1 Integrated User Interfaces

Any text creation and text presentation capabilities that are built into the RTT-capable device shall be useable for RTT text input and presentation.

7.2 External User Interfaces

The RTT-capable mobile device shall support RTT communication input and presentation via the use of external digital text input and presentation devices connected through standardized interfaces (e.g., Bluetooth) that are integrated into the RTT-capable mobile device.

7.3 Text Creation, Transport, & Presentation

1. Text creation, transport, and presentation shall be per ITU-T T.140 [Ref 4] as specified in 3GPP TS 26.114 [Ref 7], clause 5.2.3.
2. The RTT function shall use the Unicode support [Ref 8] provided by the mobile device platform.

   NOTE: It is expected that the mobile device platform Unicode support [Ref 8] also supports the graphic rendition coding as specified in ITU-T T.140 [Ref 4].

7.4 Creating & Editing Input

The RTT-capable mobile device shall provide the capability to allow the user to enter text, backspace (i.e., erase last character), and to send a line separator as defined in the ITU-T T.140 [Ref 4].

NOTE 1: Per ITU-T T.140 [Ref 4], the backspace erasure has no specific limit other than the current RTT call. A user cannot modify the text input from another user, but they can backspace and modify their own text back beyond any “message-like” delimiters introduced by the user interface.

NOTE 2: The user interface could distinguish modifications from original text.

7.5 Presentation of Audio Alerting & Call Progress Tones

1. The RTT-capable mobile device shall provide visual and tactile indication of network call progress tones and alerting signals (e.g., ringing, answer, and busy conditions). This may include text presentation.
2. The RTT-capable mobile device shall provide visual and tactile indication for received audio signal activity including but not limited to indications that help the user identify when the other party is speaking. This may include such features as a volume level status bar on the graphical display.

8 Emergency Services Requirements

This section provides the RTT-capable mobile device behavior requirements for use of emergency services. The end-to-end description and requirements for RTT emergency services are provided in the RTT End-to-End Service Description specification [Ref 6]. The network behavior for RTT emergency services is defined in ATIS-1000068 [Ref 1].

1. The user option to not transmit immediately that is described in section 9.7 should be disabled in RTT-capable mobile devices for emergency calls.
2. The dialed digits used by an RTT-capable mobile device to initiate an RTT emergency call shall be the same as the dialed digits used to initiate a voice emergency call from the mobile device (e.g., 911, 112, 999).
3. 3GPP supplemental services defined in 3GPP TS 22.173 [Ref 10], which are disabled for voice emergency calls, shall also be disabled for RTT emergency calls.
4. The location of the RTT-capable mobile device shall be determined by the same emergency call location determination methodologies used for a voice emergency call.
5. An RTT-capable mobile device shall support emergency callback with RTT to the same extent that it supports emergency callback for calls that are voice-only.

9 Mobile Device Requirements

This section defines the mobile device requirements to support RTT functionality. These mobile device requirements are grouped into the following topics:

- Device support for TTY or RTT.
- Simultaneous voice and RTT.
- Transport support in the device.
- Establishing an RTT media component during call setup.
- Addition and removal of RTT media component mid-call.
- Transmission of characters for RTT.
- User control of voice media usage during an RTT call.

9.1 Device Support for TTY or RTT

1. RTT-capable mobile devices shall implement RTT as defined by 3GPP TS 23.226 “Global Text Telephony” clause 5.1 [Ref 3], ATIS-1000068 “Support of TTY Service over IP using Global Text Telephony” [Ref 1], the text-related sections of 3GPP TS 26.114 [Ref 7], and the requirements in this Standard.
2. RTT-capable mobile devices that are also able to operate in the Circuit Switched (CS) domain may support 3GPP TS 26.226, Cellular Text Telephone Modem (CTM) [Ref 9], with an internal user interface or an externally-attached TTY when operating in the CS domain.
3. The RTT-capable mobile device shall not require an external TTY device for using RTT communication.
4. The RTT-capable mobile device shall not be required to support a locally-connected external TTY for RTT operation.

9.2 Simultaneous Voice & RTT

The RTT-capable mobile device shall support voice and RTT operation in both directions of a conversation with all four media streams occurring simultaneously.

9.3 Transport Support in the Device

The RTT-capable mobile device shall support RTT transport per RFC 4103 [Ref 5] and per ATIS-1000068 [Ref 1].

