

ENUM based Number Portability in VoIP and IMS Networks

Mario Ivček

Research & Development Centre

ERICSSON Nikola Tesla d.d.

Krapinska 45, 10002 Zagreb, Croatia

Telephone: +385 1 365 4619 Fax: +385 1 365 3548 E-mail: mario.ivcek@ericsson.com

Abstract – Number portability is today implemented only in the Public Switched Telephony Networks (PSTN) by utilizing Intelligent Network (IN) functions for number mapping. At the time being, there is no specific solution for number portability across Voice over IP (VoIP) and IP Multimedia Subsystem (IMS) networks. Limited portability can be implemented in IP telephony network using redirect and proxy servers that forward calls to gateways based on routing number. This paper discuss possibility of using E.164 Number Mapping (ENUM) for number portability in VoIP and IMS networks, based on the most recent IETF enum working group achievements.

I. INTRODUCTION

Number portability [6] enables telephony subscribers to keep their telephone numbers when they change service provider (service provider portability), move to a new location (location portability), or change the subscribed services (service portability, e.g. from the plain old telephone service to Integrated Services Digital Network).

Today, number portability is only implemented in the Public Switched Telephony Networks (PSTN). The implementation of number portability differs in different countries. Common to all implementations, is that the Called Party Number dialed by the customer (also called directory number) is mapped to either a routing number or a routing prefix.

A routing number is a hierarchical routing address, which can be analyzed digit-by-digit to reach the correct country, network provider, end-switch and subscriber line at the end. A routing prefix forms a routing number by adding some digits in front of the directory number. The ISUP Called Party Number parameter (CdPN) in the IAM will contain the routing prefix and dialed directory number. This model is used in most European countries (Austria, Belgium, Finland, France, Germany, Italy, Norway, Spain, Sweden, Switzerland, UK).

Most number portability solutions utilize Intelligent Network (IN) functions. In these solutions, the mapping database is stored in the Service Data Functions (SDF) elements.

The introduction of IP telephony adds a new type of number portability: between different network types [6]. At the time being, there is no specific solution for number portability across the network types. A number that is moved to the IP network can be handled in a normal way, using number portability solutions on the PSTN. Number

portability databases are updated with a routing number that directs call to a gateway.

Limited number portability can be implemented in IP telephony network using redirect and proxy servers. Calls to ported numbers are forwarded to a new destination by the previous signaling server. However, this feature does more correspond to call forwarding than number portability. The forwarding works with both calls to IP terminals and calls to PSTN destinations.

This paper, in chapter two, starts with a general overview of number portability service. Chapter three presents existing number portability between PSTN networks in VoIP domain. Aspects of call routing in IP environment are described in chapter four. Chapter five aims at presenting problems encountered by number portability across network types. The ENUM, enriched with newly registered 'pstn' enumservice, is considered as a potential solution to these problems, along with the NP related information that is defined on SIP signaling level. The ENUM based number portability applied to different call scenarios is discussed in chapter six. Conclusions are driven in chapter seven.

II. NUMBER PORTABILITY IN GENERAL

Number portability (NP) impacts call signaling and routing. The former impact is connected with the need to carry the NP-related information in the protocols (such as ISUP, TUP) after a Number Portability Data Base (NPDB) lookup has been performed. The later impact is for a signaling server to use the NP-related information in received ISUP message parameters to determine routing. A routing number is associated with a telephone number that has been ported out from a donor network to a serving network. A donor network is the initial network where a telephone number (e.g. +385 1 123 4567) was assigned to a subscriber before ever being ported. A "non-porting" telephone number does not have any routing number associated with it because the first N digits of the number can be used for routing. The serving network is the network that currently serves the ported number.

In this paper, we will discuss number portability in modern Next-Generation Network (NGN), which is characterized by separated application and media control layer. The NGN includes a telephony softswitch, media gateway, signaling gateway and service management platform.

A telephony softswitch (TSS) is open-standard software that can perform distributed communications functions on an open computing platform and has the functionality of a traditional TDM telephone switch. A softswitch can integrate voice, data and video, and it can execute protocol translations between separate networks such as PSTN and IP. The media gateway communicates between networks with different protocols, such as PSTN voice and IP data networks. The signaling gateway translates telephone signaling messages for transmission over packet data networks. The service management platform supports different management services, like intelligent network services for example.

