TRIP, ENUM and Number Portability

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Abstract

This paper describes the problem of locating terminals using E.164 numbers, and the problem of selecting a suitable gateway for calls from an IP telephony network to the public switched telephone network (PSTN). Generally, these are the problems of mapping the name of a destination into an address, and to find the best route to the destination in a combined IP and PSTN network. Number portability is closely related to these problems. Due to number portability the address of a destination is changed without changing its name. Number portability may also change the optimal route to a destination.

Two protocols are being developed by the Internet Engineering Task Force (IETF) to solve these problems. The Telephony Routing over IP (TRIP) protocol solves the gateway location problem by distributing routing information between entities on the IP network. The tElephony NUmbering Mapping (ENUM) provides a solution to the terminal location problem based on DNS. ENUM maps an E.164 number into an URI, which is used to locate the end point. Both protocols aim to add a part to the architecture in order to make a global hybrid PSTN-IP network possible. They also aim to enable number portability in the IP network and between the two types of networks.

In this paper, we will introduce the concept of terminal and gateway location. We describe how the current protocols locate terminals and gateways, and what the problems with the current solutions are. The TRIP and ENUM protocols are presented in detail and scenarios based on the protocols are described. Solutions to number portability are presented and some problems are discussed.

Keywords: Voice over IP, IP Telephony, TRIP, ENUM, Number portability, Routing, Address mapping

1 Introduction

When IP telephony was introduced, it was mainly used in small private networks, which were connected to the public switched telephone network (PSTN) through a single gateway. As IP telephony matured, a vision of global public IP telephony became popular. In this scenario the IP and PSTN networks are interconnected with a large number of publicly available gateways. In order to make connectivity between all IP- and PSTN-terminals possible, the problem of terminal and gateway location must be solved. In these problems, addressing is central. Number portability allows users to change operators and locations without changing the telephone number. To smoothen the transition to an IP-based telephone network, number portability is also required between PSTN and IP-networks.

The main signaling protocols for IP telephony are Session Initiation Protocol (SIP) [1] and H.323 [2]. The architectures that they define are similar, although different names are used for the network elements. Calls can be established between IP telephony terminals directly, but usually the call setup signaling passes through a gatekeeper (in H.3232) or signaling server (in SIP). The elements have similar functions in the two signaling protocols, so in this paper we will use the name signaling server for both. Some important functions of the signaling server are address translation and location of the destination terminal. For calls between an IP terminal and a terminal on the PSTN, a gateway is used to convert signaling and code the voice stream between the circuit switched and the packet network.

To identify the destination of a call on the PSTN, the caller dials the receiver's telephone number in E.164 format [3]. The telephone number is analyzed digit by digit to locate the path through switches in the PSTN towards the destination. Although the number dialed by the user traditionally is used for routing, a different address is used for routing in many cases.

The European Telecommunications Standards Institute (ETSI) defines the concept of names and addresses as follows: A name is a combination of alpha, numeric or symbols that is used to identify end-users. An address is a string or combination of digits and symbols which identifies the specific termination points of a connection/session and is used for routing. [4]

The main difference between these is that a name is an identifier for the end user, while an address is a locator. An address should typically have some form of structure that allows aggregation for routing purposes. In the Internet, the name is a domain name. The domain name is mapped into an IP address, which is used for routing. In the telephone network, E.164 numbers have traditionally been used as both names and addresses. However, due to number portability their roles have been separated. The number that the user dials, which can be regarded as a name, is then mapped into a routing number, which is an address. The dialed number is usually referred as a directory number. It is also worth noting, that in many cases entities that functionally are names are called addresses.

To transform the name into an address some type of mapping method is needed. For the mapping of host names into IP addresses, the Domain Name Service (DNS) [15], [16] is used. DNS is a distributed directory service based on DNS servers. Each server knows the mapping of a range of hosts, or the address to a server that has more detailed information. The parts of the domain names are analyzed in hierarchical order and the mapping request is forwarded to more specific DNS servers until the mapping can be completed.