9.4 Establishing an RTT Media Component During Call Origination

When enabled by the settings specified in section 10, the RTT-capable mobile device shall provide means to the user to include RTT in an outgoing call attempt. The means of invoking RTT at call origination shall be presented along with the other user call functions such as placing a voice or video call.

Independent of the settings in section 10, devices may provide functionality to automatically include RTT in outgoing calls based on preferences for individual contact list entries.

9.5 Establishing an RTT Media Component for Incoming Calls

RTT requested by the calling party in an incoming call shall be accepted by the RTT-capable mobile device.

When enabled by the settings specified in section 10, for an incoming voice-only call the user interface shall provide a choice to answer the call as voice-only or to answer the call and add RTT. When answering using a connected accessory or control device that only provides a single function for answering calls, the mobile device should answer the call as voice-only.

NOTE: If the user chooses to answer with RTT added, the device may need to automatically perform a mid-call addition of RTT (as defined in section 9.6).

9.6 Addition & Removal of RTT Media Component Mid-Call

The RTT-capable mobile device shall provide to the user the means to request addition of an RTT component at any time during an established voice-only call. The RTT-capable mobile device should also provide the ability to remove an RTT media component from an established RTT call. RTT management during a call shall be presented along with the other functions available during calls (e.g., speaker or mute).

Any request for RTT addition/removal received by the device shall be accepted without action by the user.

9.7 Transmission of Characters for RTT

1. The RTT-capable mobile device should support text transmission in 300 ms intervals as recommended in clause 9.4 of 3GPP TS 26.114 [Ref 7].
2. An RTT-capable mobile device may provide to the user the option to compose and edit text for review prior to submission for transmission in an RTT session.

9.8 User Control of Voice Media Usage during an RTT Call

All voice media control functions (e.g., mute, volume, etc.) shall continue to function during an RTT call.

9.9 Media Feature Tag Support

An RTT-capable mobile device shall include the “text” media feature tag in the Contact header field as specified in 3GPP TS 24.229 [Ref 11] regarding the use of media feature tags for streaming media types.

10 User Preference Settings

This section defines the minimum RTT user options an RTT-capable mobile device shall provide to the user.

The requirements in this section are only applicable to RTT-capable mobile devices that provide users the ability to customize VoLTE settings.

1. The RTT-capable mobile device shall allow a user to set their preference for the visibility of RTT call controls. This shall provide at least two options, referred to here as *Always Visible* and *Visible During Calls*.
    a. When the user has selected the *Always Visible* option, the RTT-capable mobile device shall make available to the user RTT call origination, answer, and upgrade capabilities as defined in section 9 of this standard.
    b. When the user has selected the *Visible During Calls* option, the RTT-capable mobile device shall make available to the user RTT call upgrade capabilities as defined in section 9.6 and shall hide from the user the RTT call origination and answer capabilities defined in sections 9.4 and 9.5.
    c. The default selection for an RTT-capable mobile device should be *Visible During Calls*.
    d. The definition of any additional options and the impacts of this selection to any other mobile device-provided functions are outside the scope of this specification.
2. The User Interface (UI) for RTT user preference selection should be integrated into other related user preference selections that the RTT-capable mobile device may provide.

    NOTE: This document has no requirements on where the specified settings are stored.

Annex A
(informative)

Annex A: Example RTT Call Setup with Voice

This informative annex provides examples on RTT call setup with voice. The examples provided in this informative annex are illustrative only based upon an assumed mobile device user interface implementation and should not be interpreted as either the only possible mobile device user interface implementation or as a preferred mobile device user interface implementation.

The sequences below illustrate simple scenarios where an end user places or receives a call that initiates an RTT conversation also including voice capabilities. Mobile device Original Equipment Manufacturers (OEMs) would be expected to adapt the support of RTT to their specific mobile device operating system environments. Device vendor creativity and product differentiation is encouraged with a basic "common look and feel" of the RTT service.

The example scenarios vary depending on the position of the "Real Time Text Call Controls" setting. The *Always Visible* value presents call controls for RTT both before and during calls, and is intended for users depending on RTT, and for users who often use RTT. This setting has value for these users for conveniently requesting to set up RTT calls and is essential in calls to 9-1-1 for RTT users. The *Visible During Calls* value is intended for users who are seldom or never using RTT. They are provided with the capability to add RTT to calls on request by either party but with call control elements for that purpose only provided after completed call establishment.

