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*Technical Specification*

## **Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); Interconnection and Routing requirements related to Numbering and Naming for NGNs; NAR Interconnect**

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**Reference**

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## Foreword

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

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# 1 Scope

The present document analyzes the issues concerned with interconnection and routing in NGN and their implications and requirements for numbering, naming and addressing and related resolution functionalities.

Several different interconnection scenarios are considered, based on TISPAN "NGN Functional Architecture" ([5]), although not all require the use of all architectural functions. The role of the transit network has also been considered in order to evaluate the knowledge of numbers/names served by different operators and related requirements for routing purposes.

Some scenarios may require the availability of infrastructure systems for numbering/naming resolutions, such as infrastructure ENUM, or other database based system for route resolution, and may require coordinated provision by involved operators.

The present document focuses on calls routed between subscribers identified by E.164 numbers, coded through either tel URI and SIP URI formats. It also applies only to the transfer of calls across interconnection points between the home network of the A-Party and the home network of the B-Party. It does not consider in detail interconnections needed to support roaming scenarios. A further issue of the present document will consider interconnect scenarios related to roaming.

The present document is relevant not just to IMS but to any NGN SIP-based interconnection.

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- |     |   |
|-----|---|
| [1] | Void.   |
| [2] | ETSI TS 181 005: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); Services and Capabilities Requirements". |

- [3] ETSI TS 182 012: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); IMS-based PSTN/ISDN Emulation Sub-system (PES); Functional architecture".
- [4] ETSI ES 282 007: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); IP Multimedia Subsystem (IMS); Functional architecture".
- [5] ETSI ES 282 001: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); NGN Functional Architecture".
- [6] ETSI ES 283 018: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); Resource and Admission Control: H.248 Profile for controlling Border Gateway Functions (BGF) in the Resource and Admission Control Subsystem (RACS); Protocol specification".
- [7] ETSI ES 283 003: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); IP Multimedia Call Control Protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP) Stage 3 [3GPP TS 24.229 [Release 7], modified]".
- [8] ETSI TS 123 228: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); IP Multimedia Subsystem (IMS); Stage 2 (3GPP TS 23.228 version 7.5.0 Release 7)".
- [9] ITU-T Recommendation Q.3401: "NGN NNI Signalling Profile (Protocol Set 1)".
- [10] 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]".
- [11] ETSI TS 123 228: "IP Multimedia Subsystem (IMS); Stage 2".
- [12] ETSI TS 124 228: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); Signalling flows for the IP multimedia call control based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP); Stage 3 (3GPP TS 24.228 Release 5)".
- [13] ETSI TS 124 229: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); Internet Protocol (IP) multimedia call control protocol based on Session Initiation Protocol (SIP) and Session Description Protocol (SDP); Stage 3 (3GPP TS 24.229 Release 8)".
- [14] ETSI TS 129 163: "Digital cellular telecommunications system (Phase 2+); Universal Mobile Telecommunications System (UMTS); Interworking between the IP Multimedia (IM) Core Network (CN) subsystem and Circuit Switched (CS) networks (3GPP TS 29.163 Release 7)".
- [15] ETSI TS 184 002: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); Identifiers (IDs) for NGN".
- [16] Void.
- [17] ITU-T Recommendation E.164: "The international public telecommunication numbering plan".
- [18] IETF RFC 3261: "SIP: Session Initiation Protocol".

## 2.2 Informative references

The following referenced documents are not essential to the use of the present document but they assist the user with regard to a particular subject area. For non-specific references, the latest version of the referenced document (including any amendments) applies.

- [i.1] ETSI TR 180 000: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); NGN Terminology".

[i.2] ETSI TR 184 007: "Telecommunications and Internet converged Services and Protocols for Advanced Networking (TISPAN); Naming/Numbering Address Resolution (NAR)".

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## 3 Abbreviations

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

3GPP	3rd Generation Partnership Project
BGCF	Breakout Gateway Control Function
BICC	Bearer-Independent Call Control
CP	Communication Provider
DNS	Domain Name System
ENUM	E.164 telephone NUmber Mapping
GRX	GRPS Roaming eXchange
GSMA	GSM Association
IBCF	Interconnection Border Control Function
I-CSCF	Interrogating-Call/Session Control Function
I-ENUM	Infrastructure ENUM
IETF	Internet Engineering Task Force
IMS	IP Multimedia System
IPX	IP Packet eXchange
ISUP	ISDN Signalling User Part
IWF	InterWorking Function
MGCF	Media Gateway Control Function
NAR	Naming/Numbering Addressing Resolution
NGN	Next generation Network
NNI	Network to Network Interface
OPID	Originating Party IDentity
P-CSCF	Proxy-Call/Session Control Function
PES	PSTN Emulation Service
PSTN	Public Switched Telephone Network
RACS	Resource and Admission Control Subsystem
S-CSCF	Serving-Call/Session Control Function
SDP	Session Description Protocol
SIP	Session Initiation Protocol
TPID	Terminating Party IDentity
UA	User Agent
URI	Uniform Resource Identifier

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## 4 Introduction

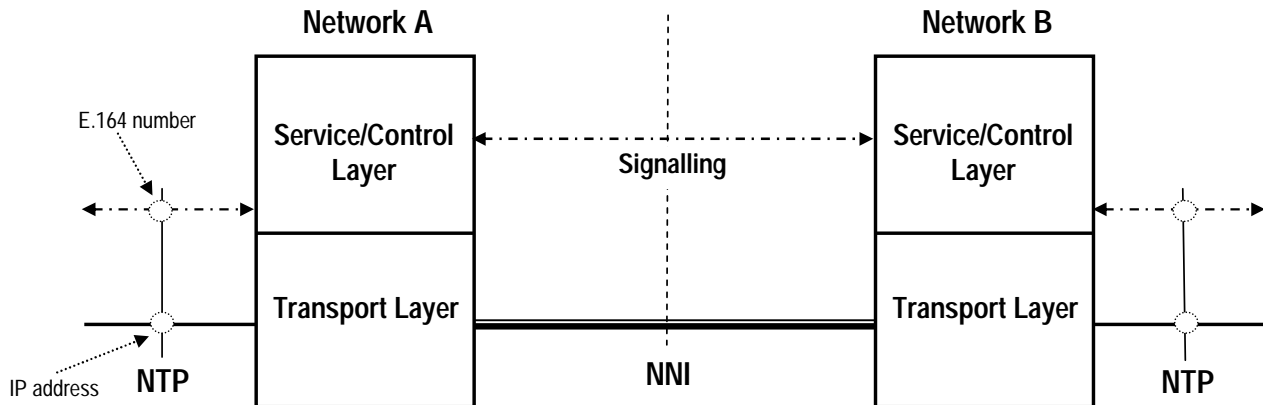
This Technical Specification:

- Defines a Routeing Model for NGN Interconnection.
- Proposes the first set of Requirements and related CP Routeing Roles.
- Defines the resolution process for tel URI and SIP URI.
- Proposes the possible basic structure of NAR [i.2] Interconnect Framework.

## 5 Routing Model for NGN Interconnection

### 5.1 Reference scheme for interconnection

Figure 1 describes the NGN layers and the reference scheme for interconnection in NGN. The routing resolution in NGN includes both the typical IP interworking routing rules that are bounded into the transport layer and the service/control layer ones.



**Figure 1: Reference scheme for Interconnection and routing**

The call control and service control functions take into account service capabilities requirements for routing. Some applications entities (for instance redirect servers and/or location application) may also be required to define the route. For the purpose of interconnection the Call Control function has to resolve routing in order to get the next hop and reach the required destination (the transport layer has to allocate the resource required based on the services characteristics). As a consequence, the NGN routing determination process includes both service and transport functions to define the correct destination and requires an appropriate service transport resources allocation at the interconnection level.

The functionalities used for routing at the transport layer follow the existing IP intrinsic routing mechanisms, and have also to satisfy the NGN service requirements and the transport resource requirements (RACS function [4], [5]). Therefore routing resolution at transport layer is outside the scope of the present document.

The destination number/name and the type of service required are essential to identify the route for service layer routing. The ETSI standards on NGN public identity [15] state that E.164 numbering is the main customer identification scheme. However in addition domain name based identifiers can be also used.

In the case of the SIP protocol, as defined in relevant ETSI standards ([7]), such public identifiers are carried through so-called SIP URI (i.e. SIP URI: <E.164 number>@<home network domain>) and tel URI (i.e. tel URI: <E.164 number>).

