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Railway Telecommunications (RT);
Global System for Mobile communications (GSM);
Usage of Session Initiation Protocol (SIP)
on the Network Switching Subsystem (NSS)
to Fixed Terminal Subsystem (FTS)
interface for GSM Operation on Railways

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Contents

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Foreword

This Technical Specification (TS) has been produced by ETSI Technical Committee Railway Telecommunications (RT).

Modal verbs terminology

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Introduction

While a number of interoperability specifications for various interfaces at various layers of GSM-R systems exist, the interface between the Network Switching Subsystem (NSS) and the Fixed Terminal Subsystem (FTS) has not yet been addressed by any interoperability specification activity.

In most of the GSM-R system deployments available at the time of the creation of the present document, the Network Switching Subsystem and the Fixed Terminal Subsystem are interconnected using TDM based interfaces such as DSS1 [i.2].

ETSI TS 102 610 [9] specifies the usage and format of UUIE for call-related end-to-end functionality in GSM-R systems but no other interworking topics.

The present document addresses the interoperability specification gap between the Network Switching Subsystem and the Fixed Terminal Subsystem with an interface based on the Internet Protocol (IP) [2], the Session Initiation Protocol (SIP) [3], the Session Description Protocol (SDP) [6] and the Real-Time Transport Protocol (RTP) [7].

In addition to the table of contents, the following explanation will help you navigate through and understand the contents of the present document:

- Clauses 1 to 3 are predefined by ETSI.
- Clause 4 shows and explains the reference system architecture and identifies the interface(s) for the present document.
- Clause 5 holds the functional requirements for the interface subject to the present document.
- Clause 6 specifies in detail the signalling interface for all supported functions and services.
- Clause 7 specifies in detail the media interface.
- Clause 8 specifies the additions and changes for a voice recorder interface.

- Annex A explains the mechanism to locate SIP entities at the present interface.
- Annex B contains recommendations on the use and implementation of standardized Quality of Service mechanisms at the present interface.
- Annex C contains recommendations about the security mechanisms.
- Annex D contains a mapping table of EIRENE [1] to interface features.
- Annex E contains a description of group call control scenarios.

1 Scope

The present document defines the signalling and media interface between the Network Switching Subsystem and the Fixed Terminal Subsystem in order to provide a clear set of services needed for GSM-R operations. This includes voice call service and available call-related supplementary services. In addition, requirements for specific implementation of the signalling and media interface within either the Network Switching Subsystem or the Fixed Terminal Subsystem are stated where applicable. The present document addresses the Internet Layer and upwards of the Internet Protocol Suite IETF RFC 1122 [i.18] on the signalling and media interface.

Any service other than voice call service and call-related supplementary services (such as data services, Short Message Service, etc.) is out of scope of the present document; additional features may be addressed in future releases.

The present document does not specify any other interface between the Network Switching Subsystem and the Fixed Terminal Subsystem nor does it cover any internal interfaces of either NSS or FTS. Such interfaces may be addressed in a future release of the present document.

The present document does not address any specific 3GPP Release or Architecture.

2 References

2.1 Normative references

References are either specific (identified by date of publication and/or edition number or version number) or non-specific. For specific references, only the cited version applies. For non-specific references, the latest version of the referenced document (including any amendments) applies.

Referenced documents which are not found to be publicly available in the expected location might be found at http://docbox.etsi.org/Reference.

NOTE: While any hyperlinks included in this clause were valid at the time of publication, ETSI cannot guarantee their long term validity.

The following referenced documents are necessary for the application of the present document.

[1] UIC P001D010 (Version 15.1): "UIC Project EIRENE System Requirements Specification".

NOTE: Available at http://www.uic.org/IMG/pdf/eirene_srs_15.1.pdf.

[2] IETF RFC 791 (1981): "Internet Protocol".

[3] IETF RFC 3261 (2002): "SIP: Session Initiation Protocol".

[4] IETF RFC 3264 (2002): "An Offer/Answer Model Session Description Protocol (SDP)".

[5] IETF RFC 4733 (2006): "RTP Payload for DTMF Digits, Telephony Tones, and Telephony Signals".

