ETSI TS 103 389 V1.2.1 (2013-09)

Technical Specification

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

Reference
RTS/RT-00021

Keywords
FTS, SIP, GSM-R, NSS, railways

ETSI

650 Route des Lucioles
F-06921 Sophia Antipolis Cedex - FRANCE

Tel.: +33 4 92 94 42 00 Fax: +33 4 93 65 47 16

Siret N° 348 623 562 00017 - NAF 742 C
Association à but non lucratif enregistrée à la
Sous-Préfecture de Grasse (06) N° 7803/88

Important notice

Individual copies of the present document can be downloaded from:
http://www.etsi.org

The present document may be made available in more than one electronic version or in print. In any case of existing or perceived difference in contents between such versions, the reference version is the Portable Document Format (PDF). In case of dispute, the reference shall be the printing on ETSI printers of the PDF version kept on a specific network drive within ETSI Secretariat.

Users of the present document should be aware that the document may be subject to revision or change of status. Information on the current status of this and other ETSI documents is available at
http://portal.etsi.org/tb/status/status.asp

If you find errors in the present document, please send your comment to one of the following services:
http://portal.etsi.org/chaircor/ETSI_support.asp

Contents

Intellectual Property Rights 5
Foreword 5
Introduction 5
1 Scope 6
2 References 6
2.1 Normative references 6
2.2 Informative references 7
3 Definitions and abbreviations 8
3.1 Definitions 8
3.2 Abbreviations 9
4 Reference System Architecture 10
5 Interface Functionality 11
5.1 Basic Call 11
5.1.1 Progress Indication 11
5.1.2 Early Media 12
5.2 Connected Parties Identity Information 12
5.3 Call Hold 12
5.4 Multi Level Precedence and Pre-emption 12
5.5 Voice Group Call and Broadcast Call Control 12
5.6 User-User-Information-Element Transport 12
5.7 Reason Transport 13
6 Signalling Interface 13
6.1 Network Layer Protocol 13
6.2 Transport Layer Protocol 13
6.3 Signalling Protocol 13
6.3.1 SIP Entities 13
6.3.1.1 SIP User Agent 14
6.3.1.2 SIP Proxy 14
6.3.2 SIP Request Methods 14
6.3.3 SIP Responses 15
6.3.4 SIP Header Fields 15
6.3.5 SIP Bodies 17
6.3.6 SIP URI Convention 18
6.3.6.1 Display Name 19
6.3.6.2 User Part 19
6.3.6.3 Host Part 19
6.3.6.4 URI Parameters 19
6.3.6.5 Use 19
6.3.6.6 Examples 20
6.3.7 Option Tags 21
6.4 Interface Functionality to Signalling Interface Mapping 22
6.4.1 Basic Call 22
6.4.2 Connected Parties Identity Information 23
6.4.3 Media Session Renegotiation and Call Hold 25
6.4.4 Early Media 27
6.4.5 Multi Level Precedence and Pre-emption 29
6.4.5.1 Resource Priority 30
6.4.5.2 Reason Indication for Precedence and Pre-emption Events 30
6.4.5.3 Signalling Procedure for Precedence Call Blocking 30
6.4.5.4 Signalling Procedure for Pre-emption 31
6.4.6 Group Call and Broadcast Call Control 32
6.4.7 User-to-User-Information-Element Transport 32

6.4.8 Release Cause Transport 33
6.4.9 SIP Session Timer 33
6.4.10 OPTIONS Processing 35
6.4.10.1 OPTIONS Heartbeating 36
6.4.11 Signalling for Group Call and Broadcast Call Control 36

7 Media Interface 38
7.1 Network Layer Protocol 38
7.2 Transport Layer Protocol 38
7.3 Real-Time Transport Protocol 38
7.3.1 Media inactivity detection 38
7.4 Media Codecs 39
7.4.1 DTMF 39
7.4.1.1 Limitations to RFC 4733 39

Annex A (normative): Locating SIP Entities 40

Annex B (informative): Quality of Service Framework 43

Annex C (informative): Security Framework 44

Annex D (informative): Mapping of EIRENE to Interface Features 45

Annex E (informative): Group Call Control Scenarios 47

Annex F (informative): Bibliography 49

History 50

Intellectual Property Rights

IPRs essential or potentially essential to the present document may have been declared to ETSI. The information pertaining to these essential IPRs, if any, is publicly available for **ETSI members and non-members**, and can be found in ETSI SR 000 314: *"Intellectual Property Rights (IPRs); Essential, or potentially Essential, IPRs notified to ETSI in respect of ETSI standards"*, which is available from the ETSI Secretariat. Latest updates are available on the ETSI Web server (http://ipr.etsi.org).