III. NUMBER PORTABILITY IN VoIP NETWORK

The four methods can be used to support number portability across PSTNs in the VoIP environment: All Call Query (ACQ), Query on Release (QoR), Call Dropback and Onward Routing (OR) [6]. The Query on Release method will be discussed further in this paper.

When using the QoR method for number portability [10], the initiating exchange routes the call to a donor exchange with an optional indication that a QoR capability is possible. If the donor exchange is not a serving exchange, the call is released with appropriate release indication. On receipt of the release indication, an exchange in the upstream determines routing information by initiating query to an external database. A database response message contains a network Routing Number (RN).

The following model depicts a typical PSTN network call scenario involving number portability based on QoR method [fig.1].

1. The originating PSTN network receives a call from the caller and routes the call to telephony softswitch, by sending the IAM message that contains a Called Party Number (CdPN) of the called subscriber.
2. Based on analysis of CdPN, the softswitch determines that call is to be routed to terminating PSTN network, in which the called subscriber is supposed to be resident. An IAM message with CdPN is sent towards terminating PSTN network.
3. Upon receive an incoming call (IAM), a switch in the receiving PSTN network detects either that the called number has been ported out to another network or, optionally, that the number is just vacant in the network. After having determined (by looking at received signalling information) that one of the preceding networks has QoR capability, the receiving PSTN network returns a release message (REL) back to softswitch, with or without a special indication that the called number is ported out. From number portability point of view, this network is referred to as Donor network.
4. The softswitch traps the release message and determines that the preceding network has no QoR capability. In this scenario, the softswitch has access to an NPDB with the complete address to Recipient network for ported-out numbers. Based on received cause for release, the softswitch makes a database query to the centrally administered NPDB, by supplying a CdPN in the query. From number

portability point of view, the CdPN is referred to as Dialed Directory Number (DDN).

5. The NPDB returns a Routing Number (RN) that is associated with the requested DDN. The RN points at the ported-to subscriber in the recipient exchange.
6. The softswitch uses obtained RN to reroute the call onward towards the Recipient network. The IAM message to Recipient network carries complete addressing information, i.e. both DDN and RN.
7. Finally, when terminating switch in the Recipient PSTN network finds its own routing information, by analysing the supplied RN, it uses received DDN to terminate the call.

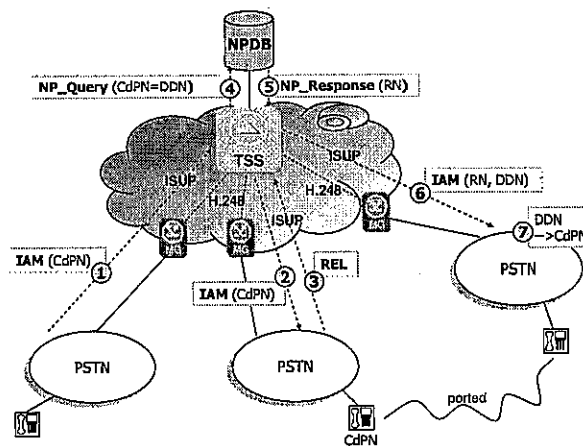


Figure 1: Number Portability in VoIP Network

IV. ROUTING IN IP NETWORK

To establish a call to a terminal on an IP network, the destination IP address must be known. Alternatively the terminal can be identified by a host name, which is translated to an IP address by DNS. As terminals are equipped with an E.164 number, DNS service can be used to map telephone numbers into IP addresses, by constructing a domain name from the E.164 number and performing DNS NAPTR lookup. The result of NAPTR lookup is a Uniform Resource Identifier (URI), which contains a signaling protocol and a host name [1].

An additional DNS lookup is thus required to map the host name to an IP address. Note that for simplicity reasons, those additional lookups will not be considered in this paper.

