SIP-I Interworking Description

Version 1.2

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1 Introduction

This document describes how the SIP-I (SIP with encapsulated ISUP) is utilized in the IPX (IP Exchange) environment.

There is growing interest from different Service Providers to deploy SIP-I based voice interworking between Service Providers using SIP-I Nodes such as 3GPP specified MSC-S (MSC-Server) in an IP based architecture instead of the current TDM based solutions. Therefore there is a need to for common recommendations ensuring that interworking with SIP-I takes place smoothly.

Purpose of the document is to illustrate a generic SIP-I profile to be used for the Packet Voice Interworking over the IPX between different mobile and fixed Service Providers, aiming to limit the number of interoperability issues caused by different implementation and deployment solutions used. Thus the features needed for a successful interoperability of SIP-I Node & MGW (Media Gateway) network elements being used in multi-operator, multi-vendor end-to-end environment.

The intended audience of this document are the different Service Providers, IPX Providers and vendors intending to deploy the Packet Voice Interworking using SIP-I.

This document is based on what was experimented as to be achievable during various trials performed within GSMA projects.

1.1 References

Ref	Doc Number	Title
[1]	ITU-T Q.1912.5	Interworking between Session Initiation Protocol (SIP) and Bearer Independent Call Control protocol or ISDN User Part
[2]	GSMA IR.34	Inter-Service Provider IP Backbone Guidelines
[3]	GSMA IR.40	Guidelines for IPv4 Addressing and AS Numbering for GRX/IPX Network Infrastructure and User Terminals
[4]	GSMA IR.77	Inter-Operator IP Backbone Security Requirements For Service Providers and Inter-operator IP backbone Providers
[5]	3GPP TS 26.103	Speech codec list for GSM and UMTS
[6]	3GPP TR 29.802 (G)MSC-S - (G)MSC-S	(G)MSC-S - (G)MSC-S Nc Interface based on the SIP-I protocol
[7]	3GPP TS 23.231	SIP-I based circuit-switched core network; Stage 2
[8]	3GPP TS 29.231	Application of SIP-I Protocols to Circuit Switched (CS) core network architecture; Stage 3
[9]	3GPP TS 29.235	Interworking between SIP-I based circuit-switched core network and other networks
[10]	3GPP TS 29.163	Interworking between the IP Multimedia (IM) Core Network (CN) subsystem and Circuit Switched (CS) networks

Ref	Doc Number	Title
[11]	3GPP TS 29.164	Interworking between the 3GPP CS Domain with BICC or ISUP as Signalling Protocol and external SIP I Networks

In addition a number of related RFCs have been produced in IETF. They are referred to in the following chapters when applicable.

1.2 Abbreviations

Term	Description
DS	Circuit Switched
DTMF	Dual Tone Multi-Frequency
FNO	Fixed Network Operator
GRX	GPRS Roaming Exchange
IMS	IP Multimedia Subsystem
IPX	IP Exchange
ISUP	ISDN User Part
MGW	Media Gateway
MNO	Mobile Network Operator
MSC-S	Mobile Switching Service Center Server
PS	Packet Switched
PVI	Packet Voice Interworking
SCL	Service Codec List
SCTP	Stream Control Transmission Protocol
SIP	Session Initiation Protocol
SIP-I	SIP with encapsulated ISUP
SIP-I Node	SIP-I end-point (for example MSC-S or Softswitch)
TCP	Transmission Control Protocol
TDM	Time-Division Multiplexing
UDP	User Datagram Protocol

2 Scope

2.1 In Scope

This document presents the technical principles and guidelines needed for the transport of packet switched traffic between SIP-I capable operators over the IPX network using a SIP-I based interface. SIP-I call scenarios are based on the Q.1912.5 Profile C. Some minor deviations from Profile C are introduced as described in Chapter 10.

