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TECHNICAL SPECIFICATION

Rail Telecommunications (RT); Global System for Mobile communications (GSM); Usage of Session Initiation Protocol with ISUP encapsulation (SIP-I) and other IP based protocols for interconnection of GSM-R networks

Reference
DTS/RT-0054

Keywords
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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

In most of the GSM-R system deployments available at the time of the creation of the present document, the GSM-R Networks are interconnected using TDM based interfaces.

The present document deals with the introduction of IP interconnection towards external networks using SIP-I as a call control protocol, which seems to be beneficial both for operators and for network element vendors. ISUP interconnection and the supporting hardware elements are getting old and less effective compared to the new IP based interconnection methods already used in public networks.

In addition, the present document deals with MTP3 and MTP2 interworking between IP based interfaces and TDM based interfaces.

SIP-I is a modern, IP-based call control protocol to be used between core networks. It is applicable in particular, if the ISDN service interworking is kept completely.

While a number of interoperability specifications for various interfaces at various layers of GSM-R systems exist, the IP interface between GSM-R networks has not yet been addressed by any interoperability specification activity.

The present document addresses the interoperability specification between any two GSM-R Core Networks, where at least one of the networks uses IP based transport (Internet Protocol (IP) IETF RFC 791 [2]) for the interconnection.

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 interfaces subject to the present document.
- Clause 6 specifies in detail the signalling interfaces for all supported functions and services.
- Clause 7 specifies in detail the media interfaces.

1 Scope

The present document defines the IP based signaling and media interfaces between any two GSM-R Core networks of a set of GSM-R Core networks that share a GIRA and that agree to interconnect via IP based technology.

Furthermore it defines the IP based signaling and media interface between any GSM-R Core network that wants to interconnect with other GSM-R Core networks via IP based technology - and a centralized hub during a distinct phase of migration.

The present document defines the IP Interconnect between 2 GSM-R Core networks - together with the interworking with BICC/ISUP at the edges of the networks - and the IP Interconnection between a GSM-R Core network and the current TDM hub - together with the interworking with ISUP at the current TDM hub and the interworking with BICC/ISUP at the edge of the GSM-R network.

Only the network layer (e.g. IP) and layers that build on the network layer are considered. The lower layers are denoted as L2 and L1 and are out of scope in the present document.

The present document will address all services (bearer services, tele services, supplementary services and railway services) that are defined by relevant EIRENE SRS [1].

The present document could address all domains of the GSM-R networks (e.g. CS domain, PS domain, IMS domain, etc.), however in a first step only the CS domain and the PS domain are covered.

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 https://docbox.etsi.org/Reference.

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

[1] UIC CODE 951(0.0.2): "UIC Project EIRENE System Requirements Specification (Version 16.0)".

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

[3] Recommendation ITU-T Q.1912.5 (2018/01): "Interworking between session initiation protocol (SIP) and bearer independent call control protocol or ISDN user part".

[4] UIC REFERENCE O-8350 2.0: "FFFS for Voice and Data Services Interconnection & Roaming between GSM-R networks".

[5] ETSI TS 129 002: " Digital cellular telecommunications system (Phase 2+) (GSM); Universal Mobile Telecommunications System (UMTS); LTE; 5G; Mobile Application Part (MAP) specification (3GPP TS 29.002)".

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

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

[8] ETSI TS 129 164 (V12.0.0): " Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); Interworking between the 3GPP CS domain with BICC or ISUP as signalling protocol and external SIP-I networks (3GPP TS 29.164 version 12.0.0 Release 12)".

[9] IETF RFC 3312 (2002): "Integration of Resource Management and Session Initiation Protocol (SIP)".

[10] IETF RFC 4165 (2005): "Signaling System 7 (SS7) Message Transfer Part 2 (MTP2) - User Peer-to-Peer Adaptation Layer (M2PA)".

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

[12] Recommendation ITU-T G.711: "Pulse Code Modulation (PCM) of voice frequencies".

[13] IETF RFC 3551 (2003): "RTP Profile for Audio and Video Conferences with Minimal Control".

[14] IETF RFC 4040 (2005): "RTP Payload Format for a 64 kbit/s Transparent Call".

2.2 Informative references

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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] ETSI TS 103 389 (V3.1.1): "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".

[i.2] IETF RFC 6086 (2011): "Session Initiation Protocol (SIP) INFO Method and Package Framework".

[i.3] Recommendation ITU-T H.248.n: "Gateway Control Protocol, version 3".

[i.4] ETSI TS 129 332: "Digital cellular telecommunications system (Phase 2+) (GSM); Universal Mobile Telecommunications System (UMTS); Media Gateway Control Function (MGCF) - IM Media Gateway; Mn interface (3GPP TS 29.332)".

[i.5] ETSI TS 129 163: "Digital cellular telecommunications system (Phase 2+) (GSM); Universal Mobile Telecommunications System (UMTS); LTE; 5G; Interworking between the IP Multimedia (IM) Core Network (CN) subsystem and Circuit Switched (CS) networks (3GPP TS 29.163)".

[i.6] IETF RFC 2663 (1999): "IP Network Address Translator (NAT) Terminology and Considerations".

[i.7] IETF RFC 5226 (2008): "Guidelines for Writing an IANA Considerations Section in RFCs".

[i.8] IETF RFC 5727 (2010): "Change Process for the Session Initiation Protocol (SIP) and the Real-time Applications and Infrastructure Area".

3 Definition of terms, symbols and abbreviations

3.1 Terms

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

downlink path: IP path from a hub to a GSM-R Core network, optionally traversing one or more transit networks

end-to-end path: IP path from one GSM-R Core network to another GSM-R Core network, optionally traversing one or more transit networks or even hubs

IP interconnect: all facilities that are necessary to transmit the PDUs of all IP based protocols for the signalling plane and for the media plane

NOTE: The IP Interconnect can take place between 2 GSM-R Core Networks or between a GSM-R Cone Network and a hub.

Logical Link (LL): end-to-end connection at layer 2

NOTE: E.g. an LLC connection over Ethernet or a Frame Relay PVC.

network connection: generic term to address one of the following:

- a tuple of UDP transport addresses;
- a TCP connection;
- an SCTP association; or
- an RTP session

TDM interconnect: all facilities that are necessary to transmit ISUP, SCCP or MTP3 PDUs or bearer traffic

NOTE 1: The TDM Interconnect can take place between 2 GSM-R Core Networks or between a GSM-R Core Network and a hub.

NOTE 2: The TDM Interconnect is not specified in the present document, but its existence is a precondition as far as the interworking between IP Interconnect and TDM Interconnect is specified here.

uplink path: IP path from a GSM-R Core network to a hub, optionally traversing one or more transit networks

3.2 Symbols

Void.

3.3 Abbreviations

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

ACM	Address Complete Message
ANM	Answer message
AoCC	Advice of Charge (Charging)
AoCI	Advice of Charge (Information)
BAIC	Barring All Incoming Calls
BAOC	Barring All Outgoing Calls
BGP	Border Gateway Protocol
BICC	Bearer Independant Call Control
BOIC	Barring All Outgoing International Calls
CAMEL	Customized Application of Mobile Enhanced Logic
CAP	CAMEL Application Part
CCBS	Completion of Call to Busy Subscribers

CDN	Call Dial Number
CFB	Call Forwarding on Busy
CFNRc	Call Forwarding on not Reachable
CFNRy	Call Forwarding on no Reply
CFU	Call Forwarding Unconditionally
CGN	Code Group Number
CLIP	Calling Line Identification Presentation
CLIR	Calling Line Identification Presentation Restriction
CoLP	Connected Line Identification Presentation
CoLR	Connected Line Identification Presentation Restriction
CON	CONnect
CPG	Call ProGress
CS	Circuit Switched
CSD	Circuit Switched Data
CUG	Closed User Group
CW	Call Waiting
DL	Downlink
DNS	Directory Name Server
DTMF	Dual Tone Multiple Frequency
E2E	End-to-End
ECT	Explicit Call Transfer
EIRENE	European Integrated Radio Enhanced Network
eMLPP	Enhanced Multilevel Precedence and Preemption
ETCS	European Train Control System
FAC	Facility message
FQDN	Full Qualified Domain Name
FTS	Fixed Terminal Subsystem
GGSN	Gateway GPRS Support Node
GIRA	GSM-R Interconnection and Roaming Agreement

NOTE: Defined at FFFS I&R [4].

GN	Group Number
GNS	GPRS domain Name System
GPRS	General Packet Radio Service
GSM	Global System for Mobile communications
GSM-R	GSM for Railways
GTP-C	GPRS Tunneling Protocol - Control plane
GTP-U	GPRS Tunneling Protocol - User plane
GW	Gateway
HLR	Home Location Register
IAM	Initial Address message
IANA	Internet Assigned Number Association
IMS	IP Multimedia Subsystem
INTL	International
IP	Internet Protocol
ISDN	Integrated Services Digital Network
ISUP	ISDN User Part
IWU	Inter Working Unit
L1	Layer 1 (Bit Transmission)
L2	Layer 2 (Link)
LL	Logical Link
LLC	Logical Link Control
M2PA	MTP2 Peer-to-Peer Adaptation
M3UA	MTP3 User Adaptation
MAP	Mobile Application Part
MGW	Media Gateway
MLPP	Multi Level Precedence and Pre-emption
MPTY	Multi Party
MSC	Mobile Switching Center
MSC-S	Mobile Switching Center - Server
MTP	Message Transfer Part

NAPT	Network Address and Port Translation
NAT	Network Address Translation
NOA	Nature Of Address
NPI	Number Plan Identification
NSS	Network and Switching Subsystem
PCM	Pulse Code Modulation
PLMN	Public Landbased Mobile Network
PS	Packet Switched
PVC	Permanent Virtual Connection
REC	Railway Emergency Call
RTCP	Real-time Control Protocol
RTP	Real-time Transport Protocol
SCCP	Signaling Connection Control Part
SCTP	Stream Control Transmission Protocol
SDP	Session Description Protocol
SGSN	Serving GPRS Support Node
SIP	Session Initiation Protocol
SIP-I	SIP with embedded ISUP
SIP-R	Service Independent Protocol for Railways
SMS MO	Short Message Service, Mobile Originated
SMS MT	Short Message Service, Mobile Terminated
SMS	Short Message Service
SRS	System Requirement Specification
STP	Signaling Transfer Point
TCAP	Transaction Capabilities Part
TCP	Transmission Control Protocol
TDM	Time Division Multiplexing
TMR	Transmission Medium Requirement
UDP	User Datagram Protocol
UIC	International Union of Railways
UL	Uplink
URI	Unified Resource Identifier
USSD	Unstructured Supplementary Service Data
UUS	User to User Signalling
VBS	Voice Broadcast Service
VGCS	Voice Group Call Service
VLR	Visitor Location Register
VNF	Virtual Network Function

4 Reference system architecture

The reference system architecture describes a layered network, which can be used to display the different use cases of a GIRA, in case that at least one of both GSM-R Core networks connects to the other via IP based transport.