ENUM uses the domain "e164.arpa" to store the mapping. Numbers are converted to domain names using the scheme defined in [3]. The E.164 number must be in its full form, including country code. All characters and symbols are removed, only the digits remain. Dots are put between the digits. The order of the digits is reversed and the string "e164.arpa" is added to the end. This procedure will map, for example, the number +385-1-1234567 into the host name "7.6.5.4.3.2.1.1.5.8.3.e164.arpa".

DNS stores information in different types of records. The Naming Authority Pointer (NAPTR) record [2] is used for identifying available ways to contact a node with a

given name. It can also be used to identify what services exist for a specific domain name. ENUM defines a new service named "E.164 to URI", which maps one E.164 number to a list of URIs. The mnemonic of the service is "E2U". ENUM can be used in conjunction with several application protocols, and can for example, map a telephone number to an email address.

Figure 2 shows an example of NAPTR records with the E2U service. These records describe a telephone number that is preferably contacted by SIP and secondly by either SMTP or using the tel URI scheme. The result of rewrite of the NAPTR record is a URL, as indicated by the "u" flag in the NAPTR record. The own resolution methods of SIP and SMTP are used. In case of SIP protocol, the URI might be a SIP URI, which is resolved as described in [5]. In case of the tel URI scheme, the procedure is restarted with this new E.164 number.

```

$ORIGIN 4.3.2.1.6.7.9.8.6.4.e164.arpa.
IN NAPTR 10 10 "u" "sip+E2U"
    "!^.*$!sip:paf@swip.net!" .
IN NAPTR 102 10 "u" "mailto+E2U"
    "!^.*$!mailto:paf@swip.net!" .
IN NAPTR 102 10 "u" "tel+E2U"
    "!^.*$!tel:+4689761234!" .

```

Figure 2: Example of NAPTR Record

V. NUMBER PORTABILITY ACROSS NETWORK TYPES

When the circuit switched network and IP telephony networks are interconnected, new call scenarios arise: PSTN-PSTN, PSTN-IP, IP-PSTN and IP-IP. When calls are setup, the first task is to determine the type of destination network. A mapping from E.164 name to network type is required.

At the time being, there is no specific solution for number portability across the network types. A number that is moved to an IP network can be handled in a normal way in the number portability solutions on PSTN. The number portability databases are updated with a routing number that directs 'ported' calls to a gateway.

The following model depicts a typical call scenario in the interconnected PSTN and IP network, which involves number portability based on QoR method [fig 3].

1. The softswitch receives SIP INVITE message from the IMS network with Request-URI that points to called user identity described via tel URI. The tel URI contains CdPN of the called subscriber.

Number (address) analysis at the softswitch is set up in a way that NAPTR lookup to external DNS database is triggered for every single call, in order to identify available services connected to the received E.164 number. For example:

```
INVITE tel:+1-202-555-1234 SIP/2.0
```

2. Based on provided CdPN, a domain name is created and used for DNS NAPTR query towards external DNS.

```
4.3.2.1.5.5.5.2.0.2.1.e164.arpa.
```

3. At receive of DNS response, softswitch applies selection of those NAPTR RRs that support E2U resolution service. These records describe a telephone number that is preferably contacted by using the tel URI.
4. The tel URI received from DNS contains a number that is equal to the original CdPN. Based on analysis of the CdPN, the softswitch determines that call is to be routed to terminating PSTN network, in which the called subscriber is supposed to be resident. An IAM message with CdPN is sent towards terminating PSTN network.
5. Upon receive an incoming call (IAM), a switch in the receiving PSTN network detects the called number has been ported out to another network so it returns a release message (REL) back to softswitch, with indication that the called number is ported out. From number portability point of view, this network is referred to as Donor network.
6. The softswitch traps the release message and, based on received cause for release, makes a database query to NPDB by supplying a CdPN. From number portability point of view, the CdPN is referred to as Dialed Directory Number (DDN).
7. The NPDB returns a Routing Number (RN) that is associated with the requested DDN.
8. The softswitch uses obtained RN to select a proper gateway for rerouting the call onward towards the Recipient VoIP/IMS network. The SIP INVITE message to Recipient network carries CdPN in the tel URI.

```
INVITE tel:+1-202-555-1234 SIP/2.0
```