[6] IETF RFC 4566 (2006): "SDP: Session Description Protocol".

[7] IETF RFC 3550 (2003): "RTP: A Transport Protocol for Real-Time Applications".

[8] IETF RFC 3326 (2002): "The Reason Header Field for the Session Initiation Protocol (SIP)".

[9] ETSI TS 102 610 (V1.1.0): "Railways Telecommunications (RT); Global System for Mobile communications (GSM); Usage of the User to User Information Element for GSM Operation on Railways".

[10] IETF RFC 5234 (2008): "Augmented BNF for Syntax Specifications: ABNF".

[11] IETF RFC 3262 (2002): "Reliability of Provisional Responses in Session Initiation Protocol (SIP)".

[12] IETF RFC 4412 (2006): "Communications Resource Priority for the Session Initiation Protocol (SIP)".

[13] IETF RFC 3325 (2002): "Private Extensions to the Session Initiation Protocol (SIP) for Asserted Identity within Trusted Networks".

[14] IETF RFC 5876 (2010): "Updates to Asserted Identity in the Session Initiation Protocol (SIP)".

[15] IETF RFC 3323 (2002): "A Privacy Mechanism for the Session Initiation Protocol (SIP)".

[16] IETF RFC 4028 (2005): "Session Timers in the Session Initiation Protocol (SIP)".

[17] IETF RFC 3311 (2002): "The Session Initiation Protocol (SIP) UPDATE Method".

[18] Void.

[19] Void.

[20] Void.

[21] Void.

[22] Recommendation ITU-T Q.850 (1998): "Usage of cause and location in the Digital Subscriber Signalling System No. 1 and the Signalling System No. 7 ISDN user part".

[23] Recommendation ITU-T E.164 (2010): "The international public telecommunication numbering plan".

[24] Recommendation ITU-T Q.955.3 (1993): "Stage 3 description for community of interest supplementary services using DSS 1: Multi-level precedence and pre-emption (MLPP)".

[25] IETF RFC 3986 (2005): "Uniform Resource Identifier (URI): Generic Syntax".

[26] IETF RFC 768 (1980): "User Datagram Protocol".

[27] Recommendation ITU-T G.711 (1988): "Pulse code modulation (PCM) of voice frequencies".

[28] IETF RFC 2833 (2000): "RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals".

[29] Void.

[30] IETF RFC 3840 (2004): "Indicating User Agent Capabilities in the Session Initiation Protocol (SIP)".

[31] IETF RFC 4574 (2006): "The Session Description Protocol (SDP) Label Attribute".

[32] Recommendation ITU-T I.255.3 (1990): "Multi-Level Precedence and Pre-emption service".

[33] IETF RFC 4579 (2006): "Session Initiation Protocol (SIP) Call Control - Conferencing for User Agents".

[34] IETF RFC 3891 (2004): "The Session Initiation Protocol (SIP) "Replaces" Header".

[35] IETF RFC 7462 (2015): "URNs for the Alert-Info Header Field of the Session Initiation Protocol (SIP)".

[36] IETF RFC 4244 (2005): "An Extension to the Session Initiation Protocol (SIP) for Request History Information".

[37] IETF RFC 4458 (2006): "Session Initiation Protocol (SIP) URIs for Applications such as Voicemail and Interactive Voice Response (IVR)".

[38] ETSI TS 129 163 (V12.13.0): "Digital cellular telecommunications system (Phase 2+) (GSM); Universal Mobile Telecommunications System (UMTS); LTE; Interworking between the IP Multimedia (IM) Core Network (CN) subsystem and Circuit Switched (CS) networks (3GPP TS 29.163 version 12.13.0 Release 12)".

2.2 Informative references

References are either specific (identified by date of publication and/or edition number or version number) or non-specific. For specific references, only the cited version applies. For non-specific references, the latest version of the referenced document (including any amendments) applies.

NOTE: While any hyperlinks included in this clause were valid at the time of publication, ETSI cannot guarantee their long term validity.

The following referenced documents are not necessary for the application of the present document but they assist the user with regard to a particular subject area.