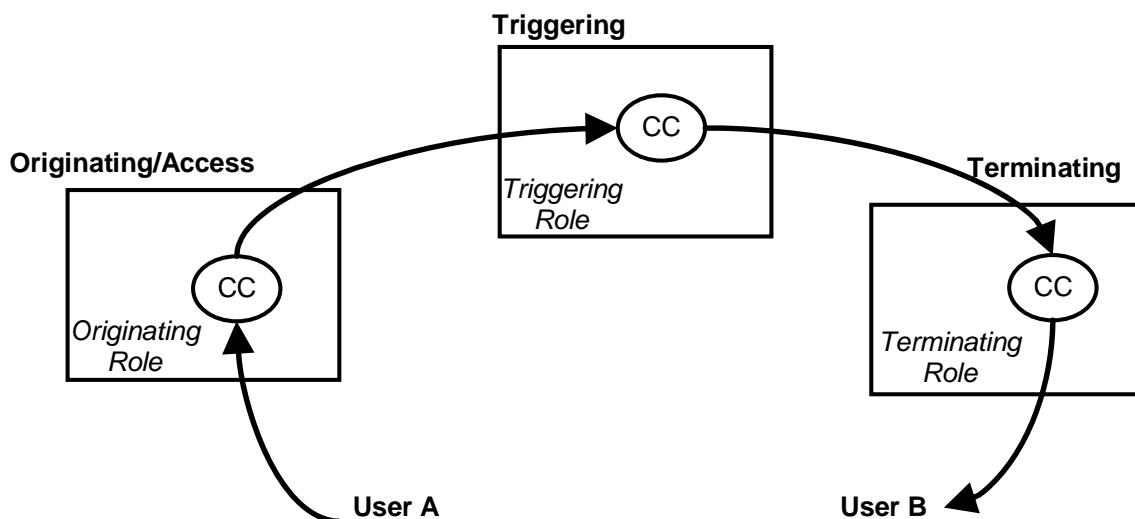
Public identifiers resolution is an essential part of the routing process to determine the next hop network entity for the session setup: this entity can either be identified by an intermediate URI translated into an appropriate IP address using DNS functionality, or directly mapped to an IP address.

### 5.2 Interconnection and Routing

The innovative approach for NGN routing requires identifying a common routing model that guarantees interoperability between networks; besides more networks, provided by different operators, can be involved in session setup and routing process and related decisions depends on its specific role within the session setup process (i.e. originating, transit, triggering, termination or interworking).

The routing model for interconnection should take in account required type of service and existing interconnection bilateral agreements in order to resolve the route.





**Figure 2: Functional framework for routing and CP roles**

Figure 2 identifies the essential Communication Providers' (CP) routing roles for the routing process of a generic session setup established for two users. The user A sends a service request to the NGN call control of the originating CP that has the objectives of processing the request and putting through the communication to other CPs or/and the user B.

An individual CP may assume different roles on defining the route of the session for a service request and it can apply particular routing procedures related also to the knowledge of the service required and the partial or full information on the final destination to reach (i.e. external routing).

The same representation can be also adopted to include the cases where the same CP fulfils all the routing roles for an internal call/session started and terminated on the same CP's domain/domains (i.e. internal routing).

The CP can implement one or more routing roles at the same time, and can also provide these capabilities for other CPs (e.g. in the case of hosting).

A description of the basic routing roles follows:

- **Originating Role:** responsible for the communication service offered to the end customer and basic service request and related call/session handling. It is influenced by the Originating customer profile.
- **Triggering Role:** Responsible for numbering resolution, routing determination toward final destination and the choice of appropriate next hop entity (internal or external) to reach it; additionally it is in charge to identify specific interworking requirements for NGN (IMS or not IMS networks) or for PSTN/ISDN interoperability.
- The numbering resolution process implies the determination of "routable SIP URI" that includes the domain associated to operator which called user is subscribed to. In the case of terminating calls toward legacy networks (i.e. PSTN/ISDN or PLMN) the numbering resolution output can be a tel URI.
- **Terminating Role:** responsible for terminating and serving the session-oriented communication to end user.

### 5.3 Transit capabilities

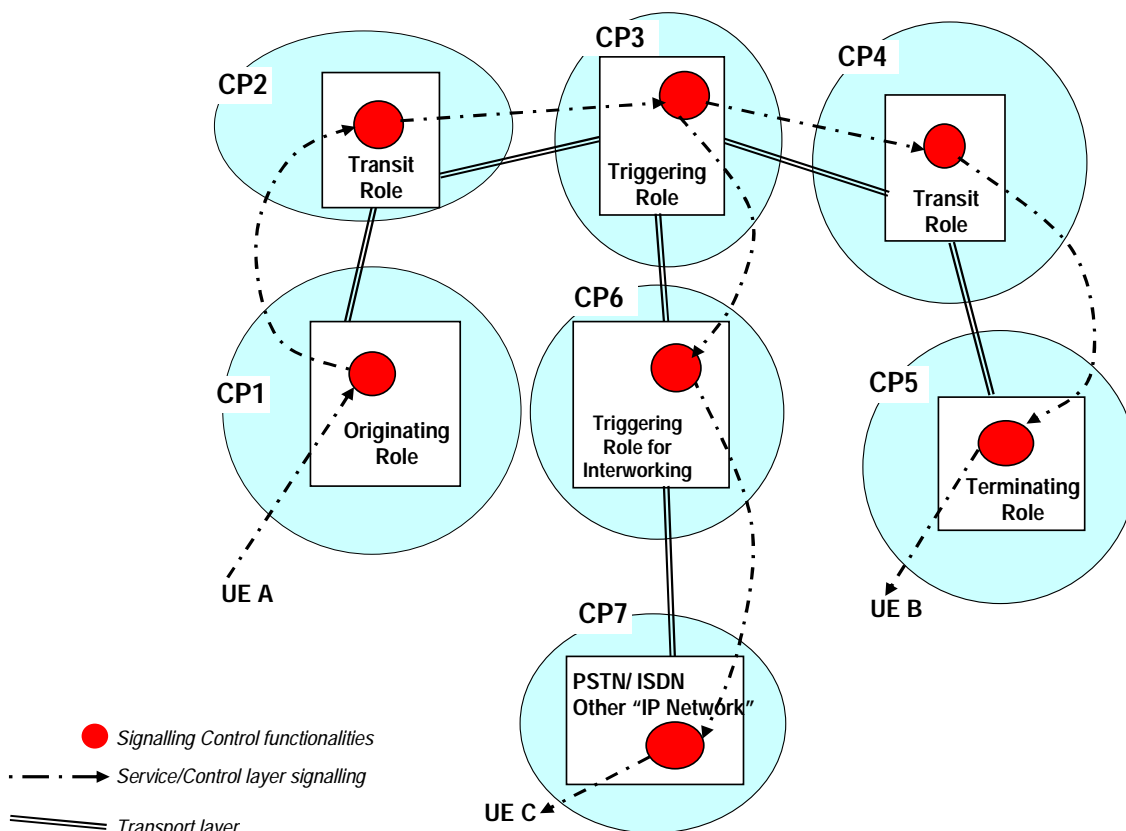
In addition to the previous basic CPs' roles the case where one or more transit networks are involved is relevant.

- **Transit Role:** Route the session-oriented communication request to the next hop CP without any SIP URI or tel URI mapping in routable SIP URI. All the session related information is carried on to next hop transparently.

The routing process for transit is limited to transparently deliver the session setup to the appropriate next hop CP (without numbering resolution or mapping in a routable SIP URI). However the transit routing process has to guarantee appropriate transport resources allocation to preserve service requirements.

## 5.4 Example of complete set of CP's roles

Figure 3 points out an example of a complete set of CP's roles for SoIx Interconnection (see SoIx in the NGN Architecture [5]) that highlights different application of routing model for route determination.



**Figure 3: CPs roles for routing process for SoIx interconnection**

Following is the routing process implementation among different CPs, and the description of guidelines for implementations:

- Originating role, on the basis of service required by session-based originated by Originating user, forwards the session setup request for routing analysis to the Triggering Role entity, which is provided internally or by a different CP on the basis of bilateral agreements.
- Triggering role is per definition responsible for resolving public user identities to determine the next hop outbound route towards the operator to which the destination user is subscribed.
- Transit role is assumed by a CP that, on the basis of bilateral agreements, transparently forwards the session setup request to the appropriate next hop CP.
- Terminating role is responsible for delivering communication service session setup to the destination end user; the numbering/naming analysis and related routing process is limited to internal identification of session control entity associated to end user for required service communication session setup and then to determination of the IP address associated to the user terminal equipment. Where a Routable SIP URI is used, the CP with the terminating role for a service communication session is identified through the domain part of the routable SIP URI.