Use Case I: Both networks use IP based transport to connect to each other (IP Interconnect).

Use Case II: One of both networks (here GSM-R Core B) still uses TDM based transport (TDM Interconnect).

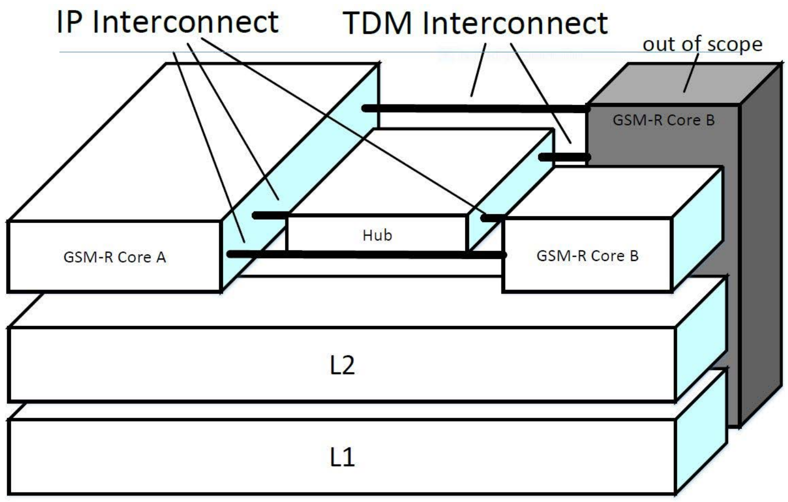


Figure 1: Interconnection of two GSM-R Core Networks subject to a GIRA

The migration from current situation over Use Case II to Use Case I will need a hub equipped with IP Interconnect technology, which shall support TDM based Interconnect, too, for the sake of Use Case II.

If considering two concrete networks (subject to a GIRA), then the nomenclature in the present document is to be read according to following time line (this ordered list of migration phases is just informational and should not prejudice any real migration strategy):

1) GSM-R Core network A is the network that is the first of the two, that connects to the hub via IP Interconnect interfaces (signalling interface and media interface). The TDM Interconnect to the hub can be removed then.

2) GSM-R Core network B still connects to the hub and to the GSM-R Core network A by the use of TDM Interconnect interfaces. This fact may be kept unchanged for some time.

3) Then GSM-R Core network B begins to support IP Interconnect interfaces towards the hub and towards GSM-R Core network A. Finally, GSM-R Core Network A can abandon the last TDM connection with GSM-R Core Network B.

4) After all GSM-R Core Networks of a hub region have switched to the use of IP Interconnect interfaces, then the TDM based hub equipment can be removed from the hub.

During the phase of migration from TDM Interconnect interfaces to IP Interconnect interfaces, all hubs within one hub region should be equipped with IP Interconnect technology as well as with TDM Interconnect technology.

Figure 2 displays the lower layer protocols and the network functions that are needed within a hub during the phase of migration.

The term "network function" shall indicate that the present document does not care if the functions are implemented on "bare metal" platforms (like server machines), on some virtualized environment or even as VNFs in a cloud environment.

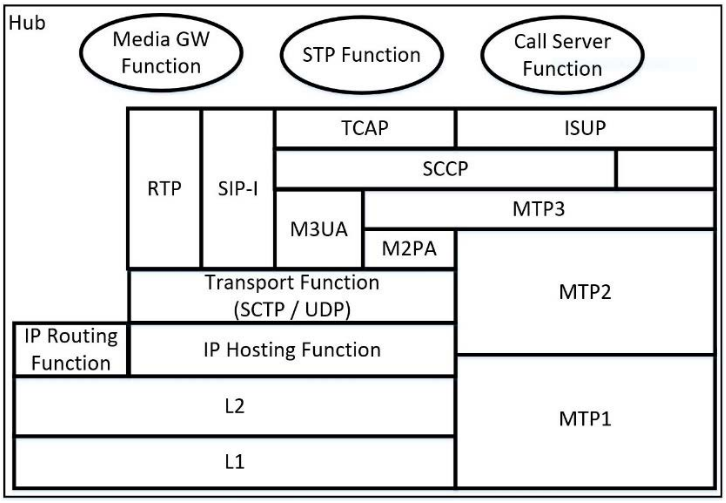


Figure 2: Lower layer protocols and network functions of a hub during migration

Figure 3 displays the lower layer protocols and network functions of a GSM-R Core network, as they are seen from an IP Interconnect interface.

The term "GSM-R Core function" means functions like MSC-S, MGW, VLR, HLR, SGSN, GGSN and so on. The term "network function" shall indicate that the present document does not care if the functions are implemented on "bare metal" platforms (like server machines), on some virtualized environment or even as VNFs in a cloud environment.

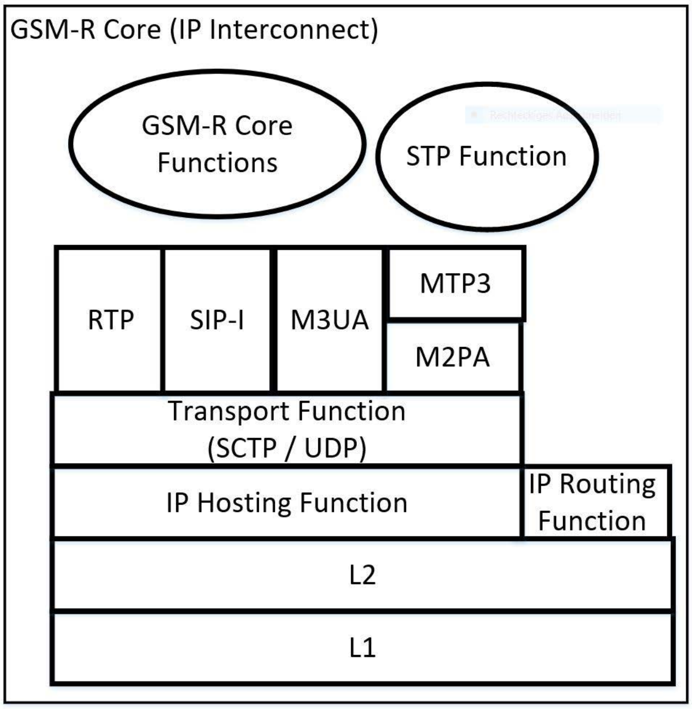


Figure 3: Lower layer protocols and network functions, as seen from IP Interconnect interfaces

5 Interface functionality

5.0 General description

This clause specifies functional requirements for the IP Interconnect interfaces, sorted by domain (CS domain, PS domain, IMS domain, etc.) and by service.

The IP Interconnect interfaces shall be used to transport following traffic of the CS domain:

- SIP-I traffic for call control (see Recommendation ITU-T Q.1912.5 [3], profile C - for the specification of SIP-I);
- SCCP/MTP3/M2PA and SCCP/M3UA traffic for mobility management (SS7 will be mainly used to transport MAP traffic, see ETSI TS 129 002 [5] for the specification of MAP); and
- RTP/RTCP traffic for media streams (see IETF RFC 3550 [6] for more information about RTP).

The IP Interconnect interfaces shall be used to transport following traffic of the PS domain:

- TCAP/SCCP/M3UA traffic to retrieve subscriber information from the HLR;
- GTP-C/UDP traffic for GPRS control plane signalling;
- GTP-U/UDP traffic for transfer of GPRS user data;
- DNS traffic towards the ROOT DNS in the IP-hub to resolve DNS APN requests.

GTP-C, GTP-U and DNS shall be used as specified by UIC REFERENCE O-8350 2.0 [4].

The IP Interconnect interfaces shall be used to transport following traffic of the IMS domain:

- The IMS domain is out of scope, currently.

Services of GSM-R networks are specified by the EIRENE SRS [1]:

- support of GSM Teleservices is specified in [1], chapter 2.2.1 (table 2-1);
- support of bearer services is specified in [1], chapter 2.3.1 (table 2-2);
- support of GSM supplementary services is specified in [1], chapter 2.4.1 (table 2-3); and
- support of railway specific services is specified in [1], chapter 2.5.1 (table 2-4).

Clauses 6 and 7 of the present document describe, how the relevant standards (e.g. Recommendation ITU-T Q.1912.5 [3]) have to be applied to the signalling interfaces and to the media interfaces, respectively, in order to fulfil the requirements from this clause. In particular clauses 6.4 and 7.4 inherit their clause structure from clause 5.

The IP Interconnect interfaces that are used to connect GSM-R Core Networks and hubs, are depicted in Figure 1.

The routing, which is applied in IP networks, constitutes one-way paths, each path originating at an IP address and terminating at another IP address.

Therefore the following terms are established for use in the present document:

- **Uplink path**: IP path from a GSM-R Core network to a hub, optionally traversing one or more transit networks.
- **Downlink path**: IP path from a hub to a GSM-R Core network, optionally traversing one or more transit networks.
- **E2E path**: IP path from one GSM-R Core network to another GSM-R Core network, optionally traversing one or more transit networks or even hubs.

Each IP path routes IP traffic in one direction.

For the sake of redundancy, more than one path should be possible to each destination.

If a link or a node gets into failure mode, then the used L2 protocol shall detect the failure and report the failure to the local IP routing function. The routing protocol (e.g. BGP, etc.), shall be used by the IP routing functions to switch over to an available path, if possible.

The same is valid for switch-back, when the L2 protocol detects the restored availability of a link or node and triggers the IP routing functions to switch back to the preferred path.

An IP path can traverse transit networks or even hubs in one of two ways:

1) either L2 switching is used to traverse the network; or

2) IP routing is used to traverse the network.

