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Foreword

This Technical Report (TR) has been produced by ETSI Technical Committee Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN).

1 Scope

The present document provides guidance for handling IMS-based NGN national IP interconnection details.

Interconnection of NGNs represents a broad area of study including such areas as signalling interconnection, media interconnection, routing, address resolution, security, charging and network management. However, as with most aspects of the NGN, there are many Standards Development Organizations (SDOs) and industry bodies that are working on this topic.

As organizations such as ITU-T, 3GPP, GSMA and the i3 Forum do focus on overarching IP interconnection specifications and agreements, a gap on guidelines for the handling of **national** IP interconnection was identified. The present document is intended to fill this gap for IMS based NGN national IP interconnection in co-relation with the IMS approach of 3GPP.

2 References

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

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

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

2.1 Normative references

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

Not applicable.

2.2 Informative references

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

[i.1] ETSI TS 122 228: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); Service requirements for the Internet Protocol (IP) multimedia core network subsystem (IMS); Stage 1 (3GPP TS 22.228)".

[i.2] ETSI TS 129 165: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Inter-IMS Network to Network Interface (NNI) (3GPP TS 29.165)".

[i.3] ETSI TS 184 011: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); Requirements and usage of E.164 numbers in NGN and NGCN".

[i.4] IETF RFC 3966 (December 2004): "The tel URI for Telephone Numbers".

[i.5] IETF RFC 4694 (October 2006): "Number Portability Parameters for the "tel" URI".

[i.6] ITU-T Recommendation Q.1912.5: "Interworking between Session Initiation Protocol (SIP) and Bearer Independent Call Control protocol or ISDN User Part".

[i.7] ETSI EN 383 001: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); Interworking between Session Initiation Protocol (SIP) and Bearer Independent Call Control (BICC) Protocol or ISDN User Part (ISUP) [ITU-T Recommendation Q.1912.5, modified]".

[i.8] IETF RFC 4904 (June 2007): "Representing Trunk Groups in tel/sip Uniform Resource Identifiers (URIs)".

[i.9] IETF RFC 6044 (October 2011): "Mapping and Interworking of Diversion Information between Diversion and History-Info Headers in the Session Initiation Protocol (SIP)".

[i.10] ETSI TS 124 229: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; IP multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP); Stage 3 (3GPP TS 24.229)".

[i.11] ETSI TS 129 163: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Interworking between the IP Multimedia (IM) Core Network (CN) subsystem and Circuit Switched (CS) networks (3GPP TS 29.163)".

[i.12] IETF RFC 4244 (November 2005): "An Extension to the Session Initiation Protocol (SIP) for Request History Information".

[i.13] IETF RFC 5806 (March 2010): "Diversion Indication in SIP".

[i.14] ETSI TS 129 235: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Interworking between SIP-I based circuit-switched core network and other networks (3GPP TS 29.235)".

[i.15] IETF RFC 3482 (September 2003): "Number Portability in the Global Switched Telephone Network (GSTN): An Overview".

[i.16] ITU-T Recommendation Q.769.1: "Signalling system No. 7 - ISDN user part enhancements for the support of number portability". .

[i.17] ETSI TR 184 003: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); Portability of telephone numbers between operators for Next Generation Networks (NGNs)".

[i.18] EC Mandate M/493: "Standardisation Mandate to the European Standards Organisations (ESO) in support of the location enhanced emergency call service".

3 Abbreviations

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

CC	Country Code
CIC	Carrier Identification Code
CN	Core Network
CS	Circuit Switched
CS-IBCF	CS (domain) IBCF
ENUM	Electronic NUMber
GSMA	Global System for Mobile communications Association
IBCF	Interconnection Border Control Function
IM	IP Multimedia
IMS	IP Multimedia Subsystem
INAP	Intelligent Network Application Part
IP	Internet Protocol
ISUP	ISDN User Part
IWU	InterWorking Unit
MGCF	Media Gateway Control Function
NDC	National Destination Code
NGN	Next Generation Network

NNI	Network to Network Interface
NP	Number Portability
P-A-Id	P-Asserted-Identity
PES	PSTN Emulation Subsystem
PSTN	Public Switched Telephone Network
QoR	Query on Release
SDO	Standards Development Organization
SDP	Session Description Protocol
SIP	Session Initiation Protocol
SIP-I	SIP with encapsulated ISUP
SN	Subscriber Number
TDM	Time Division Multiplexing
TEL	Telephony
UMTS	Universal Mobile Telecommunications System
URI	Uniform Resource Identifier
VoIP	Voice over IP

4 National IP Interconnection

It is noted that work on IP Interconnect is ongoing in other standardisation bodies like 3GPP and in organisations such as the GSMA and the i3 Forum. The activities in those SDOs primarily deal with international interconnection. Hence the information provided in the present document provides guidance on items of national IP interconnection which are not covered elsewhere.