## 5.5 Routeing process requirements and related CP roles

At least the following parameters and functional elements shall be available as input of NGN Routeing Model (see figure 4):

- Service parameters: set of parameters that represents service requirement for a specific session setup.
- Terminating Party Identity (TPID): it is an E.164 number in tel URI or SIP URI coding format; optionally a domain name identifier can be also used.
- Originating Party Identity (OPID): is an E.164 number in tel URI or SIP URI coding format; optionally a domain name identifier can be also used.
- Performance parameters: That parameter can be taken into account for defining special action on routeing (i.e. bearer resources allocation, requirements on possible route, congestion and/or unavailable resources, etc.).

It is also relevant to include the case of route determination for the case of overlap signalling (see [7]) in which a partial routeing decision can be made before dialling is completed.

At least the following parameters and functional elements shall be available as output of NGN Routeing Model (see figure 4):

- SIP URI or tel URI identifying the destination user.
- Next Hop ID: it identifies the network entity (i.e. border gateway or other signalling/control functionality) that enables or is in charge of reaching next hop CP.
- Transport resource parameters: parameters meaningful for transport addressing and routeing process at transport layer for resource allocation.

The SIP URI that it is obtained through the numbering/naming resolution process is called a routable SIP URI since its domain part identifies the appropriate terminating CP.

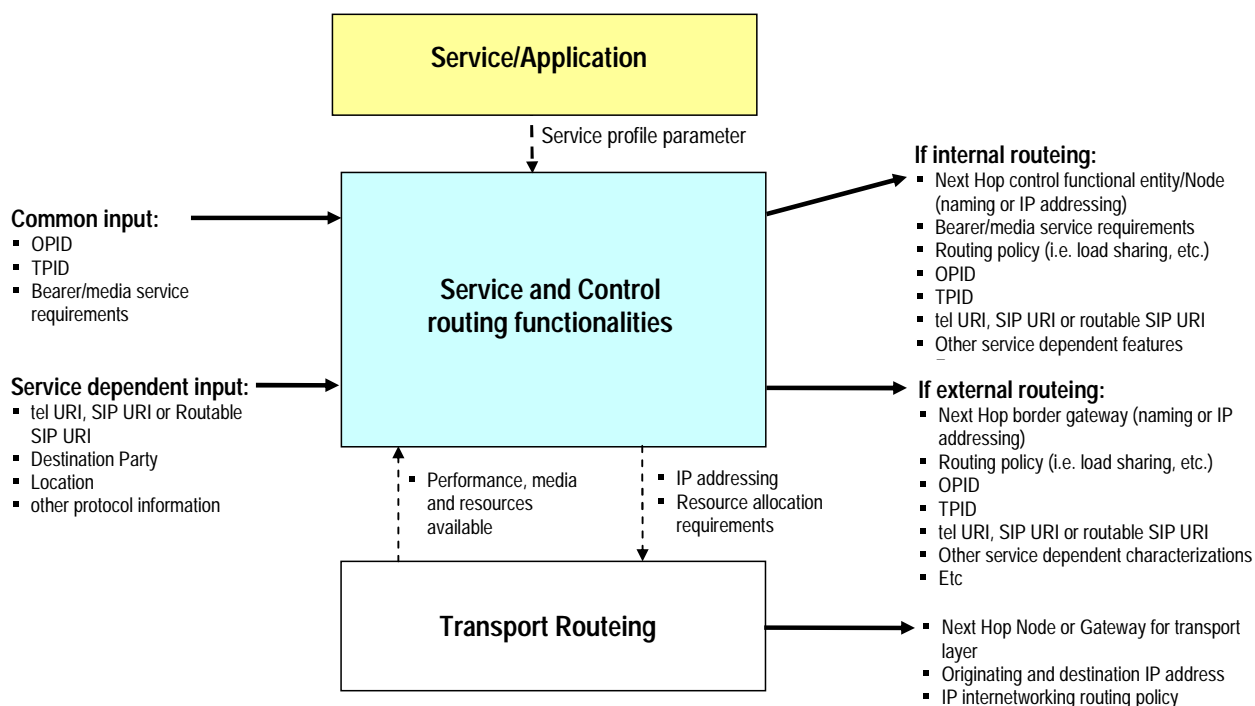


Figure 4: Common general purpose Routeing Model for NGN

Considering figure 4, the common requirements for the routing model are the following:

- Numbering and naming resolution functionality are applied for routable SIP URI determination, based on specific mapping functionalities (database with relation with numbering/naming and destination CPs, etc.).
- The routing process shall depend on the service characteristics and associated parameters that are meaningful for determination of the route to the destination network. The process shall check and validate the service requirements compatibility with user service profile and network resource availability.
- Originating and Terminating Party Public Identities shall be considered the essential information for routing process.
- Other service dependent parameters (for instance location parameters) can be considered if meaningful for routing process application.

Considering the different roles identified for routing in figure 3, the routing model for NGN will be appropriately characterized to apply associated functionalities and input/output parameters.

**Table 1: Requirements and actions for CP roles**

CP roles	Requirement and actions summing up
<b>Originating</b>	TPID is not resolved by CP's call control. Service session setup request has to be forwarded to Transit or Triggering CP.
<b>Triggering</b>	tel URI or SIP URI is resolved in a "routable" SIP URI.
<b>Triggering for interworking</b>	tel URI or SIP URI can not be resolved, since final destination is PSTN/ISDN or PLMN or an "All IP" network not compliant with ETSI NGN specifications. tel URI is forwarded to interworking function to apply specific resolution mechanisms and procedures.
<b>Terminating</b>	The domain part of the routable SIP URI corresponds to the domain of the terminating CP and end user shall be identified.
<b>Transit</b>	Call/session is to be delivered transparently to a terminating CP, basing on a tel URI or a Routable SIP URI domain part different from the transit CP. tel URI or SIP URI, after having identified routing towards terminating CP, is passed on unmodified.

## 5.6 SIP and E.164 numbering

Also at interconnection [i.2] applies and table 2 shows the relationship between the Public ID that identifies a Terminating and Originating user parties and the formats used in SIP header fields. The present document is focused on E.164 numbering in tel URI or SIP URI coding formats.

**Table 2: Public IDs and formats**

	Public ID (User parties)	NGN coding formats
<b>User/Service Identifiers</b>	<b>"User@Domain" format</b>	<b>SIP URI</b>
	<b>International or national E.164 Numbers</b>	<b>tel URI SIP URI</b>

Thus the two formats that may be used are:

- **tel URI for international or national numbers**, e.g. tel: +431505641636.
- **SIP URI**, e.g. sip: +431505641636@<servicing operator>;user=phone.

NOTE: "user=phone" is appended where the user part of the SIP URI is a string that is the same as a tel URI. Additionally, the SIP "phone-context" parameter is used in case of national E.164 numbers and other digit strings that are non-E.164 numbers.

## 5.7 Address Resolution, Routing requirements

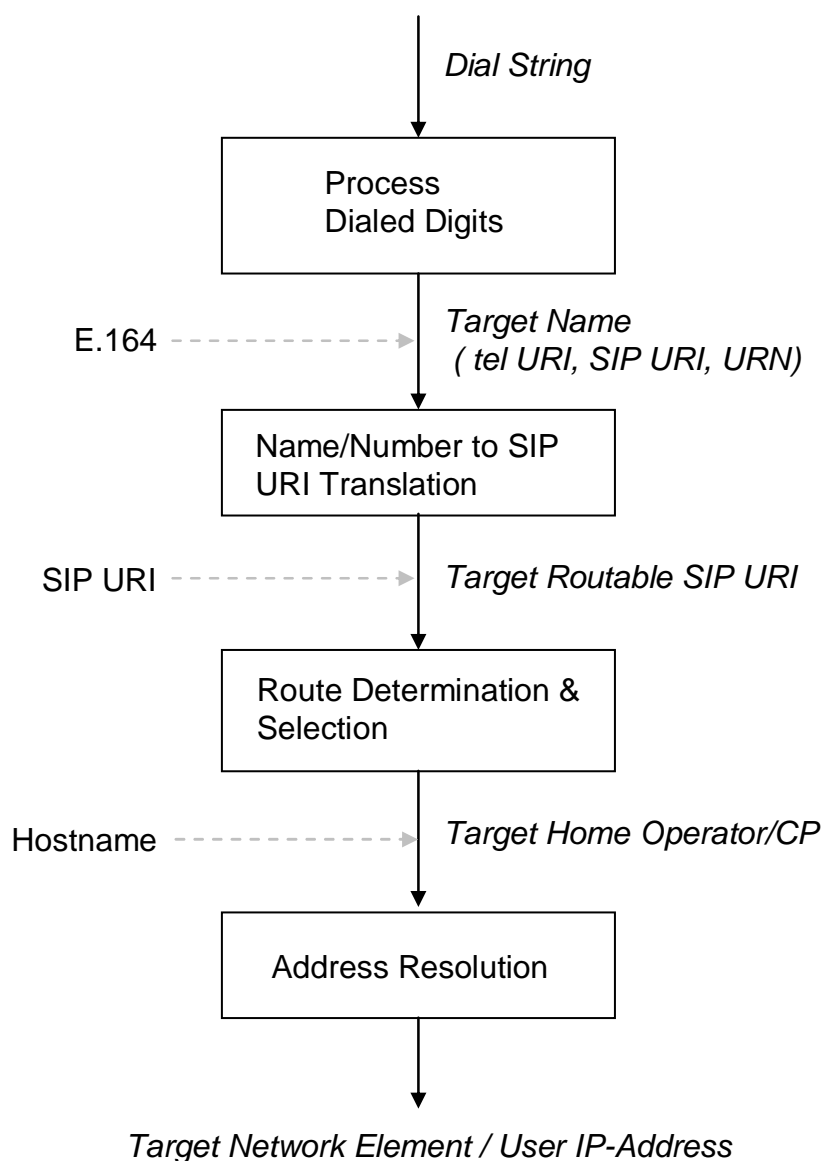
Address resolution function can be logically divided in two general phases:

- Number to naming mapping: where the number is converted to a format which can be used by the name resolution function.
- Naming resolution: where the internal domain naming, looking up in an infrastructure ENUM or in equivalent database or static table, to retrieve "Routeable" SIP URI (that contains the serving operator domain).

If number or name can not be solved locally, address resolution function passes number on, towards a pre-configured triggering internal entity or different CP network, using tel URI format.

When the number/name has been resolved, the Routeable SIP URI is used to derive the IP address of next functional entity, generally basing on DNS mechanisms or on local naming-address mapping table functionalities.

Figure 5 illustrates such a resolution process that is required for number and naming translations from the dial string to the IP address, associated to the next hop functional entity.



**Figure 5: Naming/Numbering Address Resolution overview (for details see [i.2] clause 6.3)**

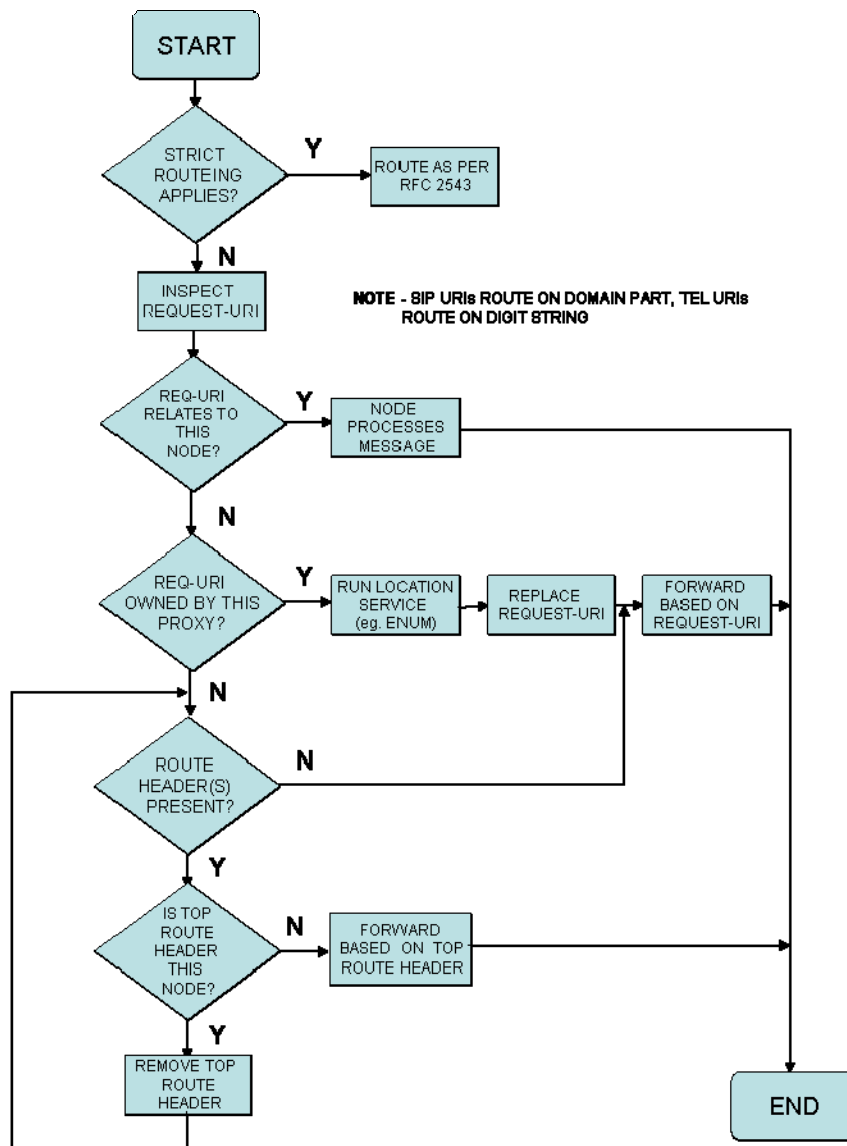
It should be noted that ENUM can provide only the domain of the home CP for the B-party or terminating CP. On the basis of such domain information, the internal routing functions, have to define the next hop entity or border gateway to handle call/session and/or the transit role CP, on the base of specific routing arrangement defined by CPs (e.g. on the base of bilateral agreements).

**NOTE:** Generally, in alignment with ETSI/3GPP specifications, the presence of an E.164 number in a specific infrastructure ENUM assures that associated terminating/serving CP is reachable through pre-defined interconnection arrangements and related networks interconnection; for instance, that is the case of IPX (GRX successor), and internal carrier ENUM infrastructures managed by GSMA and related to the fixed-mobile international interconnection context.

The following rules for SIP header coding apply:

- 1) The called/terminating user identity is inserted in the TO field.
- 2) The Request-URI field content shall be, when available, a "Routeable" SIP URI that is a SIP URI with domain part identifying terminating/serving CP determined from an authoritative source or locally pre-defined. Otherwise Request-URI field content shall be E.164 number coded by tel URI format. The number in the tel URI coding format can be in national or international E.164 number format, in accordance with ITU-T Recommendation E.164 [17].
- 3) Route header field contains the next hop(s).

The above is based on figure 6 which is derived from analysis of various parts of RFC 3261 [18].



**Figure 6: header handling in SIP**

### 5.7.1 tel URI and SIP URI resolution requirements and procedures

Figure 7 describes a general process and related functions for tel URI and SIP URI, provided by the user (Dialling) or in interconnection scenarios, based on ENUM systems or other translation database functions.

The flow shows the potential usage of infrastructure ENUM (if available) or of other DBs in order to define the correct route towards both local network terminations and out-bound interconnected network and CPs.

Note that, in figure 7, the local Number Analysis function (NAR) is used to handle cases where I-ENUM fails to find the number and associated NAPTR record and to find out the appropriate destination which delivers the call/session to; that could be the situation in which dialled number is not present in ENUM or the derived SIP URI domain part is not recognized locally as a reachable CP.

The general outputs of figure 7 procedures and functions are the following:

- if the resolution of tel URI or SIP URI has succeeded, then the session is delivered to the customer network termination point or the "Routable" SIP URI is passed on to the interconnected CP;
- if the resolution has not succeeded, then analysis functions will check if it is a circuit-switched PSTN/ISDN or PLMN destination or other "All IP" networks not integrated with NGN environment or reachable by other interconnected CPs, in which case a tel URI or SIP URI will be provided to for routing resolution.