The design decision, whether to use IP routing or L2 switching depends on non-functional requirements like:

- performance requirements;
- security requirements;
- design of Ethernet collision domains; and so on.

However the following figures 4 and 5 are intended to display the possibilities and to serve as a common base for understanding and for the definition of terms and conditions.

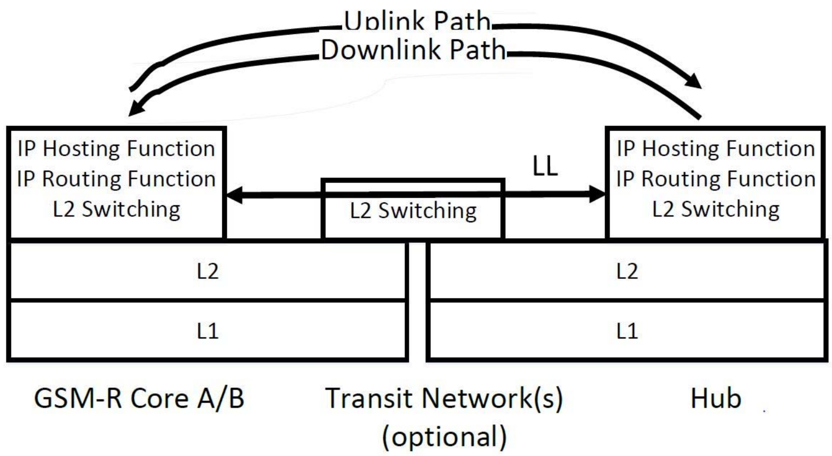


Figure 4: UL/DL IP path, when being switched at transit

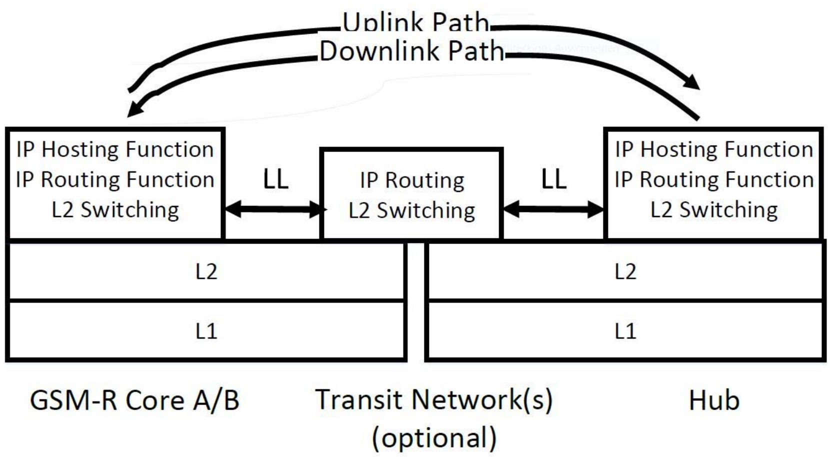


Figure 5: UL/DL IP path, when being routed at transit

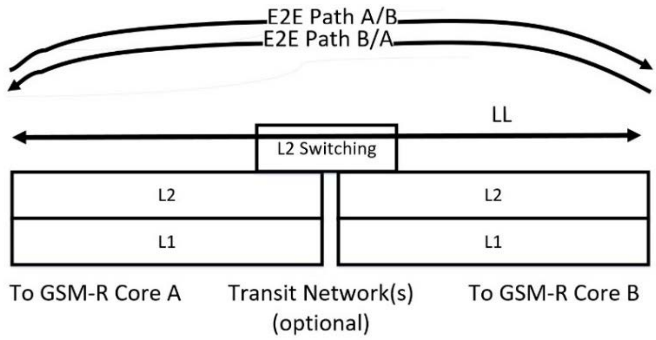


Figure 6: E2E IP path, when being switched at transit

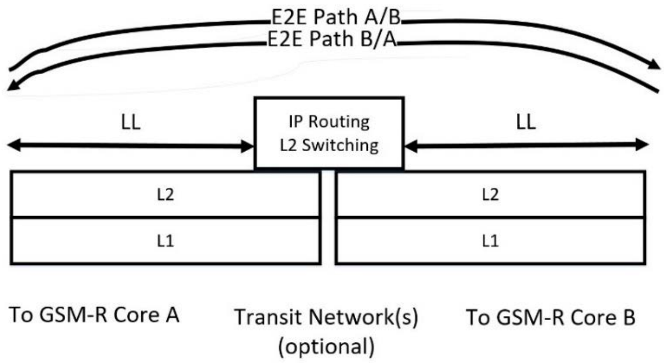


Figure 7: E2E IP path, when being routed at transit

5.1 CS domain - transport of SIP-I, SCCP/MTP3, RTP/RTCP

5.1.0 General information

Clause 5.1 establishes the requirements about how the IP paths established by clause 5.0 shall be used within the CS domain:

- Clauses 5.1.1 to 5.1.4: Common requirements for Use Case I and Use Case II.
- Clause 5.1.5: Requirements specific to Use Case II.

The paths shall be used for the transport of traffic specific to the CS Domain between two GSM-R Core networks or between a GSM-R Core network and a hub, e.g.:

1) Maintenance, e.g.:
    a) SIP OPTIONS pinging
    b) SCTP heartbeats
2) Call Control, e.g.:
    a) SIP-I traffic for setup, maintenance and teardown of point-to-point calls, e.g.:
        i) Point-to-point speech calls
        ii) Call legs of MPTY calls
        iii) Branches of VGCS/VBS calls (between anchor MSC and relay MSC)
        iv) Dispatcher legs of VGCS/VBS calls
        v) Call leg for inter PLMN handover

b) SIP-I traffic for GSM supplementary services

3) Mobility Management, e.g.:

a) MAP traffic for roaming and handover

b) MAP traffic for branches of VGCS/VBS calls

c) MAP traffic for SMS

4) Bearer Traffic, e.g.:

a) Transmission of speech data (e.g. G.711)

b) Transmission of clearmode data (e.g. CSD for ETCS)

c) Transmission of RTP events (e.g. DTMF)

5.1.1 CS bearer services

EIRENE SRS [1] specifies in clause 2.3.1, table 2-2, which Bearer Services have to be supported in GSM-R networks. Clauses 5.1.1, 6.4.1.1 and 7.4.1.1 deal with the requirements and realization of those bearer services, with the exception of GPRS and eGPRS.

The real-time transport protocol RTP shall be used for the transmission of speech and clearmode data at the media interface. DTMF tones and/or RTP events shall be transmitted, too.

The Session Description Protocol shall be used to describe the RTP Sessions that shall be setup, modified or deleted. The SDP offer and SDP answer shall be embedded into SIP messages according to IETF RFC 3264 [11].

As an operator's option, a Media Inactivity Detection may be supported, in order to tear down calls, when the MGW detects inactivity of RTP traffic for some interval of time.

5.1.2 Teleservices

5.1.2.0 General

EIRENE SRS [1] specifies in clause 2.2.1, table 2-1, which GSM Teleservices have to be supported in GSM-R networks. The headings of the sub-clauses of clause 5.1.2, of clause 6.4.1.2 and of clause 7.4.1.2 have been setup according to the rows of that table.

5.1.2.1 Speech (telephony, emergency calls)

Roaming and Handover shall be available by the MAP mobility management protocol as specified in ETSI TS 129 002 [5] for inbound and outbound roamers of GSM-R Core network A and GSM-R Core network B.

Every SS7 traffic between the two GSM-R Core networks, i.e. MTP3, SCCP, MAP, CAP and so on (except ISUP), shall be routed by the STP functions, either via an E2E path, or , e.g. in case of path failure or because the networks are not geographically adjacent - via an UL path and a DL path. The STP function at the hub need not act as a MAP endpoint nor as a CAP endpoint in the latter case.

Call setup, call teardown and call maintenance shall be available by the SIP-I call control protocol (Recommendation ITU-T Q.1912.5 [3], profile C) for point-to-point calls between the GSM-R Core networks via the IP Interconnect interface(s).

The interworking of SIP precondition procedures with ISDN continuity check procedures according to Recommendation ITU-T Q.1912.5 [3], profile C - may be supported at the IP Interconnect interfaces, as an operator's option, if the interface partners agree on it.

Use of SIP in bloc signalling is recommended at the IP Interconnect interfaces. However, the interworking between SIP overlap signalling based on the INVITE message and BICC/ISUP overlap signalling according to Recommendation ITU-T Q.1912.5 [3], profile C - may be supported at the IP interconnect interfaces, as an operator's option, if the interface partners agree on it.

If the call is routed over the Call Server function of the hub, e.g. in case of path failure or because the networks are not geographically adjacent - then this Call Server function shall perform:

- SIP-I Call processing.
- Controlling the Media GW function at the hub.
- Interworking of SIP-I with ISUP via the Media GW and/or STP function (in case of Use Case II, see clause 5.1.5.1).

For the purpose of the present document, the Call Server function at the hub does not need to support:

- Mobility Management procedures.

If the call is routed over the Call Server function of the hub, e.g. in case of path failure or because the networks are not geographically adjacent - then the Media GW function at the hub shall perform:

- RTP Relay or Transcoding of Speech Data (in case of Use Case I).
- RTP Relay of Clearmode Data (in case of Use Case I).
- Interworking of Speech/Clearmode Data with raw data over 64 kbit/s channel (in case of Use Case II, see clause 5.1.5.2).

The IP Interconnect interfaces may convey in various parameters not only E.164 numbers, but also numbers that have to be interworked with BICC/ISUP parameters according to the EIRENE numbering plan. The IP Interconnect interface shall be capable of conveying and interworking such EIRENE numbers according to EIRENE SRS [1].

5.1.2.2 Short Message Service

Interworking of SMS MT with SMS MO shall be possible by routing MAP traffic between GSM-R Core network A and GSM-R Core network B. MAP is specified at ETSI TS 129 002 [5].

NOTE: Routing of SS7 traffic via the IP Interconnect interfaces is already required generally by clause 5.1.2.1.

5.1.2.3 Facsimile transmission

N/A.

5.1.2.4 Voice Group Service (VGCS, VBS)

The support of VGCS/VBS needs an international connection to support international group calls, i.e.:

- Connecting an anchor MSC with a relay MSC of an international GSM-R network.
- A VGCS/VBS including international dispatchers.