For the subject of IP Interconnection in general (not exclusive to national IP Interconnection only) guidance is given in the following 3GPP and ETSI deliverables:

- TS 122 228: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); Service requirements for the Internet Protocol (IP) multimedia core network subsystem (IMS); Stage 1 (3GPP TS 22.228)" [i.1].
- TS 124 229: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; IP multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP); Stage 3 (3GPP TS 24.229)" [i.10].
- TS 129 165: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); LTE; Inter-IMS Network to Network Interface (NNI) (3GPP TS 29.165)" [i.2].

In TS 124 229 [i.10] stage 3 specification text can be found on general interconnection issues like:

- URIs and address assignment in clause 4.2.
- Routing principles in clause 4.3.
- Trust domains in clause 4.4.
- Additional routeing capabilities in support of transit and interconnection traffics in IM CN subsystem in Annex I.

4.1 Calling Identity

National IP Interconnection interfaces may be required to support the transfer in SIP of the Calling Identity between operators to meet national regulation matters (e.g. lawful intercept) with the following principles:

- The IP Interconnect between national operators should be considered to be a "trusted" environment.
- In SIP the P-Asserted-Identity (P-A-Id) header field should be used to convey the Calling Identity between operator networks.
- If the Calling Identity is a telephone number it should be conveyed either in the international format or in national format based upon national decisions.

- For redirected calls the History-Info header field should be used to convey the redirection information. Alternatively, the Diversion header field can be used based on bilateral agreements. See clause 4.6 for further details.

No missing elements were identified in the SIP standards to support this national IP Interconnect item.

4.2 Number Formats

Based on TS 129 165 [i.2] clause 8 and TS 184 011 [i.3] the following 2 basic practices are given as guidance on national NGN IP Interconnect interfaces to support E.164 number formats in the SIP Request URI and other SIP header fields like To, From, History-Information, Diversion and P-Asserted-Identity:

1) **Use of global number format only**
Unless bilateral agreement exist, a global number coding, as defined in RFC 3966 [i.4], should be used in a TEL URI and in the user portion of a SIP URI with the user=phone parameter. In addition, use of one or more specific National Destination Codes (NDCs) may be used in front of national-only numbers like short codes (112, 114, etc.) and service codes when no overlap with E.164 number ranges exists. In this arrangement the number coding in SIP header fields is the global number as defined in RFC 3966 [i.4] always starting with +<CC>, which implies that there is no need for the inclusion of the phone-context parameter to header fields.

2) **Use of local number format next to global number format**
If bilateral agreements exist between operators to transfer national E.164 numbers and national-only E.164 numbers in the local number coding as defined in RFC 3966 [i.4], then the phone-context parameter should be added to SIP header fields which transfer a local number in a TEL URI and in the user portion of a SIP URI with the user=phone parameter. Header fields without phone-context parameter should be in the global format and should be conveyed as such (see practice 1 above).

NOTE 1: Transfer of national numbers in local format between operators implies that the local number digits are set to <NDC><SN> (without prefix '0') and the phone-context is set to +<CC> for the transfer of the global number digits. For the transfer of short codes / service codes a bilaterally (nationally) agreed coding of the phone context parameter should be used.

NOTE 2: The use of SIP and/or SIP-I for national IP Interconnect is a bilateral (national) matter. With SIP additional transfer capabilities in specific parameters may be supported like the rn parameter according RFC 4694 [i.5] for the transfer of additional Number Portability information for which the interworking to/from ISUP is specified in and TS 129 163 [i.11]. With SIP-I such additional capabilities are not described in ITU-T Recommendation Q.1912.5 [i.6] and in EN 383 001 [i.7].

No missing elements or new capabilities were identified in the SIP standards to support this national IP Interconnect item.

4.3 Prefix Digits

Within operator domains, and sometimes based on bilateral agreements between operators on the interconnections, prefix digits may be used as extensions to numbers to indicate specific routing arrangements and/or indications.