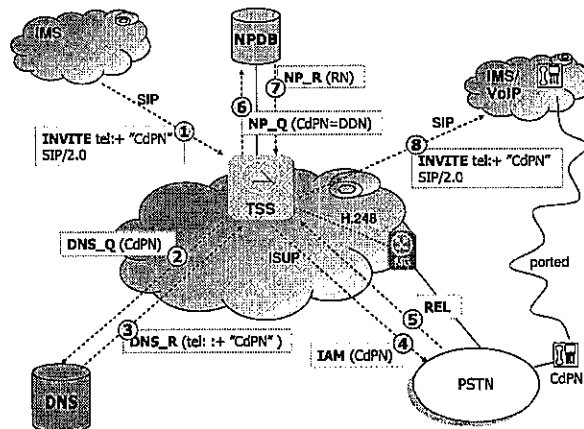


Figure 3: Number Portability across Network Types

Above depicted model reveals that two lookups to external databases (NAPTR lookup to DNS and NP lookup to NPDB) are now required, compared to one NP lookup for the case of PSTN number portability. Such extensive database lookups bring time and capacity issues in play.

Moreover, number portability enables subscribers to move from one provider's network to another, and even between network types. If a number belonging to a number block of a PSTN operator moves to an IP/VoIP network as

presented in the previous call case, calls from IP subscribers may unnecessarily be routed through PSTN.

The call can thus propagate through several network types. Each time the call goes from one network type to another, it has to pass a gateway where the media stream is converted. The conversions cause delay and jitter, which on the other hand decrease quality. Therefore, unnecessary media conversions should be avoided.

This particular problem shall be tackled from two different angles: NP database aspect and signaling/routing aspect.

A. NPDB for number portability across network types

When it comes to NP database interface, the [6] proposes several alternatives. One alternative is to use certain entities in the IP-based networks for dealing with NP query, similar to the International Switches that are used in the PSTN to interwork different national ISUP variations. This will force signaling information to be routed to those IP entities that support NP functions.

Another alternative is to define a "common" interface to be supported by all NPDBs, so that all the IP entities use that standardized protocol to query them. The existing NPDBs can support this additional interface, or new NPDBs that contain the same information but support the common IP interface can be deployed. The ENUM service has capability of both. With that respect, the [9] registers a new enumservice 'pstn' to indicate PSTN routing data, including number portability data and non-ported telephone number data. The 'pstn' enumservice could enable service providers to place ported numbers, block of numbers and their associated PSTN contact information to externally available DNS (ENUM) database. This way, all telephone number lookups would be consolidated into a single DNS NAPTR lookup, thereby simplifying call routing and network operations.

B. Signaling information for number portability across network types

When it comes to signaling, SIP would need to transport and make use of some of the ISUP signaling information, even though ISUP may be encapsulated in SIP. This is necessary due to a fact that IP-based networks can handle the domestic calls between two PSTNs, and originating PSTN could perform an NPDB query. Also, if IP-base networks have performed the NPDB query, SIP also needs to transport the NP related information while the call is being routed to a destination PSTN. There are three pieces of NP related information that SIP needs to transport:

- 1) the called directory number, so that terminating PSTN network can terminate the call,
- 2) a routing number ('rn') to route the call to terminating PSTN network,
- 3) a NPDB dip indicator ('npdi'), so that the terminating PSTN will not perform another NPDB dip.

A called directory number already exists in the tel URI. The 'rn' and 'npdi' are defined as extensions to tel URI [8]. Those extensions will be automatically supported by SIP because they can be carried as the optional parameters in the user part of the SIP URI.

VI. ENUM BASED NUMBER PORTABILITY IN INTERCONNECTED VoIP/IMS NETWORKS

ENUM service has all capabilities to become a general method for address mapping in interconnected VoIP/IMS networks. From number portability perspective, the four call scenarios involving ENUM can be separated:

- A The number is not ported,
- B The number is ported between PSTNs,
- C The number is ported between PSTN and IP network,
- D The number is ported within IP network.

A. The number is not ported

For calls originating from IMS network to a non-ported subscriber in the PSTN network, the call scenario is depicted on fig 4.