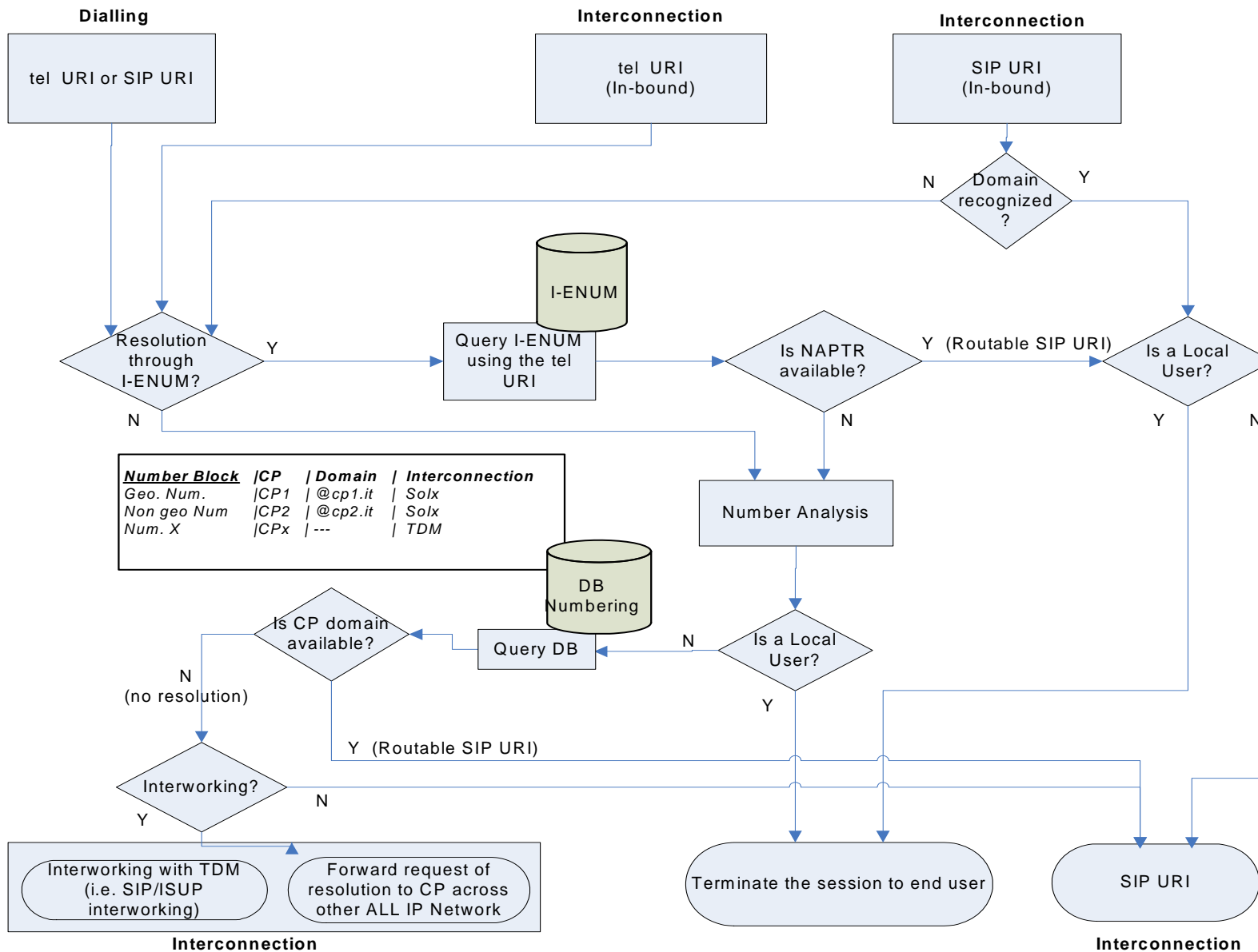


Figure 7: General functions and procedures for tel URI and SIP URI resolution in NGN



## 5.8 Relation with NAR Interconnection general framework

The gap analysis based on NAR [i.2] use cases has identified a requirement to introduce NAR functionality in ETSI NGN architecture. The present document provides further definition of NAR functionality for NGN Interconnection routing. NAR for Interconnection and routing is concentrated in the control and service layers which are responsible for handling numbering, naming and routing tasks and driving lower layer requirements (e.g. transport entities).

NAR does not correspond either to new entities or new specific reference points. Instead there is a need to focus on a homogeneous description that addresses these specific needs of NGN interconnection and routing. Because the NAR functionality is the context aware resolution of an identifier into a result which is useable for NGN routing, it shall include also the interconnection and routing requirements.

Figure 8 describes, on the basis of [4], the specific NAR functionality applied to the interconnection and routing entities that can consist in general of one or more relations with service/control layers entities.

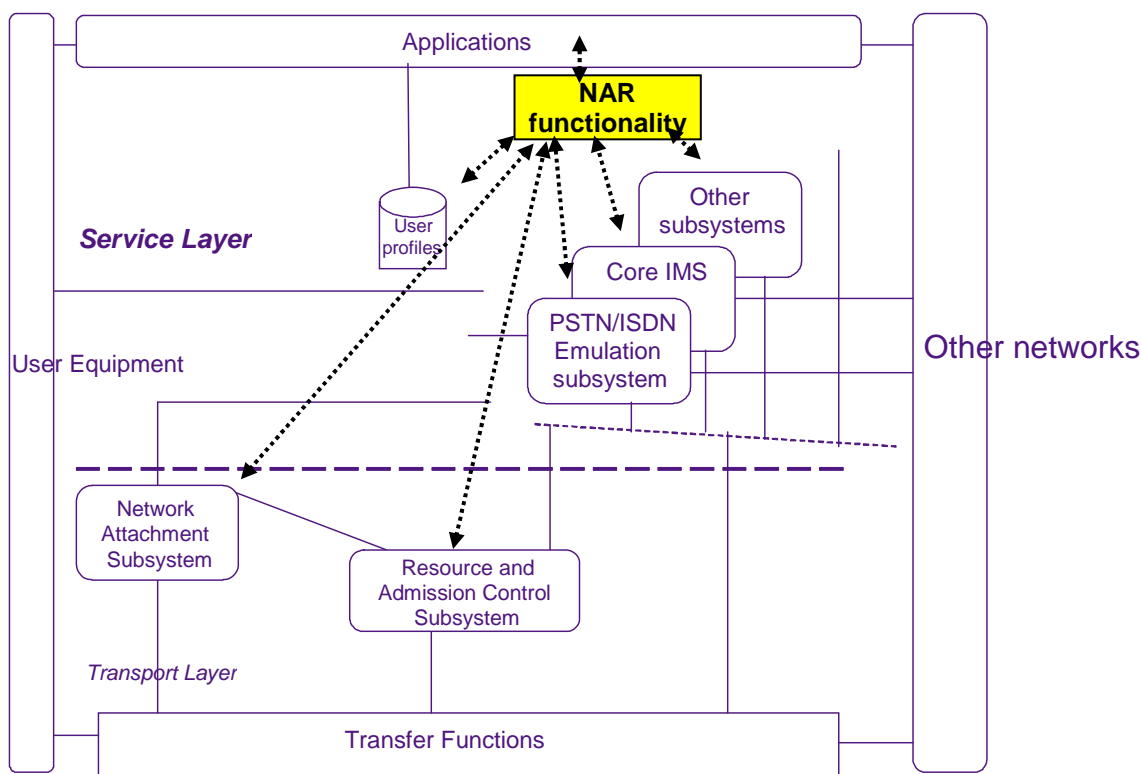


Figure 8: NAR functionality in ETSI NGN architecture for interconnection

## 6 Interconnection, interworking, interface and routing requirements

Interconnection refers to the connection of two networks so that they can support interoperable services. There could be pure transport IP connectivity interconnection, either with "best effort" or with specific QoS classes, or "service-aware" interconnection for multimedia and innovative "session-based" services.

Interconnection takes place at a reference point at which a specific Network to Network Interface (NNI) is defined, generally related to different end-to-end interoperable offered service. The characteristics of the NNI depend on the kind of networks / administrative domains to be interconnected, for instance IMS, PSTN/ISDN, and PLMN, other IP-based networks, etc.

In general two different networks are interconnected by session border functionalities, often located in specific border equipments in which any necessary interworking functionalities (for instance, the conversion from one protocol to another) are provided. That allows separating the two networks like two different domains, in order to manage them independently.

Thus for successful interconnection definition, there is a need to identify at least:

- Interconnection service required, i.e. pure IP connectivity or "session-based" voice/multimedia service.
- NNI Interface requirement for specification, also for rules between the signalling and transmission control on each side of the session border functionalities or interworking function

As a reference scheme figure 9 shall be considered.

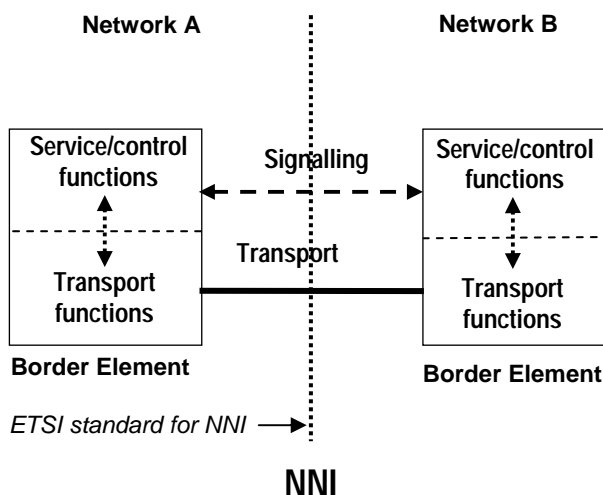


Figure 9: Reference scheme for NNI Interconnection between two networks

## 6.1 Interconnection NNI in IMS and NGN architecture

Figure 10, which is based on the relevant ETSI standards and specifications [4] and [5], provides an overview of different possible NNIs. Clause 5 of the present document, which describes the routing model for interconnection, covers the following types of service interconnections:

- "basic end-to-end service interconnection" for the routing and setting up of a call/session between two NGN domains and related end users;
- "Roaming end-to-end interconnection" to support a call originated from a roaming (visited) network or a call extended from the B-party serving (home) network to the roaming (visited) network. However this case is out scope of the present document.