2.2 Out of Scope

Out of scope for this particular document are general issues not directly related to SIP-I. For example standards compliant SIP-I Nodes are prerequisite for Packet Voice Interworking, but they are not detailed as such in this document.

Out of scope for this particular release of the document is full compliancy with 3GPP Rel-8 SIP-I on Nc work item which is expected to be complete at the of end 2008 (TS 23.231, 29.231, 29.235). However, compliancy to 3GPP Technical Specifications is the long term target.

Out of scope are intra-operator related issues, thus this document concentrates only on inter-operator scenarios. I.e. usage of NNI is always part of discussions here.

Details of charging and billing are out of scope for any IREG documentation and can be found in BARG documents.

For information about the overall environment (such as routing, DNS usage and IPX network in general) used for Packet Voice Interworking please see other GSMA documentation such as IR.34, IR.40, IR.65, IR.77 and other publicly available documentation on:

http://www.gsma.com/newsroom/gsmadocuments/

Also out of scope for this release of document are:

- Other services/applications than SIP-I based voice, video, fax and modem (including any IMS based applications)
- Interworking with TDM/PSTN
- Interworking with BICC and any other protocols such as SIP-T
- Interworking with IMS (including IMS SIP based voice)
- Commercial issues
- Testing & certification issues
- Back-office functions (e.g. O&M)
- Speech quality issues
- Border Control Functions

3 Overview

ITU-T has produced specification Q.1912.5 Interworking between Session Initiation Protocol (SIP) and Bearer Independent Call Control protocol or ISDN User Part. Profile C of this specification documents how information from the ISUP message is mapped into the SIP headers and encapsulated into the SDP media description part (and vice versa). In practise this allows the SIP session initiation protocol carry ISUP messages across IP based networks. MNOs wishing to evolve from the traditional CS based MSC architecture to IP based 3GPP Rel-4 MSC-Server architecture, as well as FNOs deploying softswitches, can utilize these encapsulation mechanisms to ensure that all the ISUP related information remains available when utilizing SIP-I as the signalling protocol.

Figure 1 below shows an example of the general architecture of Packet Voice Interworking based on Profile C between two MNOs. Signalling (using SIP-I between MSC-Server nodes) and media (using RTP between Media Gateway nodes) are carried over the IPX network using IP based NNI, while the circuit switched UNI can be still utilized as with traditional MSC architecture. Thus there is no need to modify the existing UNI in order to allow the inter-operator connection evolve from TDM to IP.

In addition to voice, SIP-I & RTP based connection can be used to transport video telephony, modem and fax traffic.

Whilst the Figure 1 shows traffic traversing the IPX Proxy, it might or might not be actually involved in NNI. For the avoidance of a doubt, it should be noted that IPX Proxy is not a mandatory component in Packet Voice Interworking per se.