1. The softswitch receives SIP INVITE message from the IMS network with Request-URI that points to a called user identity described via tel URI. The tel URI contains CdPN of the called subscriber. For example:

```
INVITE tel:+1-202-555-1234 SIP/2.0
```

2. Based on provided CdPN, a domain name is created and used for NAPTR query towards external DNS.

```
4.3.2.1.5.5.5.2.0.2.1.e164.arpa.
```

3. At receive of DNS response, the softswitch applies selection of those NAPTR RRs that support E2U resolution service. For example, a NAPTR record for non-ported number, using a 'tel' URI scheme, may look like:

```
$ORIGIN 4.3.2.1.5.5.5.2.0.2.1.e164.arpa.
NAPTR 10 100 "u" "E2U+pstn:tel"
"!^.*$!tel:+1-202-555-1234;npdi!".
```

As routing number ('rn') is not present in tel URI, the subscriber is considered as not-ported. The 'npdi' field is included in order to prevent subsequent lookups in legacy-style PSTN databases.

4. Based on analysis of the CdPN, the softswitch determines that call is to be routed to terminating PSTN network, in which the called subscriber is resident. An IAM message with CdPN is sent towards terminating PSTN network.

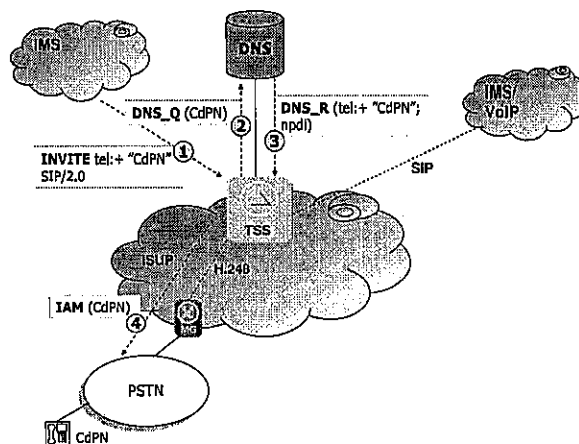


Figure 4: ENUM based NP: Number not ported

For PSTN originating calls, the steps 2, 3 and 4 would be identical to above described.

B. The number is ported between PSTNs

For calls originating from IMS network to a subscriber that is ported between two PSTN networks, which are served by the same softswitch, the call scenario is depicted on fig 5.

1. The softswitch receives SIP INVITE message from the IMS network with Request-URI that points to a called user identity described via tel URI. The tel URI contains CdPN of the called subscriber. For example:

```
INVITE tel:+1-202-555-1234 SIP/2.0
```

2. Based on provided CdPN, a domain name is created and used for NAPTR query towards external DNS.

```
4.3.2.1.5.5.5.2.0.2.1.e164.arpa.
```

3. At receive of DNS response, the softswitch applies selection of those NAPTR RRs that support E2U resolution service. For example, a NAPTR record for ported number, using a 'tel' URI scheme, may look like:

```
$ORIGIN 4.3.2.1.5.5.5.2.0.2.1.e164.arpa.
NAPTR 10 100 "u" "E2U+pstn:tel"
"!^.*$!tel:+1-202-555-1234;npdi!".
rn=+1-208-555-0199!".
```

As routing number ('rn') is present in tel URI, the subscriber is considered as ported. The 'npdi' field is included in order to prevent subsequent lookups in legacy-style PSTN databases.

4. The softswitch uses obtained 'rn' to reroute the call onward towards the Recipient network. The IAM message to Recipient network carries both RN (mapped from the 'rn') and CdPN.
5. Finally, when terminating switch in the Recipient PSTN network finds its own routing information, by analysing the supplied RN, it uses received CdPN to terminate the call.

For PSTN originating calls, the steps 2, 3, 4 and 5 would be identical to above described.