All these use cases are supported via the IP Interconnect interfaces and via the hub, if the requirements from clause 5.1.2.1 are fulfilled (point-to-point call, routing of MAP), and if following requirements are fulfilled:

- transport of DTMF for Talker Control (Mute/Unmute) and for Group Call Termination (Kill);
- transport of DTMF for the Grant Tone.

ETSI TS 103 389 [i.1] has provided the definition of a SIP INFO service, which allows to transmit out of band DTMF sequence information or explicit commands for the group call control from the dispatcher to the NSS via SIP Signalling.

However, explicit commands for group call control will fall back to DTMF sequences, whenever being interworked with ISUP.

The present document requires to support inband DTMF transport using a separate RTP dynamic payload type, according to IETF RFC 4733 [7], in both directions, at the IP Interconnect interfaces.

Full (forward and backward) interworking with out of band DTMF signaling as an option (at BICC trunks) and "real" inband DTMF (at ISUP trunks and BICC trunks) shall be supported at the edge of the GSM-R Core networks and at the hub.

If the interface partners agree on the usage of out of band DTMF at an IP Interconnect interface, then the transport of DTMF sequences shall be done via SIP INFO messages according to the format specified in ETSI TS 103 389 [i.1]. In this case, IETF RFC 6086 [i.2] should be used as a guideline to ensure compatibility with possible future info packages.

In case of using out of band DTMF signaling via SIP INFO, the explicit command for group call control should be added to the message body according to the format specified in ETSI TS 103 389 [i.1], if available (e.g. if a SIP-R trunk in the dispatcher's network is directly interworked with an IP Interconnection interface).

The Grant Tone is not specified in the list of commands of the explicit group call control and shall always be transported inband at the IP Interconnect interfaces.

5.1.3 GSM supplementary services

5.1.3.0 General

EIRENE SRS [1] specifies in clause 2.4.1, table 2-3, which GSM Supplementary services are mandatory for GSM-R networks and which are not. The headings of the sub-clauses of clause 5.1.3, of clause 6.4.1.3 and of clause 7.4.1.3 have been setup according to the rows of that table.

5.1.3.1 CLIP / CLIR

Interworking of CLIP / CLIR information at the boundary of GSM-R Core networks via IP Interconnect interfaces shall be done according to Recommendation ITU-T Q.1912.5 [3], profile C, with following changes and clarifications.

Since CLIP is mandatory according to EIRENE SRS [1], but CLIR is not, hence the support of CLIR is up to bilateral agreement between the operators of GSM-R Core network A and GSM-R Core network B.

If CLIR information is transmitted via IP Interconnect interfaces, then it shall be done according to Recommendation ITU-T Q.1912.5 [3], profile C.

5.1.3.2 CoLP / CoLR

Interworking CoLP / CoLR information at the boundary of GSM-R Core networks via IP Interconnect interfaces shall be done according to Recommendation ITU-T Q.1912.5 [3], profile C, with following changes and clarifications.

Since CoLP is mandatory according to EIRENE SRS [1], but CoLR is not, hence the support of CoLR is up to bilateral agreement between the operators of GSM-R Core network A and GSM-R Core network B.

If CoLR information is transmitted via IP Interconnect interfaces, then it shall be done according to Recommendation ITU-T Q.1912.5 [3], profile C.

5.1.3.3 Call Forwarding (CFU, CFB, CFNRy, CFNRc)

Interworking Call Forwarding information at the boundary of GSM-R Core networks via IP Interconnect interfaces shall be done according to Recommendation ITU-T Q.1912.5 [3], profile C, with following changes and clarifications.

Since CFU and CFB are mandatory according to EIRENE SRS [1], but CFNRy and CFNRc are not, hence the support of CFNRy and CFNRc are up to bilateral agreement between the operators of GSM-R Core network A and GSM-R Core network B.

If CFNRy or CFNRc information is transmitted via IP Interconnect interfaces, then it shall be done according to Recommendation ITU-T Q.1912.5 [3], profile C.

5.1.3.4 Call Waiting (CW)

Interworking Call Waiting information at the boundary of GSM-R Core networks via IP Interconnect interfaces shall be done according to Recommendation ITU-T Q.1912.5 [3], profile C.

5.1.3.5 Call Hold (HOLD)

Interworking Call Hold information at the boundary of GSM-R Core networks via IP Interconnect interfaces shall be done according to Recommendation ITU-T Q.1912.5 [3], profile C.

NOTE: Recommendation ITU-T Q.1912.5 [3] assumes that possible hold announcements are applied at the same network, at which the subscriber is attached that has set the call on hold. This is a major difference to ETSI TS 103 389 [i.1], which describes the interface between NSS and FTS and does not share this assumption. This might imply simpler handling of hold/retrieve scenarios at the IP Interconnect interface than at the interface between NSS and FTS.

5.1.3.6 Multi Party service (MPTY)

Interworking information about Manual Conferences (CONF or 3PTY) at the boundary of GSM-R Core networks via IP Interconnect interfaces shall be done according to Recommendation ITU-T Q.1912.5 [3], profile C.

5.1.3.7 Closed User Group (CUG)

Interworking CUG information at the boundary of GSM-R Core networks via IP Interconnect interfaces shall be done according to Recommendation ITU-T Q.1912.5 [3], profile C.

5.1.3.8 Advice of Charge (Information) (AoCI)

N/A.

5.1.3.9 Advice of Charge (Charging) (AoCC)

N/A.

5.1.3.10 Barring (BAOC, BOIC, BOIC-exHC, BAIC, BIC-Roam)

N/A.

5.1.3.11 Unstructured Supplementary Service Data (USSD)

During roaming scenarios, the USSD, e.g. for Follow Me service, may traverse the IP Interconnect interface.

NOTE: The routing of SCCP and MTP3 traffic at the IP Interconnect interface is already required by clause 5.1.2.1.

5.1.3.12 Follow me

During roaming scenarios, the USSD for Follow Me service, may traverse the IP Interconnect interface.

NOTE: The routing of SCCP and MTP3 traffic at the IP Interconnect interface is already required by clause 5.1.2.1.

5.1.3.13 Sub-addressing

Interworking SUB information at the boundary of GSM-R Core networks via IP Interconnect interfaces shall be done according to Recommendation ITU-T Q.1912.5 [3], profile C.

5.1.3.14 eMLPP

Interworking MLPP information at the boundary of GSM-R Core networks via IP Interconnect interfaces shall be done according to Recommendation ITU-T Q.1912.5 [3], profile C.

5.1.3.15 ECT

Interworking ECT notifications at the boundary of GSM-R Core networks via IP Interconnect interfaces shall be done according to Recommendation ITU-T Q.1912.5 [3], profile C, with following changes and clarifications.

Since ECT is optional according to EIRENE SRS [1], hence the support of ECT notifications is up to bilateral agreement between the operators of GSM-R Core network A and GSM-R Core network B.

If ECT information is transmitted via IP Interconnect interfaces, then it shall be done according to Recommendation ITU-T Q.1912.5 [3], profile C.

5.1.3.16 Completion of Calls to Busy Subscribers (CCBS)

Interworking CCBS information at the boundary of GSM-R Core networks via IP Interconnect interfaces shall be done according to Recommendation ITU-T Q.1912.5 [3], profile C, with following changes and clarifications.

Since CCBS is optional according to EIRENE SRS [1], hence the support of CCBS is up to bilateral agreement between the operators of GSM-R Core network A and GSM-R Core network B.

If CCBS information is transmitted via IP Interconnect interfaces, then it shall be done according to Recommendation ITU-T Q.1912.5 [3], profile C. This also includes the correct routing of SCCP between the two networks.

5.1.3.17 User-to-User Signalling (UUS1)

Interworking User to User information at the boundary of GSM-R Core networks via IP Interconnect interfaces shall be done according to Recommendation ITU-T Q.1912.5 [3], profile C.

5.1.4 Railway specific services

5.1.4.0 General

EIRENE SRS [1] specifies in clause 2.5.1, table 2-4, which Railway Specific Services are supported in GSM-R networks. The headings of the sub-clauses of clause 5.1.4, of clause 6.4.1.4 and of clause 7.4.1.4 have been setup according to the rows of that table.

5.1.4.1 Functional addressing

Functional addressing shall be supported by the support of EIRENE numbers at the IP Interconnect interfaces (already required by clause 5.1.2.1) and by the UUS1 (see clause 5.1.3.17).

5.1.4.2 Location dependent addressing

Functional addressing shall be supported by the support of EIRENE numbers at the IP Interconnect interfaces (already required by clause 5.1.2.1) and by the UUS1 (see clause 5.1.3.17).

5.1.4.3 Shunting mode

N/A.

5.1.4.4 Multiple driver communications

The MPTY service shall be supported at the IP Interconnect interfaces (see clause 5.1.3.6).

5.1.4.5 REC / eREC

REC and eREC shall be supported by the support of eMLPP (clause 5.1.3.14), of VGCS/VBS (clause 5.1.2.4) and of UUS1 (see clause 5.1.3.17) at the IP Interconnect interfaces.

5.1.5 IP/TDM interworking for the CS domain

5.1.5.0 Introduction

Clauses 5.1.1 to 5.1.4 describe the common requirements for Use Case I and Use Case II that are applicable to the IP Interconnect and to the interworking at the edges of the GSM-R Core Networks.