Pursuant to the ETSI IPR Policy, no investigation, including IPR searches, has been carried out by ETSI. No guarantee can be given as to the existence of other IPRs not referenced in ETSI SR 000 314 (or the updates on the ETSI Web server) which are, or may be, or may become, essential to the present document.

Foreword

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

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].

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.
- 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.

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. The present document addresses the Internet Layer and upwards of the Internet Protocol Suite [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. Voice recording and related interfaces are out of scope of the present document. 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

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 reference 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.

2.1 Normative references

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

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

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

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

[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 preemption (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 (1998): "Pulse code modulation (PCM) of voice frequencies".

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

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

2.2 Informative references

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 draft RFC draft-johnston-cuss-sip-uui-01: "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] IETF RFC 1594 (1994): "FYI on Questions and Answers to Commonly asked "New Internet User" Questions".

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

[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".

3 Definitions and abbreviations

3.1 Definitions

For the purposes of the present document, the following terms and definitions apply:

call: refers to a SIP Dialog (RFC 3261 [3]) between two Signalling Endpoints, established for the purpose of a voice communication and related data exchange

Client: As defined in RFC 3261 [3].

Dialog: As defined in RFC 3261 [3].

Final response: As defined in RFC 3261 [3].

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: As defined in RFC 1594 [i.8].

Header: As defined in RFC 3261 [3].

Header field: As defined in RFC 3261 [3].

Initiator, Calling Party, Caller: As defined in RFC 3261 [3].

Invitee, Invited User, Called Party, Callee: As defined in RFC 3261 [3].

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: As defined in RFC 3261 [3].

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: as defined in EIRENE SRS [1] different call types have call priorities during railway communications. This behaviour is mentioned as operational priority of a call

Option-tag: As defined in RFC 3261 [3].

Provisional response: As defined in RFC 3261 [3].

Proxy, proxy server: As defined in RFC 3261 [3].

Request: As defined in RFC 3261 [3].

Response: As defined in RFC 3261 [3].

Server: As defined in RFC 3261 [3].

Session: As defined in RFC 3261 [3].

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: Proxy Server as defined by RFC 3261 [3]

(SIP) transaction: As defined in RFC 3261 [3].

Tag: As defined in RFC 3261 [3].

User Agent Client: As defined in RFC 3261 [3].

User Agent Server: As defined in RFC 3261 [3].

User Agent: As defined in RFC 3261 [3].

3.2 Abbreviations

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

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
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
CoLP	Connected Line Identification Presentation
CoLR	Connected Line Identification Restriction
CUG	Closed User Group
CW	Call waiting
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
FTS	Fixed Terminal Subsystem
GSM-R	Global System Mobile-Railways
HOLD	Call hold
IP	Internet Protocol
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
PABX	Private Access Branch eXchange
PHB	Per Hop Behaviour
PLMN	Public Land Mobile Network
PRACK	Provisional Response Acknowledgement
PSTN	Public Switched Telephone Network
QoS	Quality of Service
RFC	Request For Comments
RTP	Real-Time Transport Protocol
SDP	Session Description Protocol
SIP	Session Initiation Protocol
SRTP	Secured Real-time Protocol
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
URI	Uniform Resource Identifier
USSD	Unstructured Supplementary Service Data
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.

Within the context of the present document a GSM-R system is logically divided into a GSM-R Network and a Fixed Terminal Subsystem. The interface between the Mobile Terminals and the NSS as well as the interface between the Fixed Terminals and the FTS are explicitly not addressed in the present document. The focus of the present document is solely:

- the Signalling Interface; and
- the Media Interface;

between the logical subsystem NSS and the logical subsystem FTS.

It is important to note that this architecture does not necessarily reflect any physical entities in a GSM-R system.

Figure 4.1 illustrates the reference system architecture.