A.1 Examples of Calls with Real Time Text Call Controls Setting to "Visible During Calls"

Example Scenario 1: Calling with RTT Call Controls setting to *Visible During Calls*

When the RTT Real Time Text Call Controls setting is *Visible During Calls*, the initial call screen does not show any RTT related button.

Figure A.1 – Example Call Screen with Real Time Text Call Controls Set to Visible During Calls

The user initiates the call in voice mode, and the screen for call management during the call appears.

Example Scenario 2: RTT Addition during a Call

After a voice call has been established, the user selects an option to initiate RTT by pressing “add text”. The following figure provides one possible illustration of the display of an active call with an “add text” button. When addition of the RTT media component is completed, a text screen is presented to be used for the RTT conversation. This action works equally in both the *Visible During Calls* and *Always Visible* position of the Real Time Text Call Controls setting.

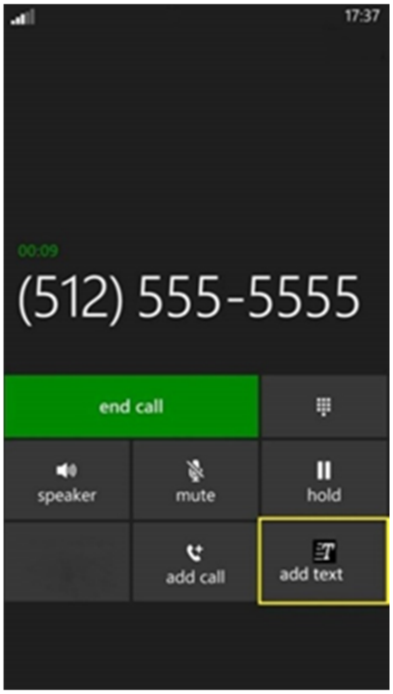

Figure A.2 – Example Active Call Display with Add Text Button

NOTE 1: The term “add text” is used as a simple description of what the user can expect from this function.

NOTE 2: The icon used on the “add text” button in the above figure is from RealTimeText.org.[9]

Example Scenario 3: RTT Conversation

Once the user selects the “add text” option, the addition of the RTT media component is requested. When successfully added, a Real Time Text screen opens. The text screen has the following three primary components:

a. The log window where both sides of the RTT conversation are shown together and received characters are updated in real time.
b. A second text box where the user’s text is currently being typed.
c. The keyboard (if not using an external keyboard).

[9] Available at: < http://realtimetext.org/standard-graphics >

This function is available in both *Visible During Calls* and *Always Visible* position of the Real Time Text Call Controls setting. The following figure shows one possible implementation of the text screen with an example RTT conversation in progress:

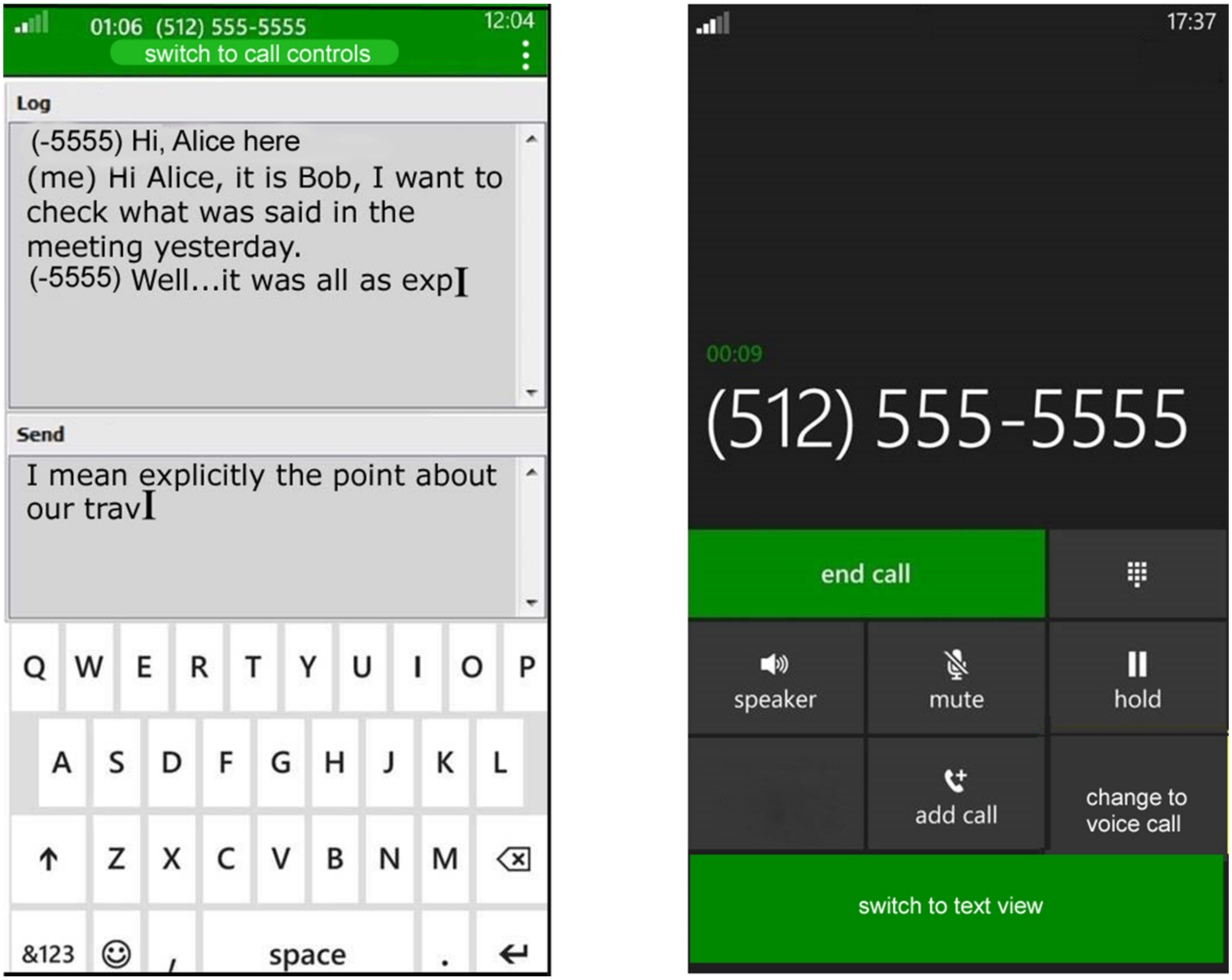

Figure A.3 – Example RTT Screen with Active RTT Conversation and Call Control Screen Available during RTT Call

NOTE 1: Other layouts for the real-time text window are possible. For example, the text window could be in a column mode where the text is divided in one column per participant in the call and text placed in an approximate vertical position corresponding to when the text was sent or received.

NOTE 2: For those mobile devices that implement the option to occasionally hold text for review and edit, a control to hold text for review and editing before transmission would also be visible as well as a "send" button for edit before transmit mode. This mode may be provided as an addition to the true real-time text mode.

NOTE 3: Voice communication is available simultaneously with the RTT text conversation (except if blocked by network functions such as TTY interworking functions when text communication is going on).

NOTE 4: The user will have the option to switch back and forth to a call control screen during the call to handle mid-call options. In the example, this function is achieved through the button in the upper edge of the screen. The call control screen example is shown to the right of the text screen.

Example Scenario 4: Incoming Voice Call with Real Time Text Call Controls Set to *Visible During Calls*

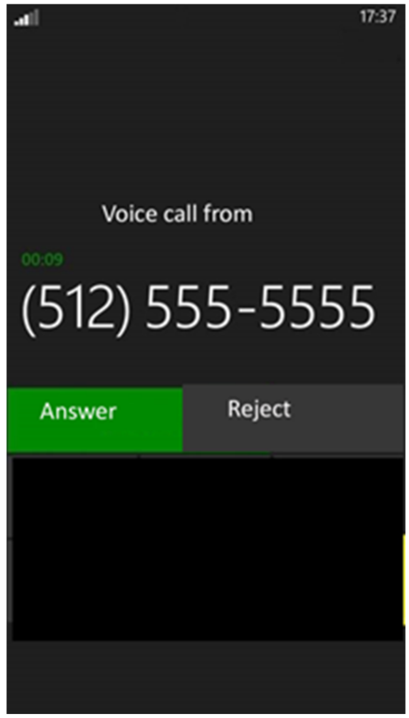

Figure A.4 – Example screen during incoming voice call with Real Time Text Call Controls set to Visible During Calls

When an incoming voice call is received by the RTT-capable mobile device with Real Time Text Call Controls in *Visible During Calls* setting, the voice media included is presented on the screen, and the answer and reject options.