Use Case I:

Normal: GSM-R Core A --- IP Interconnect --- GSM-R Core B

In case of use case I a fall back should not be necessary, because the IP Interconnect can be setup in a redundant way, however it is possible to implement the following:

Fallback: GSM-R Core A --- IP Interconnect --- Hub --- IP Interconnect --- GSM-R Core B

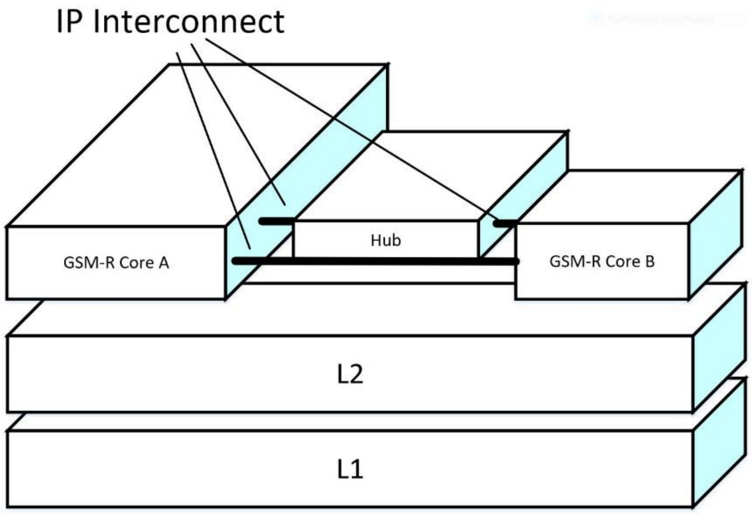
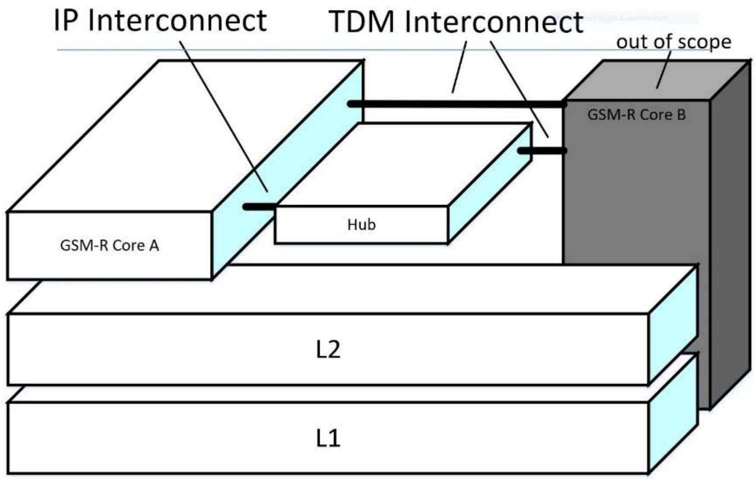


Figure 8: Reference Architecture for Use Case I

The following rest of clause 5.1.5 deals with interworking at the TDM hub, which is only applicable in case of Use Case II.

Use Case II:

- GSM-R Core A --- IP Interconnect --- Hub --- TDM Interconnect --- GSM-R Core B
- Fallback scenario A: connection via another Hub (if GSM-R operator A prefers getting rid of TDM sooner)
- Fallback scenario B: direct TDM connection (if GSM-R operator A can wait for getting rid of TDM later)



NOTE: As soon as all GSM-R Core networks will have switched to IP Interconnect, then clause 5.1.5 is no longer applicable. (R4 interconnect still there).

Figure 9: Reference Architecture for Use Case II

5.1.5.1 Interworking SIP-I with ISUP, Interworking SS7

Teleservices + Supplementary Services

GSM Teleservices and GSM Supplementary Services shall be interworked at the hub between SIP-I protocol and ISUP protocol according to ETSI TS 129 164 [8], based on ETSI TS 129 1633 [i.5], based on Recommendation ITU-T Q.1912.5 [3], profile C, in case of Use Case II.

Furthermore, the associated SS7 traffic (MAP, CAP, etc.) shall be routed between TDM based transport (MTP3, MTP2, etc.) and IP based transport (M3UA, M2PA, SCTP, etc.) at the hub, in case of Use Case II.

This shall be done for:

- MAP traffic for roaming and handover
- Setup, maintenance and teardown of point-to-point calls:
  - point-to-point speech calls
  - Call legs of MPTY calls
  - Branches of VGCS/VBS calls (between anchor MSC and relay MSC)
  - Dispatcher legs of VGCS/VBS calls
  - Call legs for inter PLMN handover
- Call control for GSM supplementary services
- MAP traffic for SMS
- MAP traffic for branches of VGCS/VBS calls

Interworking of GSM Supplementary Services between ISUP and SIP-I at the hub shall be done in an analogue way, like it is done at the edge of a GSM-R Core network (as required by clauses 5.1.3.1 to 5.1.3.17).

Railway Services

The transport of Railway Services via the IP Interconnect interface and the interworking at the edge of the GSM-R network and at the hub shall be done as it is done for the Teleservices and Supplementary Services that are combined to the Railway Services. Clause 5.1.4 describes which services are combined to Railway Services.

5.1.5.2 Interworking the bearer services

All bearer services that are routed over the hub shall be interworked at the hub, in case of Use Case II. For this purpose, the speech data or the clearmode data from the RTP payload shall be interworked with the 64 kbit/s channels at the TDM Interconnect interfaces.

Interworking of DTMF is required by clause 5.1.2.4.

5.2 PS domain - transport of GTP-C, DNS, GTP-U

5.2.1 PS bearer services

To establish connectivity between SGSN and GGSN, GPRS Tunnelling Protocol shall be used according to clause 8.5 of UIC REFERENCE O-8350 2.0 [4].

5.2.2 DNS services

The GPRS Domain Name System (GNS) shall be utilized to resolve APNs. If the local GNS is unable to resolve the DNS APN request, the local DNS shall contact the ROOT DNS operated by the IP-hub/IPX, as specified by clause 8.5 of UIC REFERENCE O-8350 2.0 [4].

5.2.3 IP/TDM interworking for the PS domain

Interworking of SCCP/MTP3 is handled in clause 5.1.5.1 completely, i.e. for the CS domain and for the PS domain.

6 Signalling interface

6.0 General description

The signalling interface (clause 6) and the media interface (clause 7) of the IP Interconnect interfaces shall be used to transport:

- SIP-I traffic for call control and maintenance;
- RTP/RTCP traffic for the handling of the bearer;
- SCCP / MTP3 / M2PA and SCCP / M3UA traffic for the handling of mobility management, group call control; and SMS.

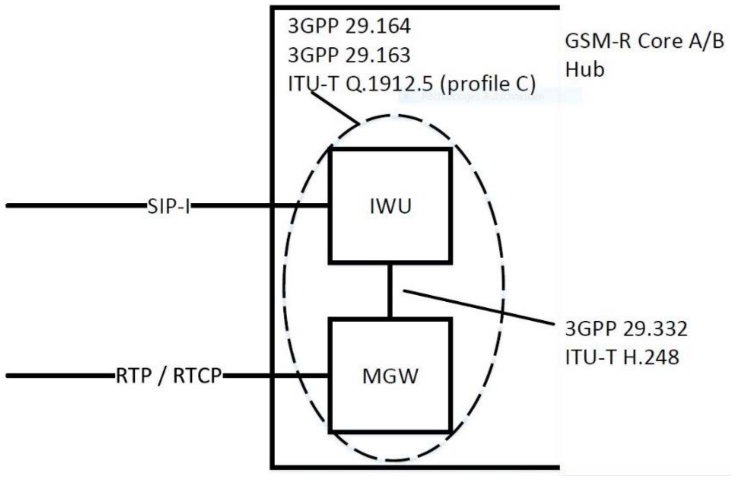


Figure 10: Major Standards for the handling of SIP-I and RTP/RTCP traffic at the edge of the network

The IWU is typically integrated with an MSC-S, but it may also be connected to an MSC externally. Recommendation ITU-T Q.1912.5 [3], profile C - shall be applied with refinements by ETSI TS 129 163 [i.5], by ETSI TS 129 164 [8] and by the present document.

If the MGW is separated from the IWU, then it should be connected to the IWU according to Recommendation ITU-T H.248 [i.3] with refinements by ETSI TS 129 332 [i.4] and by the present document.

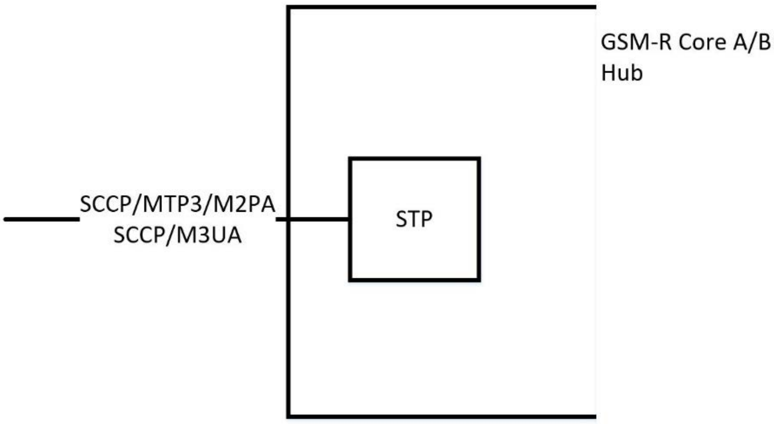


Figure 11: SCCP / MTP3 traffic at the edge of the network

The STP shall route SCCP / MTP3 traffic over the M2PA and SCTP, according to IETF RFC 4165 [10].

6.1 Network layer protocol

Clause 6.1 specifies the common properties of the network protocol IP, which is used at the media interface (see clause 7) and at the signaling interface (clause 6).

The key characteristics of the external SIP-I network are specified in ETSI TS 129 164 [8], clause 4.2, with following changes and clarifications regarding the network layer protocol.

IPv4 [2] shall be supported at the IP Interconnect interface. The support of IPv6 is not required, but may be agreed upon by two interface partners.

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 signaling interface nor at the media interface. More information on NAT can be found in IETF RFC 2663 [i.6].

6.2 Transport layer protocols

Clause 6.2 specifies the properties of the transport protocols UDP, TCP and SCTP, as they are used at the signaling interface and at the media interface (UDP is also used at the media interface).

The key characteristics of the external SIP-I network are specified in ETSI TS 129 164 [8], clause 4.2, with following changes and clarifications regarding the transport layer protocols:

- UDP shall be supported for the transport of SIP-I
- SCTP shall be supported for the transport of SIP-I
- TCP may be supported for the transport of SIP-I
- UDP shall be supported for the transport of RTP/RTCP (see clause 7)
- SCTP shall be supported for the transport of M3UA or M2PA (SS7 transport)

Network Address and Port Translation (NAPT) is a form of NAT that extends to the transport layer. For the same reasons as for pure network layer NAT, NAPT shall not be implemented in the network infrastructure at the signaling interface nor at the media interface. See IETF RFC 2663 [i.6] for more information in NAPT.