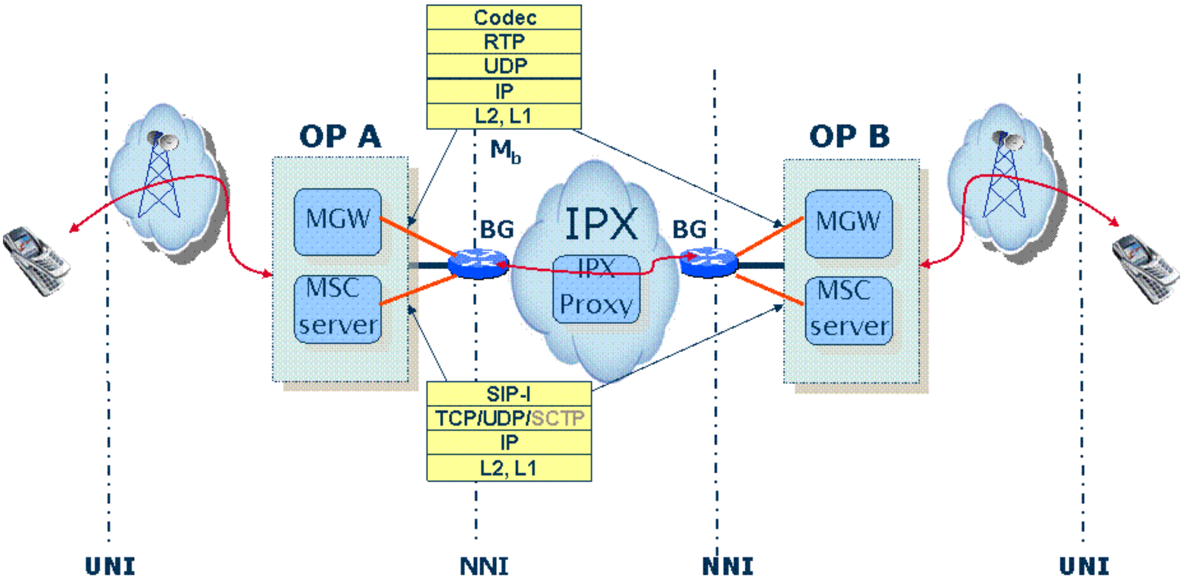

Figure 1: Example of Normal Inter-Operator Call Scenario

IPv4 will be used for the connectivity between SIP-I Nodes and MGWs, even though the possible introduction of IPv6 at some point in time is not excluded.

From the end-user point of view system described in this document allows customers to make calls based on Q.1912.5 Profile C using different voice codecs, use DTMF tones and supplementary services (such as call hold, call forward and multiparty).

Following chapters describe the actual technical details of SIP-I interworking.

4 Voice call

Requirements for signalling level:

- SDP offer/answer mechanism according to Chapter 8
- No support needed for Comfort Noise (IETF RFC 3389)
- No support needed for initial or subsequent INVITE without SDP offer

Requirements for user plane level:

- ITU-T G.711 codec RTP (PCMU or PCMA) payload format as defined in IETF 3551. Decision on using PCMU or PCMA is a part of the general codec neotiation phase (see Chapter 8)
- NB-AMR Two modes sets shall be supported in FR-AMR: set 0 (12.2, 7.4, 5.9, 475 kbps) and set 7 (12.2 kbps). This allows to negotiation of the mode set for AMR codecs.

  - WB-AMR
  - G.729 (For mobile-to-fixed or fixed-to-fixed scenarios only)
  - GSM EFR
  - Packetization period: 20ms. Other packetization period value is allowed but strongly discouraged. Use of ptime attribute in SDP to advertise the used packetization period value is encouraged. If no ptime is given, 20ms is the default value
  - Single sample / RTP packet

G.711 is a mandatory codec to be supported. Other codecs as listed here are optional but still recommended in order to maximize the possibility of transcoder-free operations with SIP-I interworking.

Any required echo cancellation remains the responsibility of the service provider as there is no echo cancellation performed within the IPX network.

5 video telephony call

Requirements for signalling level:

- SDP offer/answer procedure resulting CLEARMODE pseudo-codec (IETF RFC4040)
- Encapsulated ISUP to contain ISDN USI parameter having UI1LP encoded as "H.223&H.245". Other possibility is to dedicate individual MSISDN for BS30 (synchronous, transparent bearer service) and use that as dialled number

Requirements for user plane level:

- CLEARMODE pseudo-codec (IETF RFC4040).
- Packetization period: 20ms. Other packetization period value is allowed but strongly discouraged. Use of ptime attribute in SDP to advertise the used packetization period value is encouraged. If no ptime is given, 20ms is the default value
- Single sample / RTP packet

6 analogue modem call

Requirements for signalling level:

- Fax/modem calls require that SDP offer/answer procedure results a single ITU-T G.711 codec. If the type of the call cannot be identified at the originating end (possible a PSTN gateway) the G.711 codec shall be included along with possible compressed codecs. If the type of the call is identified to be a fax call in the gateway/terminating mobile switch only G.711 shall be returned in the SDP answer. If the type of the call still cannot be identified (due to e.g. possible diversions) compressed codec may be selected however in this case in-band fax/modem tone detection is required in speech phase to be able to identify the type of call and a mid-call codec change to G.711 is required upon detected fax/modem tone in order the call to be successful.