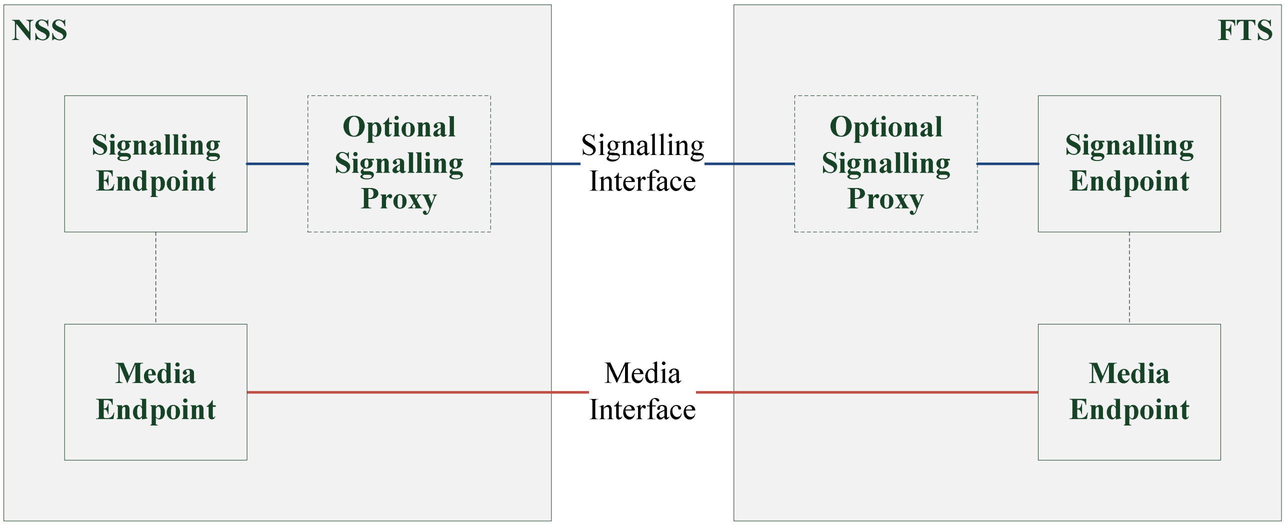

Figure 4.1: Reference System Architecture

Depending on the deployment scenario and the NSS/FTS design there may be one or more Signalling Endpoints, one or more Media Endpoints and zero or more Signalling Proxies on either side of the interface.

The Media Endpoint(s) are controlled by (a) Signalling Endpoint(s) in the same subsystem. This control mechanism is out of scope of the present document.

One Signalling Endpoint may establish more than one call. Also one Signalling Proxy and one Media Endpoint may be involved in one or more calls.

The maximum number of Signalling Endpoints allowed to be involved in a single call on the present interface is two-one on each side.

Optionally deployed Signalling Proxies may be involved in the signalling flow for either incoming traffic or outgoing traffic or both incoming and outgoing traffic at either side of the interface. This depends on the FTS/NSS design and the deployment scenario. The entities involved might differ depending on the call direction, but have to be the same for all calls in the same direction in a single deployment scenario.

Annex A includes several deployment scenario examples that illustrate some of the Signalling Proxy deployment options.

5 Interface Functionality

This clause specifies functional requirements of the interface. The technical details are specified in clauses 6 and 7 of the present document.

5.1 Basic Call

The primary function to be delivered by the present interface is the means to initiate and tear down full duplex audio (voice) connections between the NSS and FTS with a single, logical, SIP Endpoint involved per connection on each side of the present interface that controls the connection as well as its respective Media Endpoint (compare clause 4).

Such a connection can be initialized by either NSS or FTS. The initiation phase shall specifically provide means and mechanisms for per call/connection capability exchange, media negotiation, progress indication as well as error indication and handling.

5.1.1 Progress Indication

Progress indication shall be provided via explicit signalling. In addition a progress tone generation policy, clearly stating which party shall generate which progress tones and when, is defined.

5.1.2 Early Media

Furthermore, the basic call procedure shall provide a means for media (i.e. audio) exchange prior to call setup completion. This shall be possible in the direction callee to caller only. This functionality is needed in order to provide pre-call announcements to the user before the dialog is established.

5.2 Connected Parties Identity Information

The calling party shall provide its identity information with the connection request.

The calling party shall be informed about the identity of the remote party.