When "Answer" is pressed, a voice call will be established and the mid-call screen with an "add text" button will appear to be used for the case when the user wants to add text in the same way as in scenarios 2 and 3 above.

This will also be the case for incoming calls from TTYs in the Public Switched Telephone Network (PSTN) when interworking with RTT is provided by the network. These calls will look like voice calls, and the receiving user will need to either recognize the number or a TTY sound or any other indication that the call is from a TTY user and then request "add text" to enable communication in RTT mode with the TTY.

Example Scenario 5: Incoming text+voice call with Real Time Text Call Controls Set to *Visible During Calls*

A call with RTT and voice requested is received in an RTT-capable mobile device with Real Time Text Call Controls set to *Visible During Calls*. The incoming call screen shows the media offered.

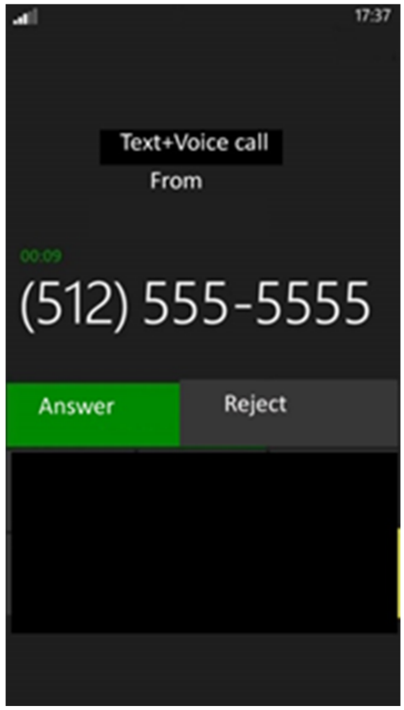

Figure A.5 – Example screen during incoming text+voice call with Real Time Text Call Controls set to Visible During Calls

In this case it is sufficient for the user to press "Answer" to establish an RTT call, because the RTT-capable mobile device will accept the offered media. The RTT screen will open and text and audio can be exchanged as in example scenario 3 above.

The user will have the option to switch to a call management screen for call options available during the call.

A.2 Call Examples with Real Time Text Call Controls Setting to "Always Visible"

Example Scenario 6: Call with Real Time Text Call Controls Set to *Always Visible*

The user selects the phone dialer, enters a phone number, and selects “Text+Voice call”. The following figure provides one potential illustration of the mobile device phone dialer display after the phone number has been entered.

Figure A.6 – Example Device Phone Dialer Display with Digits to Call Entered with Real Time Text Call Controls set to Always Visible

Once the user selects the “Text+Voice call” option and the call is answered including the RTT media component and voice, a Real Time Text window opens and the text part of the call can be performed there as shown in example scenario 3 above.

Example Scenario 7: Incoming call with Real Time Text Call Controls Set to *Always Visible*

A call with voice media is received in an RTT-capable mobile device with Real Time Text Call Controls set to *Always Visible*.

The incoming call screen shows the voice media offered.

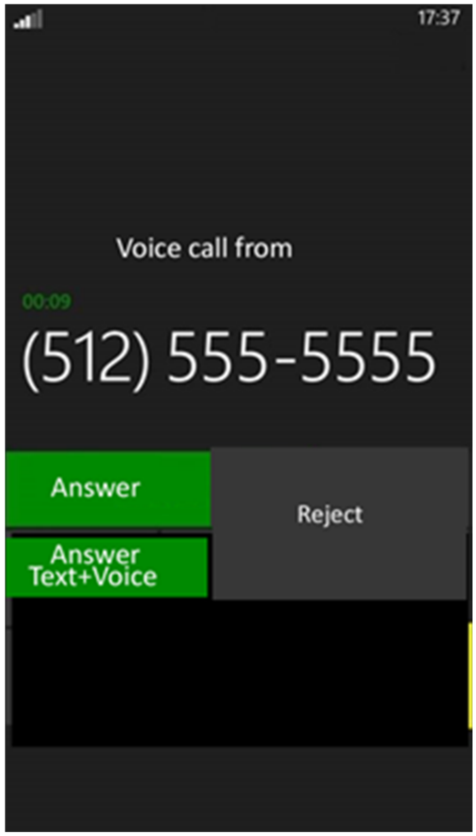

Figure A.7 – Example screen during incoming voice call with Real Time Text Call Controls set to Always Visible