The following possible locations of NNIs are also considered to be in the scope of the present document (excluding internal interfaces because they are not interconnection points towards other networks):

- 1) Signalling and transport NNI between IMS networks in different administrative domains.
- 2) Signalling and transport NNI between IMS network and other Non IMS IP networks.

Although the general NGN routing model can be adapted to include transport layer interconnection scenarios described in the NGN general architecture [4], they are outside the scope of the present document which relates only to end-to-end service interconnection.

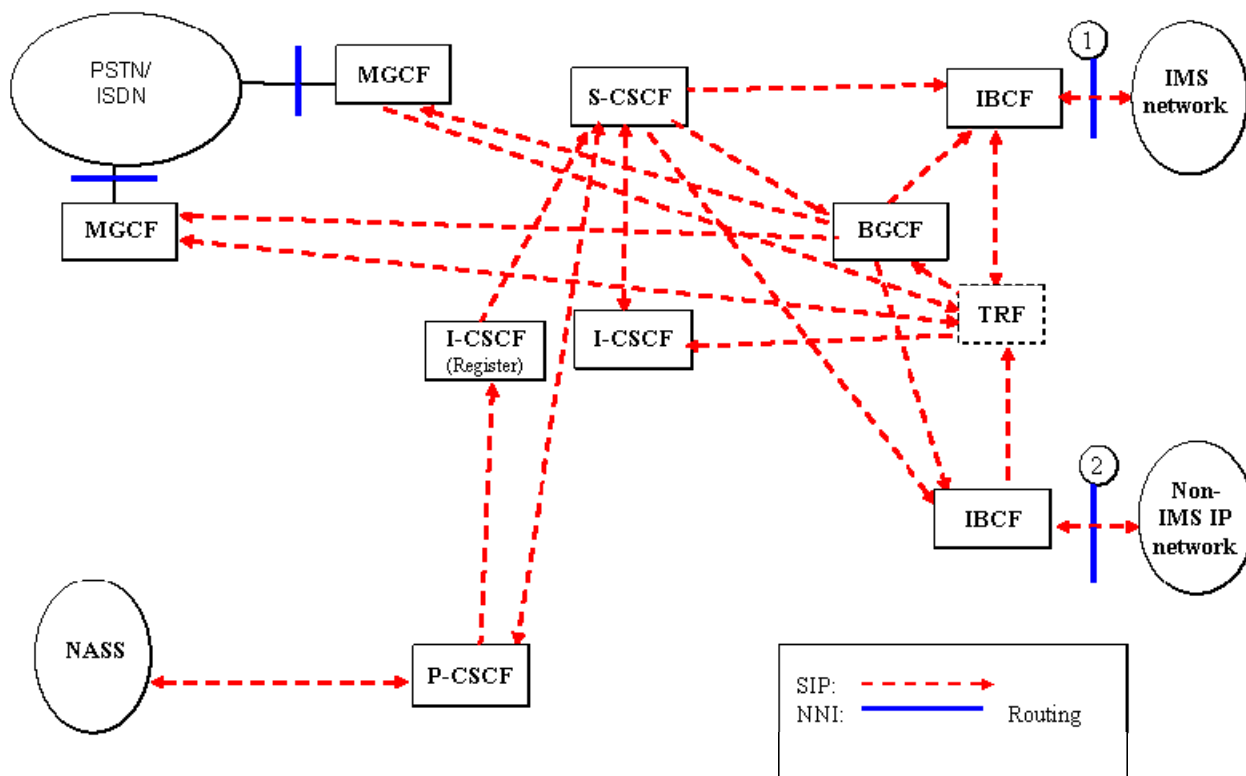


Figure 10: NNI possibilities based on [4], and [5]

## 6.2 Transit network scenarios

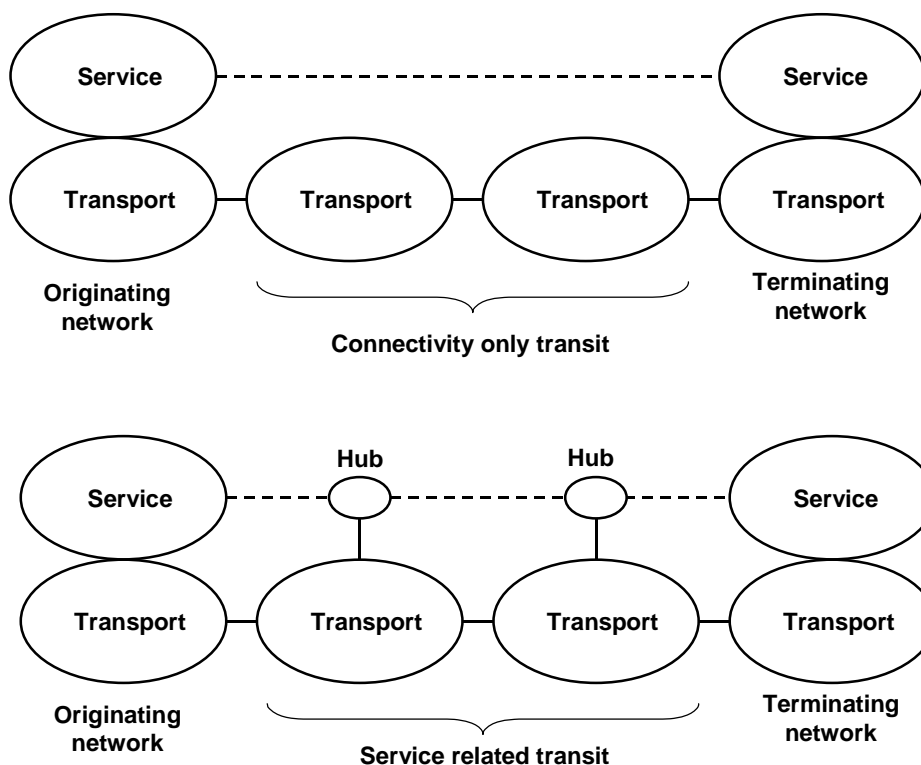
The term transit refers to the routing of calls and other communications through networks other than the originating and terminating ones.

There are essentially two general categories of NGN transit networks:

- "IP Connectivity only" transit network, where IP packets traffic is delivered by transit network without any knowledge of the carried service. This is referred to as CoIx Interconnection.
- "Service related" transit network, where the transit network interacts with the communications at the service control level for end-to-end service interoperability. This is referred to as SoIx Interconnection.

Every network provides at least some connectivity but may or may not provide service related transit. The proxy functions needed for service related transit may be provided by a third party on a connectivity platform.

Some options are illustrated in figure 11.



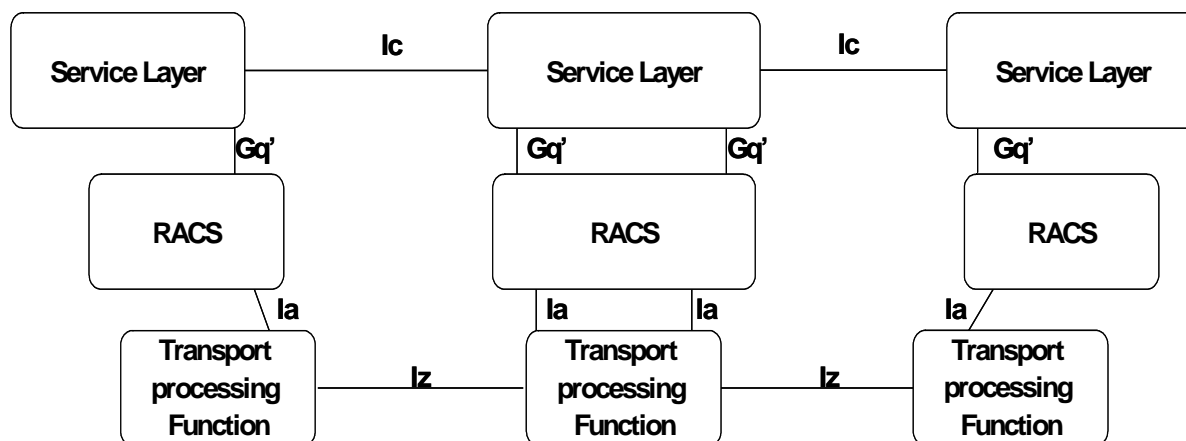
**Figure 11: Examples of different transit network scenarios**

There are two main types of platforms providing IP-level connectivity:

- Dedicated "telco type" networks (e.g. via IPX which can be viewed as an extranet).
- The public Internet.