6.3 Signalling protocols

6.3.1 SIP-I

SIP-I - as it shall be used at IP Interconnect interfaces - is specified by Recommendation ITU-T Q.1912.5 [3], profile C -, with refinements by ETSI TS 129 163 [i.5], by ETSI TS 129 164 [8] and by the present document.

NOTE 1: Not all features of the ISDN can be currently transported via the SIP protocol, unless implementing proprietary SIP extensions.

NOTE 2: Hence the SIP-I protocol circumvents this problem by copying the complete ISUP message into the body of the SIP message, thus "piggy-backing" the complete ISDN information for the peer.

NOTE 3: Many problems of interworking ISDN with SIP, e.g. the problem of distinction between audio and data calls with the help of the TMR parameter (transmission medium requirement parameter), are solved by SIP-I.

SIP-I can be transported between two IWUs, using one or more network connection(s), which use the IP paths between the two IWUs.

The term "network connection" is a generic term that is used in the present clause to describe:

- A tuple of two UDP transport addresses - if SIP-I is transported over UDP
- A TCP connection - if SIP-I is transported over TCP
- An SCTP association - if SIP-I is transported over SCTP

Figure 12 depicts a set of network connections that are subject to a GIRA.

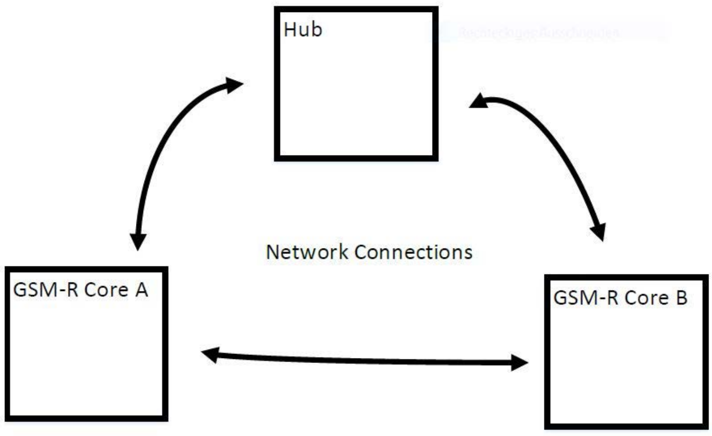


Figure 12: Network connections subject to a GIRA

6.3.2 SS7

If Figure 12 is applied to SS7 traffic, then the term "network connection" can only mean:

- An SCTP association - M2PA or M3UA is transported over SCTP;

and nothing else.

The SCTP association is then equivalent to an M2PA link according to IETF RFC 4165 [10], which is used by the MTP3 to transmit SS7 messages from one node to the next.

If SS7 traffic is interworked at the hub between IP Interconnect interfaces and TDM Interconnect interfaces, then the following protocol stack applies:

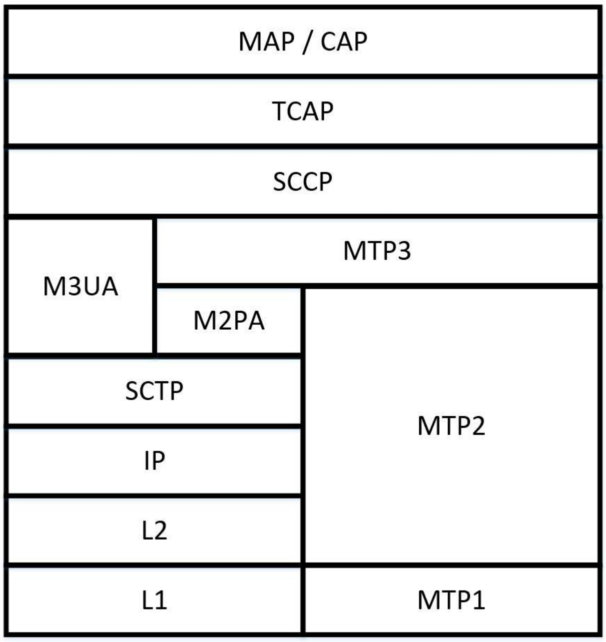


Figure 13: Protocol stack for the interworking of SS7 traffic at the hub

6.4 Interface functionality to signalling interface mapping

6.4.1 CS domain - transport of SIP-I, SCCP/MTP3

6.4.1.1 CS bearer services

The Bearer Services are actually provided by the media interface (see clause 7), however the used RTP sessions shall be setup, torn down and maintained by means of the signalling plane.

The Session Initiation Protocol (SIP) carries Session Description Protocol (SDP) information therefore, which shall be used by the MGWs to add, modify and subtract RTP sessions at their media interfaces.

Clause 7.4.1.1 describes the details, about which standards shall be obeyed, when using SDP for the management of the RTP sessions and when the speech and clearmode data are encoded/decoded to/from the RTP payload.

6.4.1.2 Teleservices

6.4.1.2.0 General

EIRENE SRS [1] specifies in clause 2.2.1, table 2-1, which GSM Teleservices have to be supported in GSM-R networks. The headings of the sub-clauses of clause 5.1.2, of clause 6.4.1.2 and of clause 7.4.1.2 have been setup according to the rows of that table.

6.4.1.2.1 Speech (telephony, emergency calls)

6.4.1.2.1.1 Roaming and handover

MAP and CAP traffic shall be routed by the STP functions of the hub or of the GSM-R Core networks via E2E paths or via UL/DL paths, in order to support mobility management (Location Update, Insert Subscriber Data, Prepare Handover, etc.) and other services during roaming scenarios (USSD, Follow Me, etc.).

General information about MAP can be found in ETSI TS 129 002 [5]. Clause 6.3.2 defines how MAP is routed via the IP Interconnect interfaces.

6.4.1.2.1.2 Call setup, maintenance and teardown

Recommendation ITU-T Q.1912.5 [3], profile C - describes the transport of basic call control information in SIP Messages with embedded ISUP messages. Furthermore it describes the interworking at the I-IWU (incoming SIP call) and at the O-IWU (outgoing SIP call) with BICC or ISUP.

Clauses 6 and 7 of [3] shall be applied at the I-IWU and at the O-IWU, respectively, when basic call control information is interworked between BICC/ISUP and SIP-I at the edge of the GSM-R Core network or at the Call Server Function of the hub.

The "Additional Rules for EIRENE Numbers in SIP-I messages" (clause 6.4.1.2.1.5), shall be applied to the CDN, CGN, orig. CDN and GN (addl. CGN) parameters, as applicable. These rules take precedence over Recommendation ITU-T Q.1912.5 [3] and ETSI TS 129 164 [8].

As required by clause 5.1.1, a GSM-R Core network operator or the operator of a hub may decide to implement a media inactivity detection at his side of the IP Interconnect interfaces.

Such Media Inactivity Detection may be supported by the MGW and by the IWU, according to Recommendation ITU-T H.248.40 [i.3].

6.4.1.2.1.3 SIP preconditions

Recommendation ITU-T Q.1912.5 [3], profile C - describes the interworking of SIP precondition procedures with ISDN continuity check procedures.

Clauses 6 and 7 of [3] shall be applied at the I-IWU and at the O-IWU, respectively, when continuity check procedures are interworked with SIP precondition procedures between BICC/ISUP and SIP-I, respectively, at the edge of the GSM-R Core network or at the Call Server Function of the hub.

6.4.1.2.1.4 SIP overlap signalling

Recommendation ITU-T Q.1912.5 [3], profile C - recommends the usage of SIP en bloc signalling at SIP-I based interfaces. However, it also describes the interworking of overlap signaling between SIP and BICC/ISUP.

Clauses 6 and 7 of [3] shall be applied at the I-IWU and at the O-IWU, respectively, when overlap signaling is interworked between BICC/ISUP and SIP-I, at the edge of the GSM-R Core network or at the Call Server Function of the hub.

6.4.1.2.1.5 Additional rules for EIRENE numbers in SIP-I messages

According to EIRENE SRS [1], the following BICC/ISUP address parameters can convey numbers according to the EIRENE numbering plan at the network-to-network interface:

- The BICC/ISUP CDN (interworked with SIP Request URI and/or To header) can contain:
  - an international E.164 number (NPI = E.164, NOA = INTL); or
  - an international EIRENE number with a "900" prefix (NPI = E.164, NOA = UNKNOWN).
- The BICC/ISUP CGN (interworked with SIP P-Asserted-Identity header) can contain:
  - an international E.164 number (NPI = E.164, NOA = INTL); or
  - an international EIRENE number with a "900" prefix (NPI = E.164, NOA = UNKNOWN).
- The BICC/ISUP GN (addl. CGN, interworked with From header) can contain:
  - an international E.164 number (NPI = E.164, NOA = INTL); or
  - an international EIRENE number with a "900" prefix (NPI = E.164, NOA = UNKNOWN).

- Also, an BICC/ISUP originally called number (orig. CDN) can contain:
  - an international E.164 number (NPI = E.164, NOA = INTL) or
  - an international EIRENE number with a "900" prefix (NPI = E.164, NOA = UNKNOWN) or
  - the "originally dialled digits" (any digits) (NPI = E.164, NOA = UNKNOWN).

This requirement is coped with by the following recommended changes with respect to Recommendation ITU-T Q.1912.5 [3].

- If the address in the outgoing embedded ISUP message at the IWU is not of NOA = INTL (e.g. of NOA = UNKNOWN), then the SIP/TEL URI in the mapped SIP header in the outgoing SIP message is built according to local number format (i.e. omitting the leading '+', but including the '900' prefix, if present).
- If the SIP/TEL URI in the incoming SIP message is of international format (if there is a leading '+'), then the NOA of the address in the forwarded BICC/ISUP message is set to INTL, else the NOA is taken from the BICC/ISUP message that is embedded in the SIP message body.

NOTE: These rules apply only, if the BICC/ISUP address parameter is actually interworked with a SIP header at an IP Interconnect interface.

6.4.1.2.2 Short Message Service

MAP traffic shall be routed by the STP functions either via E2E paths or via UL paths and DL paths, between GSM-R Core network A and GSM-R Core network B, in order to support Short Message Service (Interworking SMS MT with SMS MO).

General information about MAP can be found in ETSI TS 129 002 [5]. Clause 6.3.2 defines how MAP is routed at the IP Interconnect interfaces.

6.4.1.2.3 Facsimile transmission

N/A.