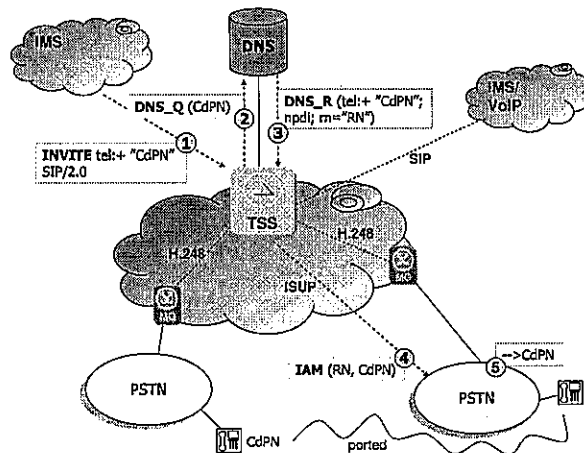


Figure 5: ENUM based NP: Number ported between PSTNs

C. The number is ported between PSTN and IP network

For calls originating from IMS network to a subscriber that is ported from PSTN to IP network, the call scenario is depicted on fig 6.

1. The softswitch receives SIP INVITE message from the IMS network with Request-URI that points to a called user identity described via tel URI. The tel URI contains CdPN of the called subscriber. For example:

```
INVITE tel:+1-202-555-1234 SIP/2.0
```

2. Based on provided CdPN, a domain name is created and used for NAPTR query towards external DNS.

```
4.3.2.1.5.5.5.2.0.2.1.e164.arpa.
```

3. At receive of DNS response, the softswitch applies selection of those NAPTR RRs that support E2U resolution service. For example, a NAPTR record for ported number, using a 'sip' URI scheme, may look like:

```
$ORIGIN 4.3.2.1.5.5.5.2.0.2.1.e164.arpa.
NAPTR 10 100 "u" "E2U+pstn:tel"
"!^.*$!sip:+1-202-555-1234;npdi;
rn=+1-208-555-0199@cd.network.com;
user=phone!".
```

The method of conversion from a tel URI to a SIP URI is demonstrated in [4], as well as in [8].

As routing number ('rn') is present in SIP URI, the subscriber is considered as ported. The 'npdi' field is included in order to prevent subsequent NP lookups in legacy-style PSTN databases.

4. The softswitch sets up a SIP call to cd.network.com domain by sending INVITE message with obtained 'rn' and 'npdi' indicator. Note that additional DNS lookup to convert domain name into IP address of next hop to contact is not presented.

```
INVITE sip:+1-202-555-1234;
rn=+1-208-555-0199@cd.network.com;
user=phone;npdi=yes SIP/2.0
```

For PSTN originating calls, the steps 2, 3 and 4 would be identical to above described.

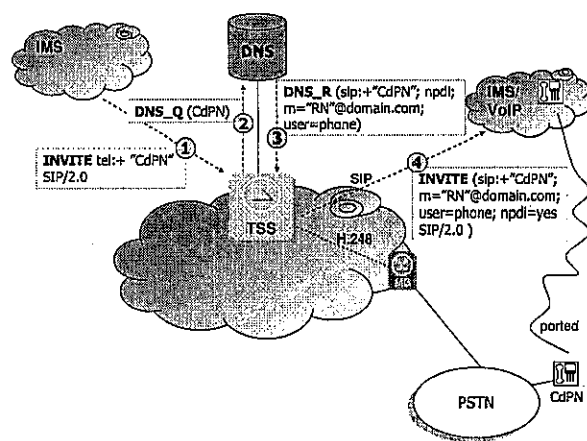


Figure 6: ENUM based NP: Number ported between PSTN and IP Network

D. The number is ported within IP network

For calls originating from IMS network to a subscriber that is ported within the IP network, the call scenario is depicted on Fig 7.

1. The softswitch receives SIP INVITE message from the IMS network with Request-URI that points to a called user identity described via SIP URI. The tel URI that is carried as optional parameter in the user portion of the sip URI contains CdPN of the called subscriber resident at ab.network.com. For example:

```
INVITE sip:+1-202-555-1234@ab.network.com;user=phone SIP/2.0
```

2. Based on provided CdPN, a domain name is created and used for NAPTR query towards external DNS.

```
4.3.2.1.5.5.5.2.0.2.1.e164.arpa.
```

3. At receive of DNS response, the softswitch applies selection of those NAPTR RRs that support E2U resolution service. For example, a NAPTR record for ported number, using a 'sip' URI scheme, may look like:

```
$ORIGIN 4.3.2.1.5.5.5.2.0.2.1.e164.arpa.
NAPTR 10 100 "u" "E2U+pstn:tel"
"!^.*$!sip:+1-202-555-1234;npdi;
rn=+1-208-555-0199@cd.network.com;
user=phone!".
```

As routing number ('rn') is present in SIP URI, the subscriber is considered as ported. The 'npdi' field is included in order to prevent subsequent NP lookups in legacy-style PSTN databases.

