ATIS STANDARD

ATIS-1000068

Support of TTY Service Over IP Using Global Text Telephony

TECHNICAL REPORT

As a leading technology and solutions development organization, the Alliance for Telecommunications Industry Solutions (ATIS) brings together the top global ICT companies to advance the industry's most pressing business priorities. ATIS' nearly 200 member companies are currently working to address the All-IP transition, 5G, network functions virtualization, big data analytics, cloud services, device solutions, emergency services, M2M, cyber security, network evolution, quality of service, billing support, operations, and much more. These priorities follow a fast-track development lifecycle — from design and innovation through standards, specifications, requirements, business use cases, software toolkits, open source solutions, and interoperability testing.

ATIS is accredited by the American National Standards Institute (ANSI). The organization is the North American Organizational Partner for the 3rd Generation Partnership Project (3GPP), a founding Partner of the oneM2M global initiative, a member of the International Telecommunication Union (ITU), as well as a member of the Inter-American Telecommunication Commission (CITEL). For more information, visit www.atis.org.

Notice of Disclaimer & Limitation of Liability

The information provided in this document is directed solely to professionals who have the appropriate degree of experience to understand and interpret its contents in accordance with generally accepted engineering or other professional standards and applicable regulations. No recommendation as to products or vendors is made or should be implied.

NO REPRESENTATION OR WARRANTY IS MADE THAT THE INFORMATION IS TECHNICALLY ACCURATE OR SUFFICIENT OR CONFORMS TO ANY STATUTE, GOVERNMENTAL RULE OR REGULATION, AND FURTHER, NO REPRESENTATION OR WARRANTY IS MADE OFMERCHANTABILITY OR FITNESS FOR ANY PARTICULAR PURPOSE OR AGAINST INFRINGEMENT OF INTELLECTUAL PROPERTY RIGHTS. ATIS SHALL NOT BE LIABLE, BEYOND THE AMOUNT OF ANY SUM RECEIVED IN PAYMENT BY ATIS FOR THIS DOCUMENT, AND IN NO EVENT SHALL ATIS BE LIABLE FOR LOST PROFITS OR OTHER INCIDENTAL OR CONSEQUENTIAL DAMAGES. ATIS EXPRESSLY ADVISES THAT ANY AND ALL USE OF OR RELIANCE UPON THE INFORMATION PROVIDED IN THIS DOCUMENT IS AT THE RISK OF THE USER.

> NOTE - The user's attention is called to the possibility that compliance with this standard may require use of an invention covered by patent rights. By publication of this standard, no position is taken with respect to whether use of an invention covered by patent rights will be required, and if any such use is required no position is taken regarding the validity of this claim or any patent rights in connection therewith. Please refer to [http://www.atis.org/legal/patentinfo.asp] to determine if any statement has been filed by a patent holder indicating a willingness to grant a license either without compensation or on reasonable and non-discriminatory terms and conditions to applicants desiring to obtain a license.

ATIS-1000068, *Support of TTY Service Over IP Using Global Text Telephony*

Is an ATIS Standard developed by the **ATIS Packet Technologies and Systems Committee (PTSC)**.

Published by

Alliance for Telecommunications Industry Solutions
1200 G Street, NW, Suite 500
Washington, DC 20005

ATIS-1000068

Technical Report on

Support of TTY Service Over IP Using Global Text Telephony

Alliance for Telecommunications Industry Solutions

Approved August 3, 2017

Abstract

This Technical Report (TR) describes the means by which the Teletypewriter (TTY) service can be provided over Internet Protocol (IP) between operators' networks through the use of the Global Text Telephony (GTT) capability which enables simultaneous audio and/or video with text media stream.

Foreword

The Alliance for Telecommunication Industry Solutions (ATIS) serves the public through improved understanding between providers, customers, and manufacturers. 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 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, PTSC, 1200 G Street NW, Suite 500, Washington, DC 20005.

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

M. Dolly, PTSC Chair (AT&T)

V. Shaikh, PTSC Vice Chair (Vencore Labs)

Table of Contents

Table of Figures

ATIS Technical Report on –

Support of TTY Service Over IP Using Global Text Telephony

1 Introduction

This Technical Report (TR) describes the means by which the Teletypewriter (TTY) service can be provided over IP between operators' networks through the use of the Global Text Telephony (GTT) capability which enables simultaneous audio and/or video with text media stream.