Some issues/use cases that could involve prefix digits include:

1) At least during a transition period interconnect between 2 operators could have PSTN and IP in parallel. How determine which "route" to take in order to avoid unnecessary PSTN-IP conversion?

2) In current PSTN interconnect routing prefixes often point to the ingress point of the destination network (usually based on geographic location of the destination). How to reflect that in SIP?

3) Monitoring tools are crucial to operations and an end-to-end view of call is necessary. With different protocols in play, correlation of different protocol messages belonging to the same call is very important and the transfer of prefix digits may assist in such an end-to-end correlation process.

4) For support of Carrier Selection as an alternative to the parameters defined in RFC 4904 [i.8]) and RFC 4694 [i.5].

5) For support of Number Portability in some countries as an alternative to the rn parameter defined in RFC 4694 [i.5].

For this purpose prefix digits may be passed through the MGCF in either way: i.e. from SIP to ISUP for calls from VoIP to PSTN and from ISUP to SIP for calls from PSTN to VoIP. This will require manipulations of the digit strings in both the Request URI and the ISUP Called Party Number parameter in either direction. The prefix digits may function to convey in a technology independent way specific information on a call-by-call basis. This of course implies that these prefix digits are firstly to be added and subsequently to be deleted as part of e.g. an ENUM trigger for the user part of the SIP Request URI or an INAP trigger for the ISUP Called Party Number parameter. In the VoIP domain the prefix digits may be used to make necessary translations to e.g. the technology specific TEL URI tags tgrp and trunk-context (RFC 4904 [i.8]), and with SIP URI encoded form of a TEL URI, for the identification of incoming/outgoing routes.

No missing elements were identified in the SIP standards to support this national IP Interconnect item.

4.4 Carrier Selection

This refers to the question how to transfer calls initiated by a PSTN user with Carrier Selection over a national IP Interconnect interface?. In this respect different mechanisms may be envisaged, e.g.:

- Use of specific trunk groups (identified in SIP by tgrp and trunk-context parameters specified in RFC 4904 [i.8]) with the receiving Carrier Selection operator.
- Identified by the SIP CIC parameter specified in RFC 4694 [i.5], with an agreed coding of the CIC parameter with the receiving Carrier Selection operator.

No specific guidelines were concluded for this national IP Interconnection item.

4.5 Domain name conventions

It is often not enough to know that a particular E.164 number is located in network X, but also it may be needed to know which of the ingress points to network X to use (assuming multiple ingress points).

For this national IP Interconnection item no definitions are proposed for domain name conventions and/or other (more elegant) solutions to reflect a network ingress point in the domain name.

4.6 Diversion header and History-Info header

Though the History-Info header field is the only normative way of SIP working in the IP Interconnection standards for the transfer of redirection information for redirected calls, it is noted that the Diversion header field is out in the field in both implementations and networks. As a result parties should be prepared to the situation that not all networks are able to support History-Info header field (for sending and/or receipt) at the start of national IP Interconnect. For national IP Interconnect the following aspects are given for considerations:

- there is an actual issue because networks may not all immediately be able to support History-Info header field while networks will support Diversion header field for a certain period of time; and
- absence of an end-to-end interworking solution for redirection information will likely cause unwanted, end-user initiated session loops when two users (un)willingly direct sessions to each other from different networks where end-to-end traversal of diversion information cannot be guaranteed; and
- RFC 6044 [i.9] provides the most appropriate direction to the interworking between the History-Info header field and Diversion header field.

As a consequence, for national IP Interconnect the following set of guiding principles are given for the transfer of redirection information for redirected calls.

1) **History-Info header field**
The only standardised way for IP Interconnection to transfer information of redirected calls as specified in TS 124 229 [i.10] and TS 129 163 [i.11] and, as a logical consequence, should also followed as the guiding principle for national IP Interconnect. This will imply that networks should normalize to and/or from History-Info header field in those cases where internally use is made of other SIP means to transfer information of redirected calls. The History-Info header field is specified in RFC 4244 [i.12].

2) **Diversion header field**
If parties agree on a bilaterally basis, use of the Diversion header field may be used as an alternative way of operation to transfer information of redirected calls. Then such parties should agree whether also the History-Info header field may be used. The Diversion header field is specified in RFC 5806 [i.13].

3) **Interworking**
If parties mutually agree to make use of the Diversion header field, the normalization to and/or from History-Info header field should be performed as specified in RFC 6044 [i.9].