4. The softswitch sets up a SIP call to cd.network.com domain by sending INVITE message with obtained 'rn' and 'npdi' indicator. Note that additional DNS lookup to convert domain name into IP address of next hop to contact is not presented.

```
INVITE sip:+1-202-555-1234;
rn=+1-208-555-0199@cd.network.com;
user=phone;npdi=yes SIP/2.0
```

For PSTN originating calls, the steps 2, 3 and 4 would be identical to above described.

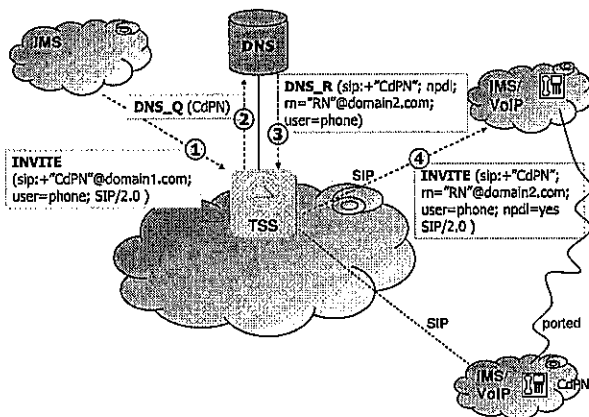


Figure 7: ENUM based NP: Number ported within IP Network

VII. CONCLUSION

When the two network technologies, PSTN and IP, are interconnected, a need for number portability across different network types (i.e. VoIP and IMS) came forth. The existing number portability solutions in the PSTN provide solutions for numbers which are ported from PSTN domain to IP domain. Limited number portability can be implemented in IP domain by call forwarding.

The ENUM service facilitates interconnection of systems that rely on telephone numbers with those that use URIs to route transactions. It has all potential to become a solution for number portability for interconnected VoIP and IMS networks.

Basic ENUM specification has reached RFC standards track stage and commercial ENUM implementations are already available. However, these implementations do not contain number portability capabilities across network types. The recent achievements in IETF enum working group go in direction that looks promising for ENUM to be used as a common number mapping database with full support for number portability. The call models presented in this paper also speak in favor of ENUM.

The future work in this area will be related to populating information in ENUM, that must be kept up-to-date to avoid wrongly or inefficiently routed calls. Currently the update is performed manually, and the process is uncoordinated between service providers. This becomes a burden, especially when number portability causes increased update frequencies. Also the risk of wrong and incompatible information is high. An automated approach for synchronizing information between providers is needed.

REFERENCES

- [1] RFC 2396, Uniform Resource Identifiers (URI): Generic Syntax, August 1998.
- [2] RFC 2915, The Naming Authority Pointer (NAPTR) DNS Resource Record, September 2000.
- [3] RFC 2916, E.164 number and DNS, September 2000.
- [4] RFC 3261, SIP: Session Initiation Protocol, June 2002.
- [5] RFC 3263: Session Initiation Protocol (SIP): Locating SIP Servers, June 2002.
- [6] RFC 3482, Number Portability in the Global Switched Telephone Network (GSTN):An Overview, February 2003.
- [7] RFC 3761: The E.164 to Uniform Resource Identifiers (URI) Dynamic Delegation Discovery System (DDDS) Application (ENUM), April 2004.
- [8] RFC 4694, Number Portability Parameters for the "tel" URI, October 2006.
- [9] RFC 4769, IANA Registration for an Enumservice Containing Public Switched Telephone Network (PSTN) Signaling Information, November 2006.
- [10] ITU-T Q. Supplement 5, Number portability – Capability set 2 requirements for service provider portability (Query on release and Dropback), 03/99