- No support needed for V.152 or V.150.1

Requirements for user plane level:

- ITU-T G.711 codec RTP payload format as defined in IETF 3551.
- Packetization period: 20ms. Other packetization period value is allowed but strongly discouraged. Use of ptime attribute in SDP to advertise the used packetization period value is encouraged. If no ptime is given, 20ms is the default value
- Single sample / RTP packet

7 Analogue facsimile call

Requirements for signalling level:

- Fax/modem calls require that SDP offer/answer procedure results a single ITU-T G.711 codec. If the type of the call cannot be identified at the originating end (possible a PSTN gateway) the G.711 codec shall be included along with possible compressed codecs. If the type of the call is identified to be a fax call in the gateway/terminating mobile switch only G.711 shall be returned in the SDP answer. If the type of the call still cannot be identified (due to e.g. possible diversions) compressed codec may be selected however in this case in-band fax/modem tone detection is required in speech phase to be able to identify the type of call and a mid-call codec change to G.711 is required upon detected fax/modem tone in order the call to be successful.
- No support needed for V.152 or T.38

Requirements for user plane level:

- ITU-T G.711 codec RTP payload format as defined in IETF 3551
- Packetization period: 20ms. Other packetization period value is allowed but strongly discouraged. Use of ptime attribute in SDP to advertise the used packetization period value is encouraged
- Single sample / RTP packet

8 codec negotiation

It is beneficial for both end points in a SIP-I network to be aware of available codecs from the other side, e.g. to minimise time needed for codec modification during handover to another radio access.

One of the major goals for Packet Voice Interworking is to minimize the need to perform transcoding when SIP-I is used. Thus the SIP-I Nodes can negotiate a common codec.

SDP offer/answer mechanisms are also used for DTMF, see Chapter 11 for further details.

The following codec negotiation scenarios shall be supported:

Originating SIP-I Node:

- Single offer/answer scheme is used, when the first answer contains only one codec (selected codec SC):
- Send SDP-offer (SCL1)
  Receive SDP-answer (SC)
- Second offer/answer is used, when the first answer contains multiple codecs (Supported Codec List SCL):
- Send SDP-offer (SCL1)
  Receive SDP-answer (SCL2)
- Send SDP-offer (SC)
  Receive SDP-answer (SC)

The second scenario corresponds to the case when the terminating SIP-I Node sends multiple codecs to the originating SIP-I Node. This is an optional behaviour for the terminating SIP-I Node.

Terminating SIP-I Node:

- Offer with codec list received
  Receive SDP-offer (SCL1)
  Send SDP-answer (SCL2)
  Receive SDP-offer (SC)
  Send SDP-answer (SC)

TrFO can work without multiple offer/answer even though it may benefit from it. Second offer/answer allows for both originating and terminating SIP-I Node to know each others codec capabilities thus in some scenarios allowing for a better/quicker codec selection in case of mid-call compressed codec to compressed codec change.

In-call codec change to G.711 shall always be possible to allow for in-band fax/modem scenarios.

Codecs of "auxiliary" payload type, i.e. RTP Telephony Event payload type (IETF RFC 2833/4733) or the comfort noise codec (IETF RFC 3389) shall not be considered as a factor when deciding single/multiple codec case as the only deciding factor is the number of speech-codecs.

G.711 shall always be present in the initial SDP offer, as last in the list. This is to ensure maximal interoperability with non-mobile service providers. FNOs are authorized to offer G.729 codec, provided that G.711 is also offered as a last chance for handling codec negotiation successfully.

Section 7 of TS 26.103 lists the supported 3GPP speech codecs for a SIP-I based CS core network.