4) **SIP-I**
The use of SIP and/or SIP-I for national IP Interconnect is a bilateral (national) matter. With SIP additional transfer capabilities in specific parameters may be supported like the History-Info header field in RFC 4244 [i.12] for the transfer of call forwarding information for which the interworking to/from ISUP is specified in TS 129 163 [i.2]. With SIP-I such additional capabilities are not described in ITU-T Recommendation Q.1912.5 [i.6], TS 129 235 [i.14] and in EN 383 001 [i.7] and should not be used on SIP-I interfaces.

No missing elements were identified in the SIP standards to support this national IP Interconnect item.

4.7 SIP to/from SIP-I interworking

The basic model for national IP Interconnect with interworking of SIP to/from SIP-I is depicted in Figure 1, whereby the usage of the IBCF is restricted to the ingress/egress of SIP relations and the usage of the MGCF is restricted to the ingress/egress of SIP-I and ISUP relations.

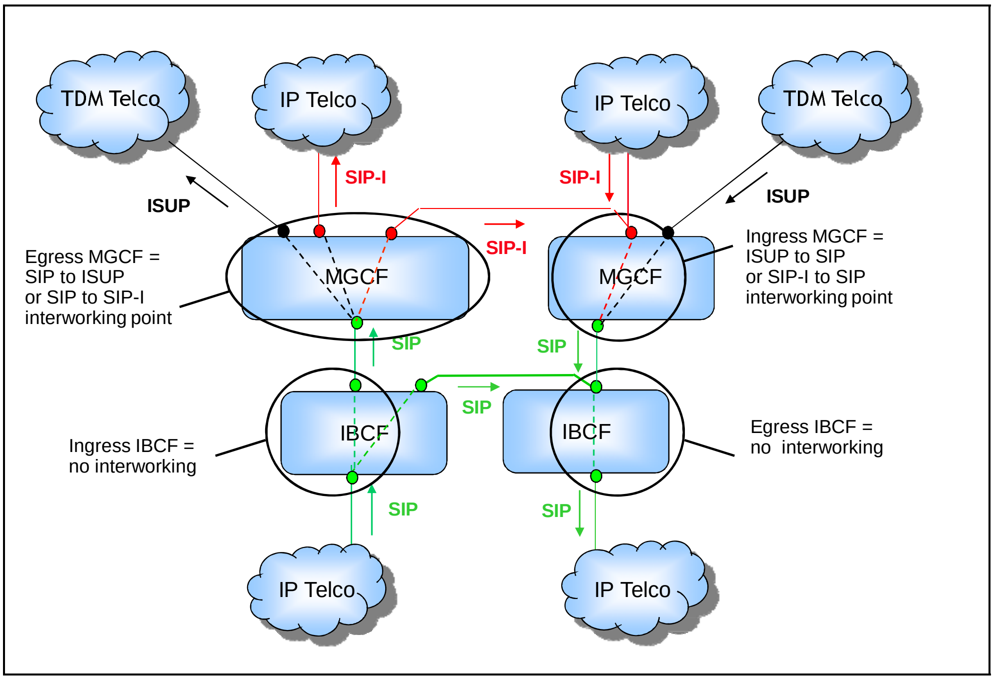

Figure 1: Basic transit model

Alternatively the IBCF may be co-located with an MGCF and/or a CS-IBCF (as defined in TS 129 235 [i.14], Annex A) to create a single physical entity that support both SIP and SIP-I relations. It should be noted that a physical entity embedding both an IBCF and a CS-IBCF does not have to support a CS call model or to terminate ISUP procedures.

Figure 2 illustrates an alternative architecture where the IBCF is co-located with a CS-IBCF and the MGCF is co-located with the InterWorking Unit (IWU) defined in TS 129 235 [i.14]. The MGCF with co-located IWU is used for both the SIP-I to ISUP and the SIP-I to SIP-I interworking situations. SIP-I to SIP-I interworking, can also be handled by the CS-IBCF to save MGCF resources if the type of interworking (including functions like billing and SIP screening as described in TS 129 235 [i.14], Annex A) can be determined at the ingress point (i.e. IBCF/CS-IBCF) and no actions need to be performed on the encapsulated ISUP body.