TTY service allows real time conversation in text between two persons having a Baudot-capable device. This service is supported through the Circuit Switched (CS) public network. Although new Internet technologies have reduced the need for this service, it still plays an important role, especially for emergency 9-1-1 calls.

Real-Time Text (RTT) is a term used to define the ability to instantly communicate text as it is typed, as opposed to after a sentence or thought is completed, in the manner of instant messaging. This term has now been replaced with Fast Text. RTT can now be signalled over IP networks. RTT is combined with voice and can be optionally combined with video. When this combined service is provided by an IP Multimedia Subsystem (IMS) network, it is referred to as GTT. GTT is supported in IMS networks via RTT capability using IETF SIP/SDP for the negotiation of the text media stream and IETF RFC 4103 [RFC 4103] RTP-text for transport with text coded according to ITU-T Recommendation T.140 [T.140].

Figure 1.1 shows a generalized view of the GTT feature architecture. It combines different networks and network types and integrates text conversation systems already existing within these networks.

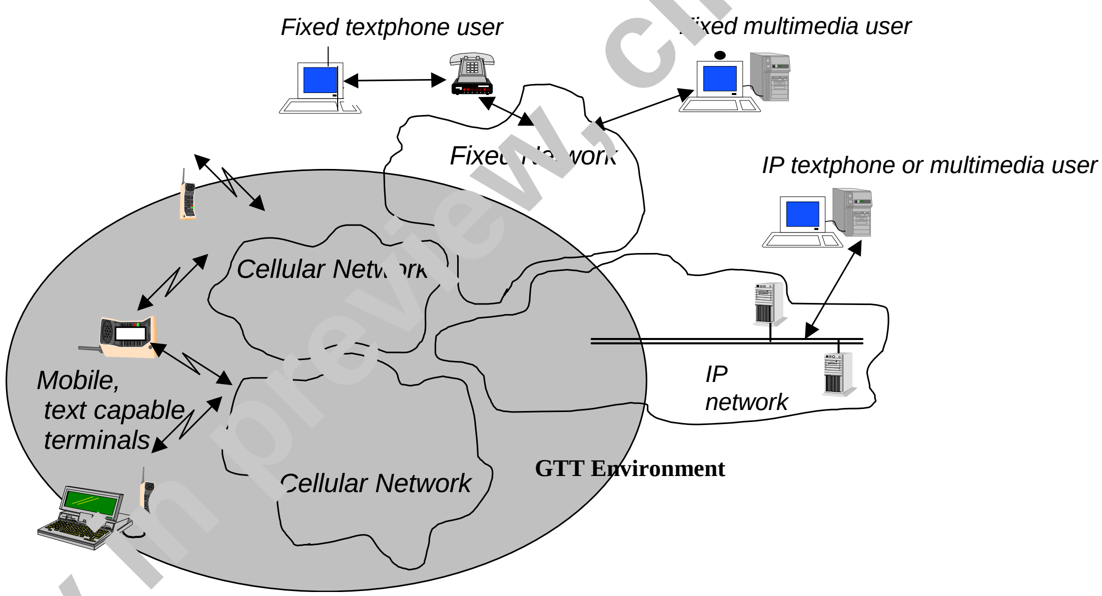

Figure 1.1 – Generalized GTT Architecture

Throughout the remainder of this document the following terms are used as described below:

- TTY – this term is used in reference to a real-time text service where the transport of text for this service is technology dependent: i.e., Baudot for circuit and RFC 4103 [RFC 4103] text for packet.
- GTT – this term is used to refer to the ability to support audio and/or video streams simultaneously with text (RTT) in an IP network.

- RTT – this term is used to refer to the text stream in an IP network as encoded per RFC 4103 [RFC 4103].