[i.1] IETF RFC 7433 (2015): "A Mechanism for Transporting User to User Call Control Information in SIP".

[i.2] ETSI ETS 300 403-1 (V1.3.2): "Integrated Services Digital Network (ISDN); Digital Subscriber Signalling System No. one (DSS1) protocol; Signalling network layer for circuit-mode basic call control; Part 1: Protocol specification [ITU-T Recommendation Q.931 (1993), modified]".

[i.3] IETF RFC 6086 (2011): "Session Initiation Protocol (SIP) INFO Method and Package Framework".

[i.4] IETF RFC 3428 (2002): "Session Initiation Protocol (SIP) Extension for Instant Messaging".

[i.5] IETF RFC 3515 (2001): "The Session Initiation Protocol (SIP) Refer Method".

[i.6] IETF RFC 3265 (2002): "Session Initiation Protocol (SIP)-Specific Event Notification".

[i.7] IETF RFC 3903 (2004): "Session Initiation Protocol (SIP) Extension for Event State Publication".

[i.8] Void.

[i.9] IETF RFC 3665 (2003): "Session Initiation Protocol (SIP) Basic Call Flow Examples".

[i.10] IETF RFC 3960 (2004): "Early Media and Ringing Tone Generation in the Session Initiation Protocol (SIP)".

[i.11] ETSI EN 300 925 (V7.0.2): "Digital cellular telecommunications system (Phase 2+) (GSM); Voice Group Call Service (VGCS) - Stage 1 (GSM 02.68 version 7.0.2 Release 1998)".

[i.12] ETSI EN 300 926 (V8.0.1): "Digital cellular telecommunications system (Phase 2+) (GSM); Voice Broadcast Service (VBS) - Stage 1 (GSM 02.69 version 8.0.1 Release 1999)".

[i.13] IETF RFC 3263 (2002): "Session Initiation Protocol (SIP): Locating SIP Servers".

[i.14] IETF RFC 1035 (1987): "Domain names - implementation and specification".

[i.15] IETF RFC 2181 (1997): "Clarifications to the DNS Specification".

[i.16] IETF RFC 2663 (1999): "IP Network Address Translator (NAT) Terminology and Considerations".

[i.17] Void.

[i.18] IETF RFC 1122 (1989): "Requirements for Internet Hosts -- Communication Layers".

[i.19] IETF RFC 3551: "RTP Profile for Audio and Video Conferences with Minimal Control".

[i.20] IETF draft RFC draft-siprec-protocol-16: "Session Recording Protocol".

[i.21] IETF RFC 5009 (2007): "Private Header (P-Header) Extension to the Session Initiation Protocol (SIP) for Authorization of Early Media".

[i.22] IETF RFC 2474 (1998): "Definitions of the Differentiated Services Field (DS Field) in the IPv4 and IPv6 Headers".

[i.23] IETF RFC 2475 (1998): "An Architecture for Differentiated Services".

[i.24] IETF RFC 4594 (2006): "Configuration Guidelines for DiffServ Service Classes".

[i.25] IETF RFC 5865 (2010): "A Differentiated Services Code Point (DSCP) for Capacity-Admitted Traffic".

3 Definitions and abbreviations

3.1 Definitions

For the purposes of the present document, the terms and definitions given in IETF RFC 3261 [3] and the following apply:

call: SIP Dialog between two Signalling Endpoints

NOTE: Established for the purpose of a voice communication and related data exchange.

client: any network element that sends SIP requests and receives SIP responses

NOTE: Clients may or may not interact directly with a human user. User agent clients and proxies are clients.

Communication Session (CS): session that is the subject of recording

dialog: peer-to-peer SIP relationship between two UAs that persists for some time

NOTE: A dialog is established by SIP messages, such as a 2xx response to an INVITE request. A dialog is identified by a call identifier, local tag, and a remote tag.

final response: response that terminates a SIP transaction, as opposed to a provisional response that does not

NOTE: All 2xx, 3xx, 4xx, 5xx and 6xx responses are final.