2 The current situation

Today IP telephony is used in mainly two situations: either as a private branch of the PSTN within an organization or for calls between terminals on the Internet. The first case involves gateways, which connects the private IP telephony network with the public switched telephone network. The second case does not usually involve gateways, since it would require publicly available gateways and the existence of a billing system. Additionally IP telephony is used to inexpensively transport long distance calls between PSTN callers through the IP network.

2.1 Locating the destination

In most of today's IP telephony applications, the IP telephony network acts as a branch of the PSTN. The gateway together with the signaling server (or gatekeeper) works like a PBX from the viewpoint of the PSTN. The called E.164 numbers are translated to IP addresses by the signaling server. Calls to and from external numbers are routed through the gateway. In such small networks with only a few gateways, the number translation and gateway selection can be performed by a single signaling server. The mappings are usually configured in the signaling server. The users do not necessary notice that IP telephony is used, since they use E.164 numbers as normally. These types of networks are, however, very limited in size and cannot be considered in a larger deployment of IP telephony. In this paper, we will mainly discuss the use of larger IP telephony networks.

For calls over the public Internet, the situation is more complicated. IP terminals are located using their IP address or host name. The signaling protocols allow various formats of addresses to be used by the users. Users prefer to use E.164 or e-mail type addresses that are familiar from traditional telephony or e-mail, respectively. To set up a call, the name of the destination is mapped to an IP address by a signaling server. The signaling server can be manually configured with the mappings for its local terminals. More usually however, the terminals must register to the signaling server. Based on registration the mapping is created. The server maintains a database of mappings for its registered clients.

The SIP architecture also includes a network element named location server. The location servers store the mappings on the behalf of the signaling servers. A location server may be used by a number of signaling servers. The location server may also be integrated with a signaling server. In this way we can generalize to say that the location server stores the mapping, even though the location server and signaling server in some cases are the same element. In case of separate servers, the information is accesses with some directory access protocol.

In a public IP telephony network there is a large number of signaling servers. There is currently no method to distribute the mappings between different servers. Because of this lack of distribution method, mappings can only be used for calls between phones registered to the same server. Thus, E.164 numbers do not work for calls between terminals registered to different signaling servers or location servers. Note that calls are always possible when the address is given as an IP address.

The SIP protocol also supports the use of names given as Universal Resource Locators (URL) [18]. The URL specifies the user, the host and other parameters. This "user@host" format can be handled in a similar way as email addresses are handled by SMTP [5]. An IP address of a signaling server for the domain is located using DNS. Thus, the host name does not have to be a complete host name. The call can be further forwarded by proxy or redirect servers. [1]

The most popular H.323 client Microsoft Netmeeting uses directory servers to locate users. These directories are propriety solutions, named ULS and ILS. Similar solutions are used by many other clients on the market. Still, there are significant drawbacks in this type of solution. The directories have a limited capacity and they do not exchange information with each other. Furthermore, many of them use non-standardized access protocols. [7]

2.2 Locating the gateway

For calls to the PSTN a gateway must be used. Today, the gateways are in most cases manually configured into the signaling server. A signaling server has a set of available gateways to use for external calls. For private internal IP telephony networks, external numbers are usually recognized by a preceding “0”.

In SIP the call can be set up using a gateway specified in the URL. The destination is then given as “number@gateway”. This requires the user to know that the destination is on the PSTN and also which gateway should be used. If the gateway is down or if all lines are busy, the user must manually select another gateway. Another method is to let the signaling server choose the gateway, whereas calls can be made by only giving a number. The server selects one from its list of available gateways. The H.323 protocol works in a much similar way.

2.3 Number portability

Number portability allows a user to change service providers, location or service type without changing the telephone number. Service provider portability is mandatory in many countries. The introduction of IP telephony adds a new type of number portability: between different network types. [8]

Today number portability is only implemented on the PSTN. The implementation of number portability differs in different countries. Common to all implementations, is that the directory number dialed by the customer is mapped to either a routing number or a routing prefix. A routing number is a hierarchical routing address, which can be digit-analyzed to reach the correct country, network provider, end-office switch and subscriber line. A routing prefix forms an routing number by adding some digits in front of the directory number. The routing number replaces the hierarchy that is lost, since the directory number space becomes flat due to number portability.