9 ISUP version

ISUP according to ITU-T Q.761-Q.764 -97 is mandatory to be used. At least ISUP according to ITU-T Q.761-Q.764 -93 "White book" should be used to have support for the defined supplementary services. Use of any national ISUP flavor is not supported for international use of the IPX. Use of national ISUP flavor for national IPX usage is per default not supported in Service Hub (i.e. multilateral) Model, except if specific commercial agreement between SP & IPX hub Provider.

SIP part: all concerned headers with subscriber numbers (i.e. Request URI, From, to, P-asserted Identity fields) use format "+CCabcdefgh...". This is aligned with TS 29.163.

ISUP part: NoA (Nature of Address) International format shall be used in all relevant fields. That convention also applies for national use of IPX except if national regulations enforce another way to proceed for national use of IPX.

10 sip profile definition

Minimum set of features used in SIP-I signalling:

- Support for 100rel/PRACK is required (IETF RFC 3262)
  Note: optional in Q.1912.5 Profile C
- Support for UPDATE is required (depending on use case, e.g. to test mid-call codec negotiation) (IETF RFC 3311)
- Support for INFO method (IETF RFC 2976) is required (used for ISUP messages that does not cause state changes such as notifications)
- Support for SIP session timers (session expires header) is optional (IETF RFC 4028)
  Note: not covered by Q.1912.5 Profile C
- Support for privacy mechanism (IETF RFC 3323) and privacy header (Appendix C.2 in Q.1912.5) is required (use with CLIP)
- Support for OPTIONS method between SIP-I Nodes is recommended. Any SIP-I Node shall reply to incoming SIP OPTIONS request. OPTIONS messages shall not be blocked anywhere in IPX network
- Also the following RFCs are supported as defined in Q.1912.5 (note: latest RFCs listed here)

  - RFC 2046 (MIME Media Types)
  - RFC 4566 (SDP)
  - RFC 3966 (tel URI). Note: also SIP URI can be used for addressing
  - RFC 3204 (MIME media types for ISUP and QSIG Objects)
  - RFC 3261 (SIP)
  - RFC 3264 (Offer/Answer Model with SDP)
  - RFC 3326 (Reason Header Field for SIP)

The basic SIP as defined by IETF RFC 3261 does not include a "keep alive" mechanism. As such, it is possible that one end of a session may fail and be unable to signal the release of the session. Therefore it is recommended that SIP-I Nodes support the SIP Session Timer as specified in IETF RFC 4028 as a means to determine whether a SIP session is still active by attempting to perform a session refresh, and therefore as a means to know when resources may be released if one end of the session fails. During call origination each SIP-Node negotiates the use of the SIP Session Timer. It is recommended to that session refresh request is an UPDATE request without SDP.

A SIP-I Node not supporting the SIP Session Timer shall accept incoming UPDATE requests without SDP in order to allow the remote node to use the SIP Session Timer. In addition a SIP-I Node not supporting the SIP Session Timer must provide other mechanisms to detect that a session has failed, which must not impact any IPX node or remote SIP-I Node.

Generation of RTCP packets is recommended but not mandatory. In any case RTCP shall not be blocked anywhere in the IPX network.

11 DTMF interworking

DTMF signalling via the RTP telephony-event according to IETF RFC 4733 shall be supported for compressed codecs and is recommended for PCM. Inband DTMF signalling (generation, detection) shall also be supported to ensure open interoperability between two SIP-I Nodes from different vendors, when terminating SIP-I Node selects PCM and does not offer RTP telephony-event in the answer.

SIP-I Node initiating an SDP offer shall include the MIME type "telephone events" with default events in the first SDP offer, if it offers any codecs other than the default PCM codec. SIP-I Node initiating an SDP offer with only the default PCM codec as speech codec may include the MIME type "telephone events" in the first SDP offer.