Fixed Terminal Subsystem (FTS): part of the EIRENE [1] system that provides access to this network (and services) via controller equipment (in general referred to as Fixed Terminals)

Fully Qualified Domain Name (FQDN): domain name that includes all higher level domains relevant to the entity named

header: component of a SIP message that conveys information about the message

header field: component of the SIP message header

NOTE: A header field can appear as one or more header field rows. Header field rows consist of a header field name and zero or more header field values. Multiple header field values on a given header field row are separated by commas. Some header fields can only have a single header field value, and as a result, always appear as a single header field row.

initiator, calling party, caller: party initiating a session (and dialog) with an INVITE request

NOTE: A caller retains this role from the time it sends the initial INVITE that established a dialog until the termination of that dialog.

invitee, invited user, called party, callee: party that receives an INVITE request for the purpose of establishing a new session

NOTE: A callee retains this role from the time it receives the INVITE until the termination of the dialog established by that INVITE.

Media Endpoint, RTP Endpoint: entity that terminates RTP stream(s) under the control of a single SIP Endpoint in the same subsystem

NOTE: This entity may be physically separated from the SIP Endpoint.

method: primary function that a request is meant to invoke on a server

NOTE: The method is carried in the request message itself. Example methods are INVITE and BYE.

Network Switching Subsystem (NSS): part of the PLMN infrastructure that performs all necessary functions in order to handle the call services to and from the mobile stations as well as to and from fixed terminals

operational priority: different call types have call priorities during railway communications

NOTE 1: This is the definition given in EIRENE SRS [1].

NOTE 2: This behaviour is mentioned as operational priority of a call.

option tag: unique identifiers used to designate new options (extensions) in SIP

NOTE: The composition of option tags is defined in IETF RFC 3261 [3].

provisional response: response used by the server to indicate progress, but that does not terminate a SIP transaction

NOTE: 1xx responses are provisional, other responses are considered final.

proxy, proxy server: intermediary entity that acts as both a server and a client for the purpose of making requests on behalf of other clients

NOTE: A proxy server primarily plays the role of routing, which means its job is to ensure that a request is sent to another entity "closer" to the targeted user. Proxies are also useful for enforcing policy (for example, making sure a user is allowed to make a call). A proxy interprets, and, if necessary, rewrites specific parts of a request message before forwarding it.

Recording Session (RS): SIP session created between SRC and SRS for the purpose of recording a Communication Session

request: message sent from a client to a server, for the purpose of invoking a particular operation

response: message sent from a server to a client, for indicating the status of a request sent from the client to the server

server: element that receives requests in order to service them and sends back responses to those request

NOTE: Examples of servers are proxies, user agent servers, redirect servers, and registrars.

session: set of multimedia senders and receivers and the data streams flowing from senders to receivers

NOTE: A callee can be invited several times, by different calls, to the same session. If SDP is used, a session is defined by the concatenation of the SDP user name, session id, network type, address type, and address elements in the origin field.

Signalling Endpoint, SIP Endpoint: entity that acts as a SIP User Agent

NOTE: Within the scope of the present document this term refers to NSS and FTS.

Signalling Proxy, SIP Proxy: logical entity to route requests to the user's current location, authenticate and authorize users for services, implement provider call-routing policies, and provide features to users

(SIP) transaction: single request and any responses to that request, which include zero or more provisional responses and one or more final responses

NOTE: In the case of a transaction where the request was an INVITE (known as an INVITE transaction), the transaction also includes the ACK only if the final response was not a 2xx response. If the response was a 2xx, the ACK is not considered part of the transaction.

tag: parameter which is used in the To and From header fields of SIP messages to identify a dialog

NOTE: The composition of tags is defined in IETF RFC 3261 [3].

User Agent (UA): Internet endpoint

User Agent Client (UAC): logical entity that creates a new request, and then uses the client transaction state machinery to send it

NOTE: The role of UAC lasts only for the duration of that transaction. In other words, if a piece of software initiates a request, it acts as a UAC for the duration of that transaction. If it receives a request later, it assumes the role of a user agent server for the processing of that transaction.