6.4.1.2.4 Voice Group Service (VGCS, VBS)

Transport of inband DTMF information shall be supported at the media interfaces of the IP Interconnect interfaces, as described in clause 7.4.1.2.4.

If the interface partners agree, then the SIP INFO method may be used at the signalling interfaces of the IP Interconnect interfaces, according to the syntax defined at ETSI TS 103 389 [i.1], clause 6.4.11, with following changes and clarifications.

The sender of the SIP INFO message shall add the Info-Package SIP header containing the info package name etsi.groupcall.control, in order to ease compatibility with possible future SIP INFO packages.

IETF RFC 6086 [i.2] should be used as a guideline to prepare future usage of the SIP INFO Package mechanism.

The presence of the body parameter "action" is syntactically optional at the IP Interconnect interface. It should be added by the sender, if the information is available at the sending IWU and if usage of SIP INFO has been agreed upon.

The presence of the body parameters "sequence", "tone-length" and "tone-pause" is syntactically optional at the IP Interconnect interface, too, but they shall be added anyway, if usage of SIP INFO has been agreed upon.

NOTE: See informative annex C for some words about an informational IETF RFC that should be written to register the etsi.groupcall.control package at the IANA.

Details about interworking with the network internal group call control and with the TDM Interconnect interfaces are specified in clause 7.4.1.2.4.

6.4.1.3 GSM supplementary services

6.4.1.3.0 General

EIRENE SRS [1] specifies in clause 2.4.1, table 2-3, which GSM Supplementary services are mandatory for GSM-R networks and which are not. The headings of the sub-clauses of clause 5.1.3, of clause 6.4.1.3 and of clause 7.4.1.3 have been setup according to the rows of that table.

6.4.1.3.1 CLIP / CLIR

Recommendation ITU-T Q.1912.5 [3] describes the transport of CLIP / CLIR information in SIP INVITE messages (with embedded ISUP IAM).

Appendix B.1 and clause 6.1.3.6 of [3], profile C - shall be applied at the O-IWU and at the I-IWU, when CLIP / CLIR information is interworked between BICC/ISUP and SIP-I at the edge of the GSM-R Core network or at the Call Server Function of the Hub.

The "Additional Rules for EIRENE Numbers in SIP-I messages" (clause 6.4.1.2.1.5), shall be applied to the CGN BICC/ISUP parameter and to the P-Asserted-Identity SIP header.

6.4.1.3.2 CoLP / CoLR

Recommendation ITU-T Q.1912.5 [3] describes the transport of CoLP / CoLR information in SIP INVITE messages (with embedded ISUP IAM) and SIP 200 OK messages (with embedded ISUP CON or ISUP ANM).

Appendix B.2 of [3], profile C - shall be applied at the O-IWU and at the I-IWU, when CoLP / CoLR information is interworked between BICC/ISUP and SIP-I at the edge of the GSM-R Core network or at the Call Server Function of the hub.

6.4.1.3.3 Call Forwarding (CFU, CFB, CFNRy, CFNRc)

Recommendation ITU-T Q.1912.5 [3] describes the transport of Call Forwarding information in SIP INVITE messages (with embedded ISUP IAM) and SIP 183 Session Progress messages (with embedded ISUP CPG).

Appendix B.6 of [3], profile C - shall be applied at the O-IWU and at the I-IWU, when Call Forwarding information is interworked between BICC/ISUP and SIP-I at the edge of the GSM-R Core network or at the Call Server Function of the hub.

6.4.1.3.4 Call Waiting (CW)

Recommendation ITU-T Q.1912.5 [3] describes the transport of Call Waiting information in SIP 180 Ringing messages and SIP 183 Session Progress messages (with embedded ISUP ACM or ISUP CPG), in addition to an optional call waiting announcement, which can be transported via the media interface within the stream of early media.

Appendix B.8 of [3], profile C - shall be applied at the O-IWU and at the I-IWU, when Call Waiting information is interworked between BICC/ISUP and SIP-I at the edge of the GSM-R Core network or at the Call Server Function of the hub.

6.4.1.3.5 Call hold (HOLD)

Recommendation ITU-T Q.1912.5 [3] describes the transport of Call Hold information in SIP INVITE messages and SIP UPDATE messages (with embedded ISUP CPG messages), in addition to optional call hold announcements, which can be transported via the media interface within the stream of media.

Appendix B.9 of [3], profile C - shall be applied at the O-IWU and at the I-IWU, when Call Forwarding information is interworked between BICC/ISUP and SIP-I at the edge of the GSM-R Core network or at the Call Server Function of the hub.

6.4.1.3.6 Multi Party service (MPTY)

Recommendation ITU-T Q.1912.5 [3] describes the transport of Manual Conferencing information in SIP messages (with embedded ISUP CPG messages).

Appendices B.12 and B.13 of [3], profile C - shall be applied at the O-IWU and at the I-IWU, when Manual Conferencing information is interworked between BICC/ISUP and SIP-I at the edge of the GSM-R Core network or at the Call Server Function of the hub.

6.4.1.3.7 Closed User Group (CUG)

Recommendation ITU-T Q.1912.5 [3] describes the transport of CUG information in SIP INVITE messages (with embedded ISUP IAM messages).

Appendix B.14 of [3], profile C - shall be applied at the O-IWU and at the I-IWU, when CUG information is interworked between BICC/ISUP and SIP-I at the edge of the GSM-R Core network or at the Call Server Function of the hub.

6.4.1.3.8 Advice of Charge (Information) (AoCI)

N/A.

6.4.1.3.9 Advice of Charge (Charging) (AoCC)

N/A.

6.4.1.3.10 Barring (BAOC, BOIC, BOIC-exHC, BAIC, BIC-Roam)

N/A.

6.4.1.3.11 Unstructured Supplementary Service Data (USSD)

During roaming scenarios, the USSD, e.g. for Follow Me service, may traverse the IP Interconnect interface.

NOTE: The routing of SCCP and MTP3 traffic at the IP Interconnect interface is already defined by clause 6.4.1.2.1.1.

6.4.1.3.12 Follow me

During roaming scenarios, the USSD for Follow Me service, may traverse the IP Interconnect interface.

NOTE: The routing of SCCP and MTP3 traffic at the IP Interconnect interface is already defined by clause 6.4.1.2.1.1.

6.4.1.3.13 Sub-addressing

Recommendation ITU-T Q.1912.5 [3] describes the transport of called party sub-addresses and calling party sub-addresses in SIP INVITE messages and SIP 200 OK messages (with embedded ISUP IAM and ISUP ANM messages, respectively).

Appendix B.5 of [3] shall be applied at the O-IWU and at the I-IWU, when called-party sub-addresses or calling-party sub-addresses are interworked between BICC/ISUP and SIP-I at the edge of the GSM-R Core network or at the Call Server Function of the hub.

6.4.1.3.14 eMLPP

Recommendation ITU-T Q.1912.5 [3] describes the transport of the precedence parameter in SIP INVITE messages (with embedded ISUP IAM messages).

Appendix B.15 of [3], profile C - shall be applied at the O-IWU and at the I-IWU, when MLPP parameters are interworked between BICC/ISUP and SIP-I at the edge of the GSM-R Core network or at the Call Server Function of the hub.

NOTE: Only the MLPP parameters need to be interworked at the IP Interconnect interface, the actual pre-emption can take place within one of the GSM-R Core networks, if applicable.

6.4.1.3.15 ECT

Recommendation ITU-T Q.1912.5 [3] describes the transport of ECT notifications in SIP UPDATE messages (with embedded ISUP FAC messages).

Appendix B.7 of [3], profile C - shall be applied at the O-IWU and at the I-IWU, when ECT notifications are interworked between BICC/ISUP and SIP-I at the edge of the GSM-R Core network or at the Call Server Function of the hub.

6.4.1.3.16 Completion of Calls to Busy Subscribers (CCBS)

Recommendation ITU-T Q.1912.5 [3] describes the transport of CCBS information in SIP INVITE, SIP 180 Ringing and SIP 486 Busy Here messages (with embedded ISUP IAM, ISUP ACM, ISUP CPG and ISUP REL messages).

Appendix B.10 of [3], profile C - shall be applied at the O-IWU and at the I-IWU, when CCBS information is interworked between BICC/ISUP and SIP-I at the edge of the GSM-R Core network or at the Call Server Function of the hub.

6.4.1.3.17 User-to-User Signalling (UUS1)

Recommendation ITU-T Q.1912.5 [3] describes the transport of User to User information in SIP INVITE, SIP 180 Ringing, SIP 200 OK and SIP BYE messages (with embedded ISUP IAM, ISUP ACM, ISUP CPG, ISUP ANM and ISUP REL messages).

Appendix B.19 of [3], profile C - shall be applied at the O-IWU and at the I-IWU, when User to User information is interworked between BICC/ISUP and SIP-I at the edge of the GSM-R Core network or at the Call Server Function of the hub.

6.4.1.4 Railway specific services

6.4.1.4.0 General

EIRENE SRS [1] specifies in clause 2.5.1, table 2-4, which Railway Specific Services are supported in GSM-R networks. The headings of the sub-clauses of clause 5.1.4, of clause 6.4.1.4 and of clause 7.4.1.4 have been setup according to the rows of that table.

6.4.1.4.1 Functional addressing

Functional addressing is transported at the IP Interconnect interface as a combination of GSM Teleservices and Supplementary Services (see clause 5.1.4.1 to find the services that have to be supported).

6.4.1.4.2 Location dependent addressing

Location Dependent addressing is transported at the IP Interconnect interface as a combination of GSM Teleservices and Supplementary Services (see clause 5.1.4.2 to find the services that have to be supported).

6.4.1.4.3 Shunting mode

N/A.

6.4.1.4.4 Multiple driver communications

Multiple Driver Communications is transported at the IP Interconnect interface as a combination of GSM Teleservices and Supplementary Services (see clause 5.1.4.4 to find the services that have to be supported).

6.4.1.4.5 REC / eREC

REC and eREC are transported at the IP Interconnect interface as a combination of GSM Teleservices and Supplementary Services (see clause 5.1.4.5 to find the services that have to be supported).

6.4.1.5 IP/TDM interworking for the CS domain (Use Case II)

6.4.1.5.1 Interworking SIP-I with ISUP, interworking SS7

The details of the interworking of SIP-I with ISUP at the hub in case of Use Case II have been described as an integral part of clauses 6.4.1.1 to 6.4.1.4.