SIP-I Node terminating an SDP offer shall accept the MIME type "telephone events" with default events in any SDP answer, if it selects any codec other than the default PCM codec. SIP-I Node terminating an SDP offer that selects the default PCM codec as speech codec may include the MIME type "telephone events" in the SDP answer, if present in the SDP offer.

- When G.711 is used as speech codec then DTMF transferred as in-band within G.711 coded speech
- RFC 4733 shall be used with all compressed codecs
- RTP payload according to IETF RFC 4733
- INFO method shall not be used for this purpose (because not standardised)
- If the SIP-I Node has negotiated DTMF events, then the SIP-I Node is not required to handle in parallel DTMF inband in G.711

12 transport protocol

UDP/TCP is used for Packet Voice Interworking, at least in the first phase. Details of utilizing SCTP for the purpose of SIP-I are still under discussion in 3GPP. TCP is used when SIP signalling message size exceeds 1300 octets (for requests, UDP responses can carry up-to 1500 octets)

The following rules shall be applied for a SIP-I application using TCP:

- TCP is used to establish a SIP session if INVITE does not fit into previously described size limits
  An existing TCP connection can be re-used for any other transaction
- A TCP connection shall be released by the SIP-I node that established the transaction
- A TCP connection shall be kept open as long as it serves an open SIP transaction
- In case of a lost TCP connection, the Via header shall specify where replies are to be addressed to if the TCP connection is terminated prior to all replies being successfully sent

- The SIP-I node which initiates a TCP connection shall use an ephemeral port number for the local port of the connection and the server port number 5060 for the remote port (unless the remote site indicates a different port number to be used instead of the well-known SIP port number 5060)

13 supplementary services

The following supplementary services are included:

- Calling Line Identity Presentation (CLIP)
- Calling Line Identity presentation Restriction (CLIR)
- Call Forwarding Busy (CFB)
- Call Forwarding No Reply (CFNR)
- Call Forwarding Unconditional (CFU)
- Call Waiting (CW)
- Call Hold (CH) - according to Q.1912.5, no RTCP
- Call Forward (CF)
- Explicit Call Transfer (ECT)
- Terminal Portability - this service cannot be verified in mobile-to-mobile call scenarios
- Connected Line Presentation/Restriction
- User to User Service
- Multiparty

14 Announcement and ringback tone

Only ISUP codes should be used for the purpose of announcements as stated in Q.1912.5 (SIP codes not correctly mapped in all cases, so SIP codes to be used only as a last resort).

Generation of ringback tone (whether provided by originating or terminating SIP-I Node) is for further study, based on e.g. operator input regarding how this feature is handled at the moment in the existing TDM based inter-.operator voice call environment.

15 Cause codes

Cause codes within a SIP-I message shall not conflict with each other. These cause codes are:

1. ISUP cause code
2. Reason header
3. SIP response code

If there is a conflict between Cause information of different protocol elements then the priority of usage shall be according to the above listed order.

> Note: ISUP cause codes shall be transported within and between networks end-to-end. If an ISUP cause code cannot be transferred within a network (e.g. IMS), then the Reason header shall be used to transport the ISUP cause code. If SIP response codes are used, mapping between SIP and ISUP response codes/causes shall follow Q.1912.5.

Annex A Document Management

A.1 Document History

Version	Date	Brief Description of Change	Approval Authority	Editor / Company
1.0	17/02/09	New PRD IR.83.	IREG#55 EMC	Tero Jalkanen (TeliaSonera Finland Oyj)
1.1	27/04/09	Incorporated Minor CR01 (PACKET Doc 39_013 Rev1)	PWP#39	Tero Jalkanen (TeliaSonera Finland Oyj)
1.2	8/06/09	Incorporated Minor CR02 (PACKET Doc 39_024 Rev2)	Networks Group	Tero Jalkanen (TeliaSonera Finland Oyj)

A.2 Other Information

Type	Description
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Editor / Company	Tero Jalkanen / TeliaSonera

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