Note also that transit level functionality may be provided on a hop-by-hop basis across multiple networks that each run their own service related transit function.

Figure 12 illustrates SoIx interconnection where the transit network does interact with the signalling.



**Figure 12: Solx-to-Solx Transit Interconnection**

In this figure, end-to-end service interoperability can be guaranteed by the Transit Domain as a result of SoIx interconnection on both sides. End-to-end service requirements are delivered by all interconnected networks through the control of transport resources via the RACS functions.

## 6.3 Application to different NGN scenarios

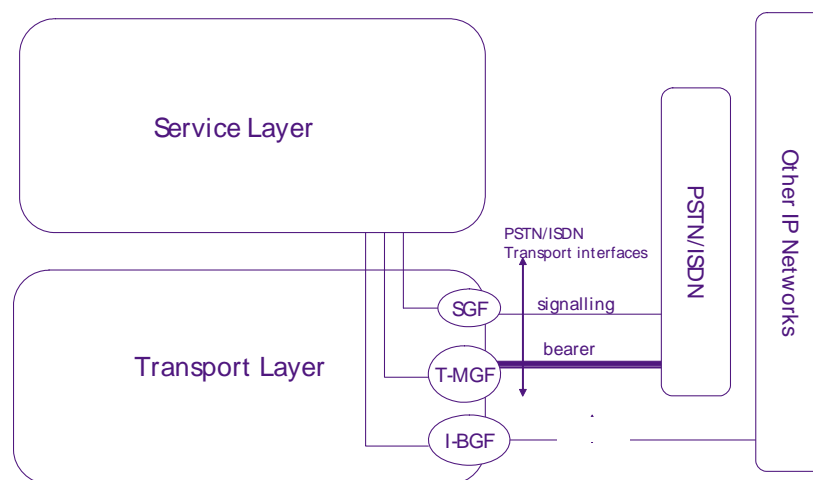
Table 3 shows the application of NGN scenarios, based on an "IP-based, session-based" NNI and related naming/numbering requirements. Starting from ETSI NGN architectures and related subsystems (i.e. PES, IMS, etc.) and considering concrete "IP-based" NNI "session-based" interconnection scenarios, a limited number of cases can be identified.

**Table 3: Main interconnection scenarios, protocols and related numbering/naming requirements**

"IP-based" session-based interconnection scenario	ETSI NNI signalling protocol	NNI transport (control and user plane)	Naming/numbering requirements
NGN (CS domain)↔NGN (CS domain)	SIP-I	IP-based	E.164 number or tel URI (in case of SIP URI the user part is the key for routing. The domain part could be used for media gateway selection)
NGN (PS domain)↔NGN (PS domain)	SIP	IP-based	tel URI or SIP URI
NGN (PS domain)↔NGN (CS domain)	SIP-I or SIP	IP-based	E.164 number, tel URI or SIP URI
NGN↔ other IP network	SIP	IP-based	SIP URI
NGN↔PSTN/ISDN/PLMN	ISUP/BICC	SS7 and TDM	E.164 number

Note that in the table above, the term "CS domain" means that the control plane is NGN subsystem (IMS, PES) and the user plane or bearer is based on circuit switch technology.

Figure 13 describes the interworking of NGN with PSTN/ISDN and Other IP Network (that has been defined into [5]) that are described into the last two lines of table 3.



**Figure 13: Network interconnection at transport layer [5]**

The following conclusions can be drawn:

- NNI definitions should not depend on the specific NGN subsystem involved, since it is a fundamental objective to converge toward strictly limited NNI interfaces and protocols that cover all possible scenarios.
- NGN must be able to support interworking with circuit-switched networks.

---

## Annex A (informative): Protocols at the NNI points and interworking

The signalling NNI protocol used for IMS interconnection is SIP and both ETSI and ITU-T Recommendation Q.3401 [9] (NNI for voice band services) have made adaptations of SIP, defined by IETF in RFC 3261 [18], for different network applications.

Interconnection between networks adherent to different versions of the SIP standard can require an interworking function. The main different SIP standards are the following:

- The ITU-T profile for the pure SIP NNI specified in ITU-T Recommendation Q.3401 [9].
- The ITU-T Recommendation Q.1912.5 "Interworking between Session Initiation Protocol (SIP) and Bearer Independent Call Control protocol or ISDN User Part" [10]. ITU-T Recommendation Q.1912.5 [10] adapts and references RFC 3261 [18] for interworking with BICC/ISUP networks.
- The 3GPP pure SIP IMS call control documents:
  - TS 123 228 [11] Stage 2 which gives the main signalling flows for the IMS core network.
  - TS 124 228 [12] Stage 3 signalling flows which gives example signalling flows for the IMS call control and expands on the information in TS 123 228 [11].
  - TS 124 229 [13] Stage 3 protocol which specifies the IMS call control protocol using SIP and SDP for the IMS core network. TS 124 229 [13] makes is an endorsement/modification of RFC 3261 [18] and other RFCs.
  - TS 129 163 [14] Interworking between the IP Multimedia (IM) Core Network (CN) subsystem and Circuit Switched (CS) networks, based on ITU-T Recommendation Q.1912.5 [10];
- The TISPAN pure SIP IMS call control document:
  - ES 283 003 [7] which modifies and endorses TS 124 229 [13].

Figure A.1 shows the development from the base SIP specification of the three different forms of SIP used for NNIs, i.e. SIP-I, IMS SIP, and non-IMS pure SIP.

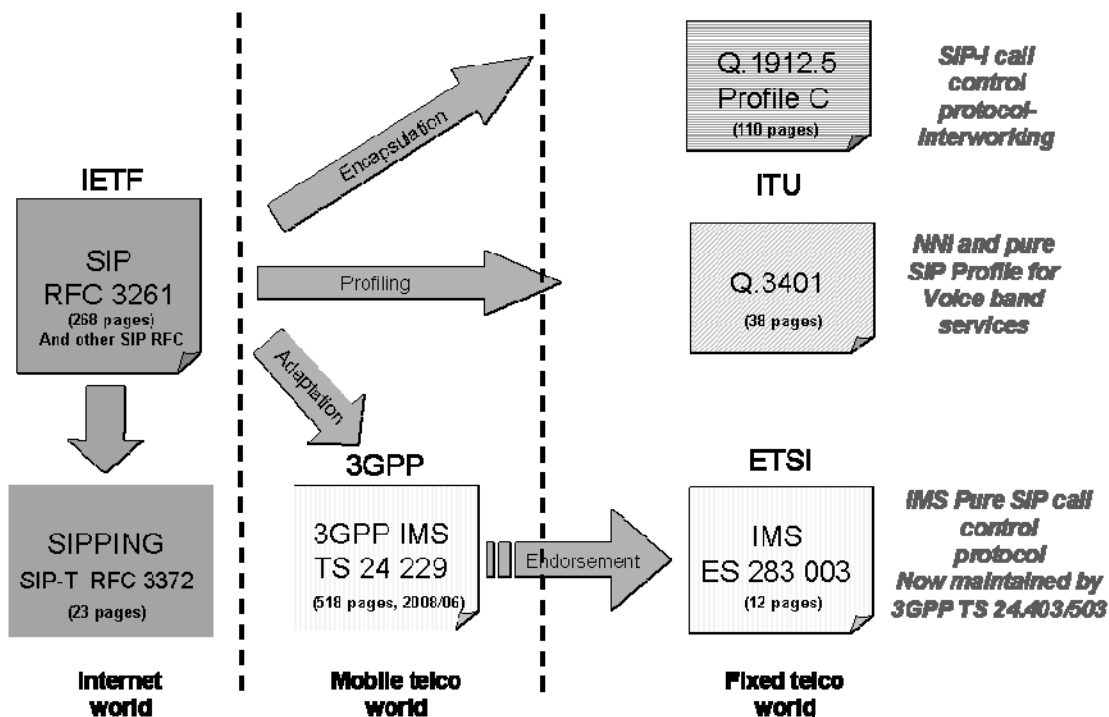


Figure A.1: Development of SIP variants for NNIs

Figure A.2 shows the development of the interworking specifications that relate the different forms of SIP to ISUP and BICC. The figure again shows SIP-I, the IMS SIP, and the non-IMS pure SIP.