Most number portability solutions utilize Intelligent Network (IN) functions. In these solutions, the mapping database is stored in the Service Data Functions (SDF) elements. Depending on the implementation, the call may have to pass through the previous operator’s network before it reaches the current destination network. [8]

At the time being, there is no specific solutions for number portability across the network types. A number that is moved to the IP network can be handled in a normal way in the number portability solutions on the PSTN. The number portability databases are updated with a routing number that directs calls to a gateway. Limited number portability can be implemented in IP telephony network using redirect and proxy servers. Calls to a moved numbers are forwarded to the 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.

3 Problem description

3.1 Naming

Traditionally E.164 numbers have been used on the telephone network and e-mail type addresses of format “user@domain” on the Internet. The signaling protocols SIP and H.323 allows using multiple types of names, including both the above methods as well as IP addresses. For Internet users, who have a keyboard available, textual names are preferred since they are easy to remember and deduce.

However, the problem arises when the networks are interconnected. Callers on the PSTN have no keyboard and a scheme for entering characters using number keypad would be too complicated. This limits the PSTN users to entering numeric names. Consequently, an IP terminal must have a telephone number to be accessible from the PSTN. The problem was recognized by TIPHON, which chose to equip IP-terminals with an E.164 number. For calls within the IP network, other types of addressing can be used. Unfortunately, this would require the user to know on what type of network the destination is. When IP telephony is largely deployed, customers do not necessary even know the underlying technology of their own connection.

As we saw in section 2.1, E.164 numbers can currently only be used between host registered to the same signaling server. Using some propriety protocol, mapping can be distributed between smaller groups of servers, but there is no protocol for global distribution.

3.2 Problem categories

The name entered by the user, usually given as an E.164 number, must be mapped to at least one routing address. For calls from IP telephony terminals to other IP telephony terminals, the host address of the destination must be found. For calls over the network boundary to the PSTN, a gateway must be located. Also in the opposite direction, from the PSTN to the IP network, a gateway must be selected.

The TRIP framework [1] divides the problem into three subproblems:

1. Given a phone number corresponding to a specific host on the IP network, determine the IP address of the host. This is required for calls from the PSTN to the IP network, but also for calls within the IP network if E.164 numbers are used. The mapping may be variable, for example if DHCP is used.
2. Given a phone number corresponding to a terminal on the PSTN, determine the IP address of a gateway capable of completing calls to that phone. The choice is influenced by a number of factors, such as policies, location, availability and features. This is called the gateway location problem.
3. Given a phone number corresponding to a user of a terminal on the PSTN, determine the IP address of an IP terminal owned by the same user. This type of mapping may be used if the PC services as an interface for the phone, for example for delivering a message to the PC when the phone rings.

For calls from the PSTN to the IP network, the selection of gateway is performed using normal routing in the switched circuit network, which is static. On longer sight, it would also be necessary to dynamically select a gateway for these calls. This gives us a fourth subproblem.

3.3 The address mapping problem

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, a new mapping is required: from an E.164 name to an IP address. The address mapping problem usually refers to the task of locating terminals on the IP network.

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

The required mappings could be solved with some type of directory. At a minimum, the mapping from E.164 number to network type and IP address must be supported. The directory must be scaleable too store large amounts of mappings, possibly for all telephones in the world. It must be capable to reply to a high rate of lookups, for each call that is set up. In practice, the directory must therefore be distributed. The directory must also propagate updates rather quickly when the information changes.

Additionally the mapping is expected to be used with several different services. In addition to voice calls, the IP network allows for video conferencing and e-mail among others. Some method of locating the available contact modes and services is desired.

3.4 Routing and number portability

For economic or quality related reasons a transit network of different type can be used, giving two more call scenarios: PSTN-IP-PSTN and IP-PSTN-IP. Even when only two network types are used, the transit network must be selected. It is usually more cost effective to hand over calls to IP destinations to the IP network near the origination point. On the other hand, the voice quality is better if the call uses PSTN most of the path. Possibly the caller could choose whether to route the calls via IP or PSTN using carrier selection mechanisms. Typically this would imply the use of a prefix to select carrier. [23]

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 decrease the quality. Therefore, unnecessary media conversions should be avoided. It would be good to know the type of the destination network already in the originating network.