The identity information shall contain a routable number in the underlying network's address space and in addition - if available - an EIRENE functional number.

Upon change of identity of either connected party an immediate identity update shall be transmitted to the other side.

5.3 Call Hold

Both endpoints of an established call shall be able to suspend an associated media stream and resume it at a later time. The endpoint holding the call shall inform the other party that the call is suspended and shall further inform the other party when the call is reconnected. The media stream is not just interrupted, but possibly redirected to some other source which generates, for example, an announcement or "music on hold".

5.4 Multi Level Precedence and Pre-emption

In order to allow differentiated/preferred treatment of calls of different/higher operational priority (e.g. emergency calls) when facing resource limits, the interface shall support signalling mechanisms and procedures to provide multi level precedence signalling and as well as call pre-emption functionality. Additionally, MLPP is used by the Signalling Endpoints to handle different priorities at the operational level. In particular this includes flagging of session priority and signalling flows for precedence blocking as well as pre-emption of calls, but not for reservation of resources.

The present document defines how call precedence and pre-emption is performed, but it does not define the algorithm that causes precedence blocking or call pre-emption to be performed.

5.5 Voice Group Call and Broadcast Call Control

Voice Group Calls [i.11] and Voice Broadcast Calls [i.12] are service implementations in the NSS.

The interface subject to the present document shall provide mechanisms for the FTS to control voice group calls and voice broadcast calls from the perspective of Fixed Terminal users as defined by EIRENE [1]. Control of voice group calls and voice broadcast calls from the perspective of other users is out-of-scope of the present document.

The following control mechanisms shall be supported on the present interface:

- Termination ("kill") of VGCS/VBS calls.
- Requests for muting and unmuting of the mobile terminal downlink of VGCS calls.

When an FTS subscriber is involved in such a call then it is connected by means of a point to point call on the present interface. The VGCS/VBS call is identified at the application level purely on the basis of the NSS subscriber number contained in the call signalling.

5.6 User-User-Information-Element Transport

User-to-User-Information-Elements (UUIE) [i.2] are used in GSM-R systems to carry EIRENE specific information and are exchanged within basic call control messages. TS 102 610 [9] specifies in detail the use and content of UUIEs in GSM-R and also distinguishes between international and national EIRENE UUIE tags.

The interface subject to the present document shall support a mechanism to transparently transport the content of the UUIE specified by TS 102 610 [9]. The transport of international and national tags and their values shall be supported.

5.7 Reason Transport

In most currently deployed GSM-R systems fixed and mobile terminals make use of end-to-end release cause signalling.

The present interface shall therefore support the transparent, end-to-end transport of release and disconnect reasons between the fixed terminals and GSM-R mobile terminals and vice versa.

6 Signalling Interface

6.1 Network Layer Protocol

NSS and FTS shall use IPv4 [2] as the network layer protocol.

Network Address Translation (NAT) is a method used for IP address translation between address realms. NAT adds complexity to higher layer protocols that is not dealt with in the present document. Therefore no form of NAT shall be implemented in the network infrastructure at the signalling interface. See [i.16] for more information on NAT.

6.2 Transport Layer Protocol

NSS and FTS shall use UDP [26] as the transport layer protocol.

Network Address and Port Translation (NAPT) is a form of NAT that extends to the transport layer. For the same reason as for pure network layer NAT, NAPT shall not be implemented in the network infrastructure at the signalling interface. See [i.16] for more information on NAPT.

6.3 Signalling Protocol

NSS and FTS shall support SIP in accordance with:

- RFC 3261 [3];

and SDP in accordance with:

- RFC 4566 [6]; and
- RFC 3264 [4];

as signalling protocols further qualified by statements in later clauses of the present document.

Requirements for support of other IETF RFCs and other standards are as stated in later clauses of the present document.

Deviations from and/or limited applicability of those RFCs are explicitly pointed out where appropriate.

6.3.1 SIP Entities

A SIP network can be composed of many logical SIP entities. Each entity provides specific functions and participates in SIP communication as a client (initiates requests), as a server (responds to requests), or as both. One physical entity can implement the functionality of more than one logical SIP entity.

On the interface subject to the present document the allowed, and therefore relevant, SIP entities are SIP User Agents (UAs) and SIP Proxy Servers, both defined in RFC 3261 [3].