In this example scenario the user needs RTT, but the caller was not aware of that and called in voice mode. The user presses "Answer Text+Voice" to establish an RTT call. The RTT-capable mobile device will then request to add RTT media to the voice call. When that is successful, the text screen will appear as in example scenario 3 above.

This will also be the case for incoming calls from TTYs in the PSTN when interworking with RTT is provided by the network. These calls will look like voice calls, and the receiving user will need to either recognize the number or a TTY sound or any other indication that the call is from a TTY user and then request "Answer Text+Voice" to enable communication in RTT mode with the TTY.

Example Scenario 8: Incoming text+voice call with Real Time Text Call Controls Set to *Always Visible*

A call with RTT and voice requested is received in an RTT-capable mobile device with Real Time Text Call Controls set to *Always Visible*.

The incoming call screen shows the media offered.

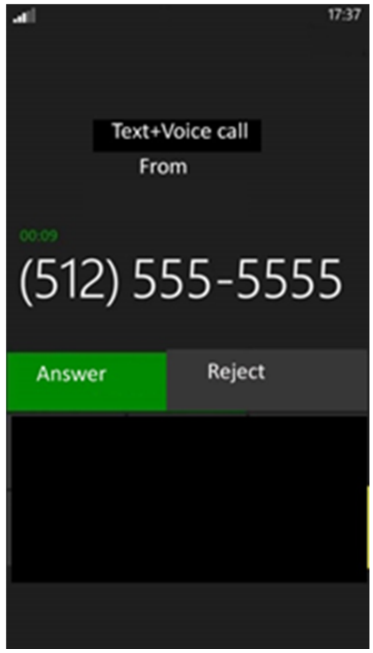

Figure A.8 – Example screen during incoming text+voice call with Real Time Text Call Controls set to Always Visible

In this case the user presses "Answer" to establish an RTT call. The RTT-capable mobile device will accept the offered media. The RTT screen will open and text and audio can be exchanged as in example scenario 3 above.

The user will have the option to switch back and forth to a call management screen during the call for mid-call options.

Example Scenario 9: Preference for RTT Stored in Contact List and used for Calling with RTT

In this example, the RTT-capable mobile device has a contact list implementation that can store preferences for different call types. Calling with RTT is one of these supported call types. When looking up a contact, the call type marked as preferred is presented so that the preference is made clear.

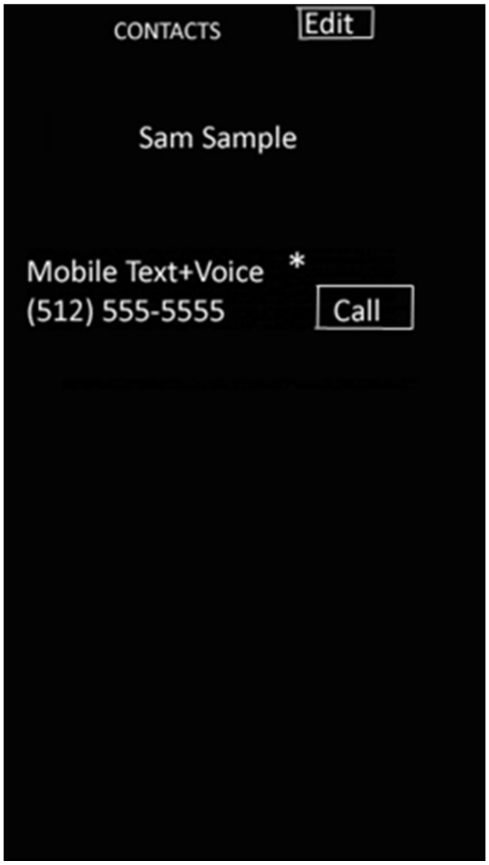

Figure A.9 – Contact list entry with preference for RTT marked

When making the call, it is easy for the user to just click on the preferred call type to make the call in the suitable mode, in this example being an RTT call with text and voice.