The details of interworking SS7 between TDM based transport and IP based transport have been described as an integral part of clause 6.3.2.

6.4.1.5.2 Interworking the bearer services

N/A.

7 Media interface

7.0 General description

The content of clause 6.0 applies to clause 7.0.

7.1 Network layer protocol

Clause 6.1 specifies the common properties of the network protocol IP, which is used at the media interface and at the signaling interface.

The content of clause 6.1 applies to clause 7.1.

7.2 Transport layer protocols

Clause 6.2 specifies the common properties of the transport protocol UDP, which is not only used at the media interface, but also at the signaling interface.

The content of clause 6.2 that applies to UDP, applies to clause 7.2.

7.3 Real-Time Protocol / Real-Time Control Protocol

If Figure 12 is applied to RTP/RTCP traffic, then the term "network connection" can only mean:

- An RTP Session;

and nothing else.

Only uni-cast RTP streams shall be supported at the present interface.

If G.711 or Clearmode traffic is interworked at the hub between IP Interconnect interfaces and TDM Interconnect interfaces, then not only the audio streams or data streams shall be interworked, but also the DTMF information shall be interworked. This is described in clause 7.4.1.2.4.

7.4 Interface functionality to media interface mapping

7.4.1 CS domain - transport of RTP, RTCP

7.4.1.1 CS bearer services

The MGWs at the edge of the GSM-R Core networks and at the hub shall receive and transmit voice samples and clearmode data, as well as telephony events using the Real-time Transport Protocol (RTP) as specified by IETF RFC 3550 [6].

The following Codecs shall be supported at the media interface:

- Pulse Code Modulation (PCM) - as specified by Recommendation ITU-T G.711 [12].
  PCM is encoded according to RTP A/V profile IETF RFC 3551 [13].

- 64 kbit/s Transparent Call (Clearmode)
  CLEARMODE is encoded according to IETF RFC 4040 [14]

The bearer services can be supported, when the used transport network is designed accordingly. Some hints are given in Annex B "Quality of Service Framework".

Media Inactivity Detection is described in clause 7.4.1.2.1.2.

7.4.1.2 Teleservices

7.4.1.2.0 General

EIRENE SRS [1] specifies in clause 2.2.1, table 2-1, which GSM Teleservices have to be supported in GSM-R networks. The headings of the sub-clauses of clause 5.1.2, of clause 6.4.1.2 and of clause 7.4.1.2 have been setup according to the rows of that table.

7.4.1.2.1 Speech (telephony, emergency calls)

7.4.1.2.1.1 Roaming and handover

N/A.

7.4.1.2.1.2 Call setup, maintenance and teardown

As required by clause 5.1.1, a GSM-R Core network operator or the operator of a hub may decide to implement a media inactivity detection at the IP Interconnect interfaces.

Such Media Inactivity Detection may be supported by the MGW and by the IWU, according to H.248.40 [i.3].

7.4.1.2.1.3 SIP preconditions

The bearer handling of SIP QoS Preconditions shall be supported according to IETF RFC 3312 [9], if SIP Preconditions are supported at the IP Interconnect interface.

7.4.1.2.1.4 SIP overlap signalling

N/A.

7.4.1.2.1.5 Additional rules for EIRENE numbers in SIP-I messages

N/A.

7.4.1.2.2 Short Message Service

N/A.

7.4.1.2.3 Facsimile transmission

N/A.

7.4.1.2.4 Voice Group Service (VGCS, VBS)

Group Call Control information may be transmitted via the IP Interconnect interface as inband information.

Inband transport of DTMF sequences shall be supported using a separate RTP payload type according to IETF RFC 4733 [7].

Out of band transport of group call control information may be supported, as described in clause 6.4.1.2.4.

The interworking at the edge of GSM-R Core networks or at the hub (in case of Use Case II), shall be done according to the procedures described in ETSI TS 129 164 [8], clause 8.2.2, with following changes and clarifications.

RTP events according IETF RFC 4733 [7] should be used at the sending termination of the MGW for the transport of group call control, if out of band transport shall not be used according to local policy at the sending trunk of the IWU.

The same group call Information shall not be sent twice at the same instance of an IP interconnect interface (e.g. out of band and inband).

7.4.1.3 GSM supplementary services

7.4.1.3.0 General

EIRENE SRS [1] specifies in clause 2.4.1, table 2-3, which GSM Supplementary services are mandatory for GSM-R networks and which are not. The headings of the sub-clauses of clause 5.1.3, of clause 6.4.1.3 and of clause 7.4.1.3 have been setup according to the rows of that table.

7.4.1.3.1 CLIP / CLIR

N/A.

7.4.1.3.2 CoLP / CoLR

N/A.

7.4.1.3.3 Call Forwarding (CFU, CFB, CFNRy, CFNRc)

N/A.

7.4.1.3.4 Call Waiting (CW)

In addition to a call waiting notification, which is transported via the signalling interface (see clause 6.4.1.3.4), the media interface may transport a call waiting announcement or a tone within the stream of early media.

7.4.1.3.5 Call hold (HOLD)

In addition to a call hold/retrieve notification, which is transported via the signalling interface (see clause 6.4.1.3.5), the media interface may transport call hold announcements or a tone within the stream of media.

7.4.1.3.6 Multi Party service (MPTY)

N/A.

7.4.1.3.7 Closed User Group (CUG)

N/A.

7.4.1.3.8 Advice of Charge (Information) (AoCI)

N/A.

7.4.1.3.9 Advice of Charge (Charging) (AoCC)

N/A.

7.4.1.3.10 Barring (BAOC, BOIC, BOIC-exHC, BAIC, BIC-Roam)

N/A.

7.4.1.3.11 Unstructured Supplementary Service Data (USSD)

N/A.

7.4.1.3.12 Follow me

N/A.

7.4.1.3.13 Sub-addressing

N/A.

7.4.1.3.14 eMLPP

N/A.

7.4.1.3.15 ECT

N/A.

7.4.1.3.16 Completion of Calls to Busy Subscribers (CCBS)

N/A.

7.4.1.3.17 User-to-User Signalling (UUS1)

N/A.

7.4.1.4 Railway specific services

7.4.1.4.1 Functional addressing

N/A.

7.4.1.4.2 Location dependent addressing

N/A.

7.4.1.4.3 Shunting mode

N/A.

7.4.1.4.4 Multiple driver communications

N/A.

7.4.1.4.5 REC / eREC

N/A.

7.4.1.5 IP/TDM interworking for the CS domain (Use Case II)

7.4.1.5.1 Interworking SIP-I with ISUP, interworking SS7

N/A.

7.4.1.5.2 Interworking the bearer services

The details of the interworking of RTP/RTCP with 64 kbit/s channels at the hub in case of Use Case II have been described as an integral part of clause 7.3.

Annex A (normative): Locating SIP entities

ETSI TS 103 389 [i.1] has elaborated in its annex A some guidelines for the locating of SIP entities at the interface between FTS and NSS.

Those guidelines help to circumvent the lack of a DNS and shall be applied to the IP Interconnect interfaces, too, with the following changes and clarifications:

1) The reference system architecture from clause 4 of the present document shall be referred to instead of the reference system architecture from ETSI TS 103 389 [i.1].

2) The SIP URI Convention from ETSI TS 103 389 [i.1] does not apply at the IP Interconnect interface.

3) The terms subsystem, NSS and FTS shall be replaced by the terms GSM-R Core Network, GSM-R Core Network A, GSM-R Core network B and hub, respectively.

4) Not all outgoing SIP-I traffic from one network has the same FQDN in the host part of the Request URI.

5) The FQDN in the host part of a Request URI will not necessarily address all SIP-I entities of the target network.

6) On the contrary:

    a) The host part of the Request URI in any SIP request at an IP Interconnect interface may address one IWU of the destination network, via one or more transport addresses.

    b) The host part of the Request URI in any SIP out-of-dialog request at an IP interconnect interface may address one SIP Proxy of the destination network, via one or more transport addresses.

    c) OPTIONS requests should address exactly one transport address, if they are used for OPTIONS pinging.

7) The deployment scenarios A.1, A.2 and A.3 are "valid" scenarios at the IP Interconnect interfaces, too, but they should not be considered "typical" scenarios.

Annex B (informative): Quality of service framework

Annex B of ETSI TS 103 389 [i.1] applies, with following changes and clarifications:

- The terms NSS, FTS and network operator is replaced by GSM-R Core network operator, GSM-R Core network A, GSM-R Core network B and hub, respectively.

Annex C (informative): Group call control - additional information

Annex E of ETSI TS 103 389 [i.1] defines some possible scenarios of group call control.

As indicated by clause 6.4.1.2.4, an informational IETF RFC should be written to officially register the INFO Package etsi.groupcall.control at the IANA.

Citing IETF RFC 6086 [i.2]:

```
11.4.  Creation of the Info Packages Registry

   IANA created the following registry under "Session Initiation
   Protocol (SIP) Parameters":

   Info Packages

      Note to the reviewer:

      The policy for review of Info Packages is "Specification
      Required", as defined in [RFC5226 [i.7]]. This policy was selected
      because Info Packages re-use an existing mechanism for transport
      of arbitrary session-associated data within SIP; therefore, new
      Info Packages do not require the more extensive review required by
      specifications that make fundamental protocol changes. However,
      the reviewer is expected to verify that each Info Package
      registration is in fact consistent with this definition.  Changes
      to the SIP protocol and state machine are outside of the allowable
      scope for an Info Package and are governed by other procedures
      including IETF RFC 5727 [i.8] and its successors, if any.

   The following data elements populate the Info Packages Registry.

   o  Info Package Name: The Info Package Name is a case-sensitive
      token. In addition, IANA shall not register multiple Info Package
      names that have identical case-insensitive values.

   o  Reference: A reference to a specification that describes the Info
      Package.

   The initial population of this table shall be:

   Name         Reference
```

Chapter 10 "Info Package Requirements" of IETF RFC 6086 [i.2] explains in detail how to officially create an INFO package at IANA.

NOTE: IETF RFC 6086 [i.2] is still valid and in status "proposed standard" at the IETF, as by 2019-10-23.

History

Document history		
V1.1.1	December 2020	Publication