User Agent Server (UAS): logical entity that generates a response to a SIP request

NOTE: The response accepts, rejects, or redirects the request. This role lasts only for the duration of that transaction. In other words, if a piece of software responds to a request, it acts as a UAS for the duration of that transaction. If it generates a request later, it assumes the role of a user agent client for the processing of that transaction.

3.2 Abbreviations

For the purposes of the present document, the following abbreviations apply:

ACK	ACKnowledgement
AF	Assured Forwarding
AoCC	Advice of Charge (Charging)
AoCI	Advice of Charge (Information)
B2BUA	Back to Back User Agent
BAIC	Barring of All Incoming Calls
BAOC	Barring of All Outgoing Calls
BIC-Roam	Barring of Incoming Calls when Roaming Outside the Home PLMN Country
BNF	Backus Naur Form
BOIC	Barring of Outgoing International Calls
BOIC-exHC	BOIC except those to Home PLMN Country
CCBS	Completion of Calls to Busy Subscribers
CFB	Call Forwarding on Mobile Subscriber Busy
CFNRc	Call Forwarding on Mobile Subscriber Not Reachable
CFNRy	Call Forwarding on No Reply
CFU	Call Forwarding Unconditional
CLIP	Calling Line Identification Presentation
CLIR	Calling Line Identification Restriction
CN	Core Node
CoLP	Connected Line Identification Presentation
CoLR	Connected Line Identification Restriction
CS	Communication Session
CSRC	Contributing SouRCe
CUG	Closed User Group
CW	Call Waiting
DL	Down Link
DNS	Domain Name Service
DSCP	Differentiated Service Code Point
DTMF	Dual Tone Multi Frequency
ECT	Explicit Call Transfer
EF	Expedited Forwarding
EIRENE	European Integrated Railway Radio Enhanced Network
eMLPP	enhanced Multi-Level Precedence and Pre-emption
FQDN	Fully Qualified Domain Name
FTP	File Transfer Protocol
FTS	Fixed Terminal Subsystem
GSM-R	Global System Mobile-Railways
HOLD	Call hold
HTTP	Hyper Text Transfer Protocol
IETF	Internet Engineering Task Force
IN	Intelligent Network
INF	INForm
INV	INVite
IP	Internet Protocol

ITU-T	International Telecommunication Union - Telecommunication standardization sector
MLPP	Multi-Level Precedence and Pre-emption
MO/PP	Mobile Originated/Point-to-Point
MPTY	Multi Party Service
MT/PP	Mobile Terminated/Point-to-Point
NAPT	Network Address Port Translation
NAT	Network Address Translation
NSS	Network Switching Subsystem
OK	OKay
OPT	OPTion
OSI	Open Systems Interconnection
PABX	Private Access Branch eXchange
PCM	Pulse Code Modulation
PCMA	Pulse Code Modulation - A law
PCM-A	Pulse Code Modulation - A law
PCMU	Pulse Code Modulation - u-law
PHB	Per Hop Behaviour
PLMN	Public Land Mobile Network
PRA	PRovisional Acknowledgment
PRACK	Provisional Response Acknowledgement
PSTN	Public Switched Telephone Network
QoS	Quality of Service
RFC	Request For Comments
RS	Recording Session
RTP	Real-Time Transport Protocol
SDP	Session Description Protocol
SE	Session Expires
SIP	Session Initiation Protocol
SRC	Session Recording Client
SRS	Session Recording Server
SRTP	Secured Real-time Protocol
SSRC	Synchronization SouRCe
TDM	Time Division Multiplexing
ToS	Type of Service
UA	User Agent
UAC	User Agent Client
UAS	User Agent Server
UDP	User Datagram Protocol
UIC	Union Internationale des Chemins de Fer, International Union of Railways
UPD	UPDate
URI	Uniform Resource Identifier
URL	Uniform Resource Locator
URN	Uniform Resource Name
USSD	Unstructured Supplementary Service Data
UUI	User-to-User Information
UUIE	User to User Information Element
UUS1	User-to-User Signalling 1
VBS	Voice Broadcast Service
VGCS	Voice Group Call Service

4 Reference System Architecture

The system architecture used to identify the interface that is the subject of the present document is a simplification of a GSM-R system down to a minimum of logical entities relevant to the present document.