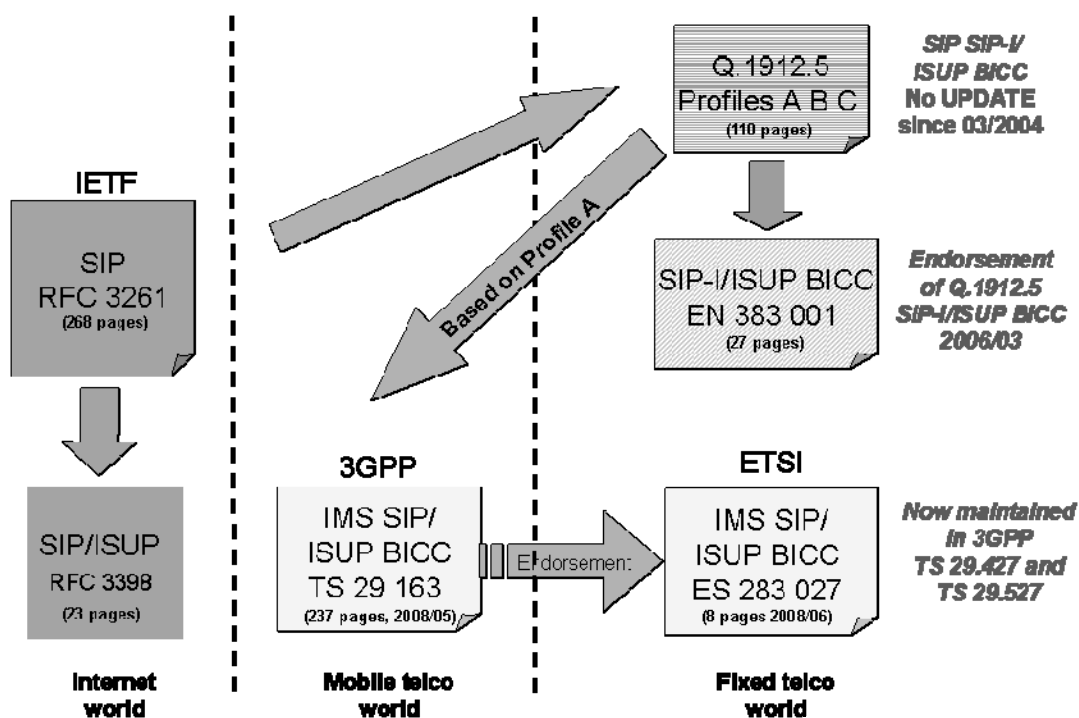


Figure A.2: Development of SIP interworking specifications

## Annex B (informative): SIP Header Fields at the NNI relevant to routing and identification

The critical SIP element from a routing point of view is the INVITE method. An example of an INVITE method is shown in figure B.1. It is a text string, and consists of:

- The word INVITE followed by either a tel URI or a SIP URI, which may be followed by one or more parameters.
- A number of subsequent lines called headers, where each line starts with a pre-defined word and is followed by one or more header fields.

Each header field may contain one or more parameters.

These terms are shown in figure B.1.

```

Session Initiation Protocol
Request-Line:
  INVITE sip:+41454000191@cp-domain;user=phone SIP/2.0
Message Header
  To: <sip:+41454000191@cp-domain;user=phone>
  From: "Name" <sip:+41454000201@cp-domain;transport=udp>; tag= <?>
  Call-ID: <?>
  CSeq: 1 INVITE
  Max-Forwards: 68
  Content-Length: 343
  Via: SIP/2.0/UDP 123.45.123.48 :5060;branch= <?>
  Via: SIP/2.0/UDP 123.45.123.150:5060;branch= <?>
  Via: SIP/2.0/UDP 192.168.1.36 :2725;
  received=10.7.7.149;branch= <?>;
  rport=56608
  Route: <sip:scscfl.intranet.net:6050;lr>
  Record-Route: <sip: <?>cp-domain@scscfl.intranet.net:6050;
  maddr=123.45.123.48;lr>
  Contact: <sip:+41454000201@123.45.123.150:5060; transport=udp>;
  flow-id=1
  Content-Type: application/sdp
  Allow: INVITE, ACK, CANCEL, BYE, REFER, OPTIONS,
  NOTIFY, SUBSCRIBE, PRACK, MESSAGE, INFO
  Allow-Events: talk
  Allow-Events: hold
  Allow-Events: refer
  Accept: application/sdp
  Supported: 100rel, replaces, callerid
  P-Asserted-Identity: "Name" <sip:+41454000201@cp-domain>
  User-Agent: snom320/SC0825
  P-Charging-Vector: icid-value=<?>
  Session-Expires: 360
  Min-SE: 360
  P-Key-Flags: keys="3"
Message body
Session Description Protocol
Session Description Protocol Version (v): 0
Owner/Creator, Session Id (o): root xx yy IN IP4 10.1.1.149
Session Name (s): call
Connection Information (c): IN IP4 123.45.123.85
Time Description, active time (t): 0 0
Media Description, name and address (m):
  audio 14734 RTP/AVP 18 8 9 2 3 4 101
Media Attribute (a): rtpmap:18 g729/8000
Media Attribute (a): rtpmap:8 pcma/8000
Media Attribute (a): rtpmap:9 g722/8000
Media Attribute (a): rtpmap:2 g726-32/8000
Media Attribute (a): rtpmap:3 gsm/8000
Media Attribute (a): rtpmap:4 g723/8000
Media Attribute (a): rtpmap:101 telephone-event/8000
Media Attribute (a): ftmp:101 0-16
Media Attribute (a):ptime:20
Media Attribute (a): sendrecv

```

Headers

Figure B.1: Structure of the INVITE Message



The specification of SIP protocol is defined in ES 283 003 [7]. The following explains the main header fields relevant to routing and identification as used in TISpan.

NOTE 1: Some of these uses are more restrictive than are specified in RFC 3261 [18].

- Request-URI

The Request-URI field identifies the incoming call server (the I-CSCF in the case of IMS) of the serving network. A proxy can only change the Request-URI of a request during forwarding if it is responsible for that URI.

- VIA

The VIA header field(s) keeps track of all the proxies a request has traversed. The response uses these VIA(s) to be routed back via the same proxies as traversed by the request. The VIA header is mandatory in all SIP messages, and as a minimum lists the involved UAs.

If the host portion of the "sent-by" parameter contains a domain name, or if it contains an IP address that differs from the packet source address, the server must add a "received" parameter to that VIA header field value. This parameter must contain the source address from which the packet was received. (This is to assist the server transport layer in sending the response, since it must be sent to the source IP address from which the request came.)

Consider a request received by the server transport which looks like, in part:

```
INVITE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP bobspc.biloxi.com:5060
```

The request is received with a source IP address of 192.0.2.4. Before passing the request up, the transport adds a "received" parameter, so that the request would look like, in part:

```
INVITE sip:bob@biloxi.com SIP/2.0
Via: SIP/2.0/UDP bobspc.biloxi.com:5060; received=192.0.2.4
```

- Contact

Contact headers are exchanged in order to let peer UAs exchange SIP signalling directly with no intermediaries. This applies except for the proxies that have explicitly requested to remain in the signalling path by adding their URI in a previous Record-Route.

- Record-Route

The header is added by a Proxy during a transaction in order for the Proxy to remain in the path for all requests sent within the dialog. The set of all recorded routes are carried in the response to this request. Route headers are added by the initiator in subsequent transactions to enforce the routing as recorded by the Record-Routes. The Record-Route header field is inserted by proxies in a request to force future requests in the dialog to be routed through the proxy.

- Route

Enforces the routing as recorded in a previous transaction by applying Record-Routes.

- P-Called-Party-ID

Identifies which of the several possible identities an INVITE is addressed to.

- To

This header is used for UA-UA purposes and so should be passed unchanged and unused by transit networks.

NOTE 2: This header is used by the networks in the REGISTER process.

NOTE 3: The "tag" parameter is used in the To and From header fields of SIP messages. It serves as a general mechanism to identify a dialog, which is the combination of the Call-ID along with two tags, one from each participant in the dialog.

- Call-ID:

The Call-ID header field acts as a unique identifier to group together a series of messages. It shall be the same for all requests and responses sent by either UA in a dialog. It should be the same in each registration from a UA.

- Branch:

The Via header field value shall contain a branch parameter. This parameter is used to identify the transaction created by that request. This parameter is used by both the client and the server. The branch parameter value shall be unique across space and time for all requests sent by the UA.

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## History

<b>Document history</b>		
V2.1.1	September 2008	Publication