2 Abbreviations

ADA	Americans with Disabilities Act
ALG	Application Level Gateway
AS	Application Server
ATIS	Alliance for Telecommunications Industry Solutions
AVP	Audio Visual Profile
AVPF	Audio Visual Profile with Feedback
BGCF	Border Gateway Control Function
CS	Circuit Switched
CTM	Cellular Text Telephone Modem
E-CSCF	Emergency Call Session Control Function
EDT	European Deaf Telephone
IBCF	Interconnection Border Control Function
IETF	Internet Engineering Task Force
ITU-T	International Telecommunication Union - Telephony
GTT	Global Text Telephony
IM-MGW	Internet Protocol-Multimedia Media Gateway
IMS	IP Multimedia Subsystem
IP	Internet Protocol
IWF	Interworking Function
LNG	Legacy Network Gateway
LPG	Legacy PSAP Gateway
LRF	Location Retrieval Function
LS	Location Server
MGCF	Media Gateway Control Function
MGW	Media Gateway
MRFC	Media Resource Function Controller
MRFP	Media Resource Function Processor
P-CSCF	Proxy Call Session Control Function
PCM	Pulse Code Modulation
PCMU	Pulse Code Modulation mu-law
POTS	Plain Old Telephone Service
PSAP	Public Safety Answering Point
PSTN	Public Switched Telephone Network
RDF	Routing Determination Function
RTT	Real-Time Text

S-CSCF	Serving Call Session Control Function
SDP	Session Description Protocol
SIP	Session Initiation Protocol
TDM	Time Division Multiplexed
TN	Telephone Number
TTY	Teletypewriter
UDP	User Datagram Protocol
VoIP	Voice over Internet Protocol
VoLTE	Voice over Long Term Evolution

3 Support of TTY Using Global Text Telephony (GTT)

3.1 Overview

The support of the TTY service is a regulatory requirement for North American public networks. As these networks migrate to packet technologies, the TTY service must also be migrated to IP. GTT offers real time conversation in text, optionally combined with voice and/or video. GTT is mainly used for distant conversation with hearing or speech impaired users.

On the PSTN, different systems for text telephony exist and are used in different regions, e.g., Baudot (in US), or EDT, V.21, Bell103, Minitel, and V.18 in other countries. They all use different modem technologies within PCM and different character coding for the transmission of text. They are described in the annexes of ITU-T Recommendation V.18 [V.18]. Baudot is a protocol used to signal a limited set of uni-case letters and figures represented by five bit codes at a rate of 45 baud in the U.S. No error correction is provided. Any party of a GTT call may at any time initiate text or send voice. Speech and text may be used in an alternating manner during a conversation on the PSTN. It is also possible that speech is transferred in one direction and text in the opposite direction. However, speech and text cannot be used in the same direction at the same time.

In the 3G radio interface, a dedicated CTM modem is used (see 3GPP TS 26.226 [TS 26.226]), which is terminated within the CS domain and interworked to PSTN using an in-band text telephony format.

A generic Interworking Function (IWF) can be introduced into the call whenever a conversion is needed between different text encoding, e.g., between RTT and Baudot. Interworking between RTT over RTP from IP networks with CS network text telephony is provided by the MGCF and IM-MGW or IMS triggering the insertion of an interworking (conversion) function. The interworking capabilities in the MGW support the detection of Baudot on the CS side and the conversion between Baudot and RTT over RTP.

Interworking between RTT over RTP and Baudot may also be necessary with other packet networks or other components within the same IP network not supporting RTT. In this case, the IWF may be placed closer to the access interface point. This same approach may be used as an alternate solution for interworking with CS networks thus avoiding the introduction of RTT interworking.

The procedures to detect and convert text/modem involve valuable MGW resources. The GTT interworking procedures specified in this document allow more efficient use of MGW resources by focusing the application of text/modem conversion on the minority of calls that use text telephony.

It is assumed that SIP terminals supporting text media will not automatically offer text media, but that this will be instead governed by SIP terminal configuration options and user interactions to suit the communication preferences and abilities of the user. However, a SIP terminal desiring to set up a GTT call will offer RTT media ***(along with voice media),*** possibly with video media. The interworking function then provides the conversion between RTT over RTP and text/modem signals. Conversely, if the SIP terminal does not request RTT support, no Interworking function is necessary. An IMS Multimedia SIP terminal configured to use RTT Telephony but receiving an SDP offer for voice-only media will accept this offer and then send its own subsequent SDP offer adding text media. When receiving such a subsequent offer for text media, the MGW will provide the conversion between RTT over RTP and text/modem signals at the CS interface. If the terminating mobile device does not offer RTT, no interworking function is necessary.