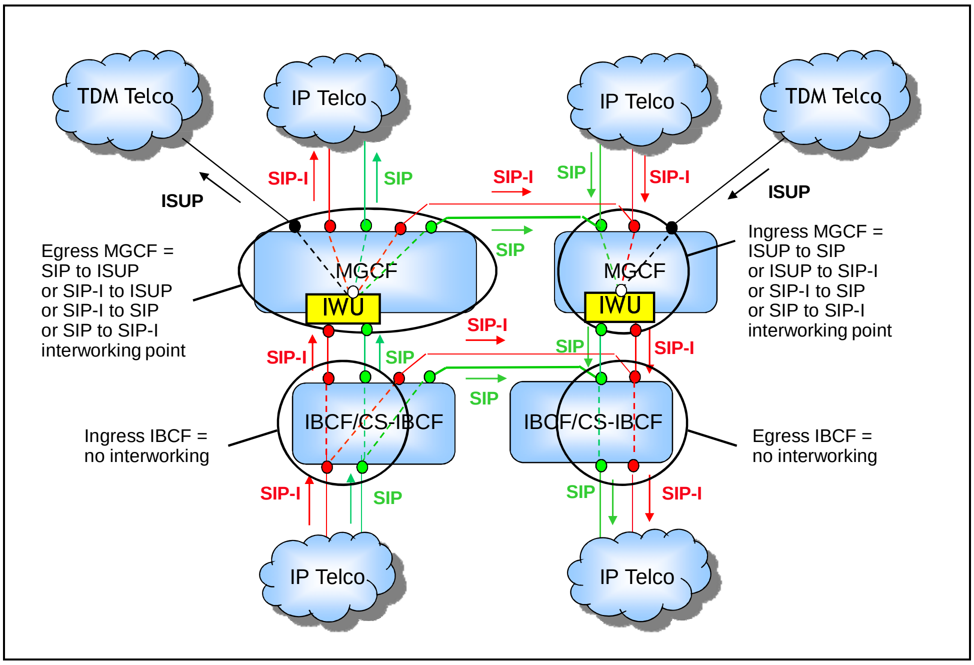

Figure 2: Alternative transit model

Remarks:

- The SIP to SIP relation may also be valid for the MGCF if non-IMS scenarios are considered as well.

NOTE: In this context non-IMS scenarios typically refer to interconnection with SoftSwitch based networks. More details can be found in TS 122 228 [i.1] and TS 124 229 [i.10].

- The objective of the SIP to/from SIP-I transit model is to achieve guidelines independent of the technology of the IP Telco network whether the last being an IMS based network, a PES network, a SoftSwitch network or else.

- The interworking between SIP-I and SIP is specified in TS 129 235 [i.14] wherein for a transit scenario the role of the originating network is played collectively by the originating network and the transit network.

- For the interworking to ISUP please refer to TS 129 163 [i.11] for NGNs using IMS. Alternatively if interworking to the SIP profile(s) outlined within EN 383 001 [i.7] then use of the present document is recommended. EN 383 001 [i.7] is a modified endorsement of ITU-T Recommendation Q.1912.5 [i.6] whilst TS 129 163 [i.11] is originally based on Q.1912.5 [i.6] Profile A but have since evolved as IMS has developed.

No missing elements were identified in the SIP standards to support this national IP Interconnect item.

4.8 Number Portability

For the support of Number Portability (NP) different scenarios may be envisaged as outlined in TR 184 003 [i.17] and RFC 3482 [i.15]. For the support in SIP these NP scenarios may be accompanied with parameters like "npdi", "rn" and "rn-context" as defined in RFC 4694 [i.5].

For the interworking between SIP and ISUP mapping between the NP information in ISUP (as defined in ITU-T Recommendation Q.769.1 [i.16]) and SIP is specified in TS 129 163 [i.11]. With SIP-I such additional capabilities are not described in ITU-T Recommendation Q.1912.5 [i.6] and in EN 383 001 [i.7].

NOTE 1: The observation is made that support of the NP Query on Release (QoR) scenario is not covered (there is an equivalent SIP response code missing for ISUP cause value #14 as defined in ITU-T Recommendation Q.769.1 [i.16]).

NOTE 2: In this context also special attention should be given to situations that may result in double TDM/IP conversions with routings like PSTN-IMS-PSTN or IMS-PSTN-IMS for ported numbers.

4.9 Emergency caller location data conveyance

National IP Interconnection interfaces may be required to support the transfer in SIP of the emergency caller location data between operators to meet national regulation matters for emergency calls. Note that also other protocols may be used for this transport instead. For this subject the outcome of the work in ETSI on EC Mandate M/493 [i.18] can become relevant.

History

Document history		
V3.1.1	April 2012	Publication