With number portability numbers may 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 network, calls from IP subscribers may unnecessarily be routed through the PSTN.

3.5 The gateway location problem

As the usage of IP telephony grows and the number of gateways increases, the management of gateways and routes between the IP- and PSTN networks becomes increasingly complex. In a situation where the IP network approaches the size of the PSTN, a large part of the calls will pass through one or even several gateways on their path. For calls from the IP network to the PSTN, the caller must locate a gateway that is able to complete calls to the desired destination. There may be several available gateways, and selecting the most suitable one is a nontrivial process.

Currently the gateway must be selected by the user or by the signaling servers. The selection and configuration of gateways to use involves manual work. The list of available gateways must be configured into the signaling servers and updated when new gateways become available. Additionally, gateways may become blocked when all lines are in use. The signaling server do not know which gateways are accessible.

Connectivity to the PSTN means that every gateway is able to connect to nearly any terminal on the PSTN. The number of available gateways can thus be very large. The selection of which gateway to use is influenced by a number of factors. Firstly, the location of the gateway is important. For example, there is no reason to use a gateway in a country far away to connect parties in the same city. To minimize usage of resources it is important that the gateway is near the path between the parties.

Secondly, business relationships are important. The gateway service involves costs when calls are completed to PSTN destination. Gateway providers, in most cases, want to charge for using their gateways. Because of this, the usage of gateways may be restricted to the groups of users that have some type of established relationship with the gateway provider. The end user will probably not pay for the gateway service directly. Instead, the end user may have a relationship with an IP telephony service provider (ITSP). The ITSP may have own gateways or use the gateways of a separate gateway provider. All these policies and relationships influence in the selection of gateway.

Additionally, the end user may have requirements on the gateway. The end user may prefer a certain provider or require a specific feature. The caller may use a specific signaling protocol or media codec that is supported by only some gateways.

Keeping in mind that also the gateway capacity is limited, it is obvious that an automatic method for gateway selection is required. Since the selection is largely driven by policies, some type of global directory of gateways is not suitable. Instead, a protocol for exchanging gateway information between the providers would be a better solution.

4 Telephony Routing over IP

To solve the gateway selection problem, the Internet Engineering Task Force (IETF) working group IP Telephony (IPTEL) began working on a protocol for distributing gateway information between gateway providers and IP telephony providers. The protocol was first called Gateway Location Protocol (GLP) but after finding the problem larger than merely locating gateways, the protocol was renamed to Telephony Routing over IP (TRIP). The most important documents of the work are the TRIP framework [6] and the draft protocol specification [9].

The working group found that a global directory for gateway information is not feasible. The selection of gateway is in large part driven by the policies of the parties along the path of the call. Gateway information is exchanged between the providers and depending on policies, made available locally and propagated to other providers. The providers create their own databases of reachable phone numbers and the routes towards them. These databases can be different for each provider.

TRIP is modeled after the Border Gateway Protocol 4 (BGP-4) [10]. TRIP is like BGP-4 an inter-domain routing protocol driven by policies. The nodes of TRIP are the location servers (LS), which exchange information with other location servers. The information includes reachability information about telephony destination, the routes towards these destinations and properties of the gateways connecting the PSTN and IP network.

TRIP uses the concept of Internet Telephony Administrative Domains (ITAD) in a similar way as BGP-4 uses autonomous systems. The location servers that are administered by a single provider form an ITAD. The ITAD may contain zero or more gateways. The border of the ITAD does not have to correspond to the border of an autonomous system. The main function of TRIP is to distribute information between ITADs, but TRIP also contains functions for inter-domain synchronization of routing information. It is not required that all ITADs in the world are connected. Groups of ITADs can be formed that exchange information with TRIP.

TRIP connects location servers with administratively created peer relationships. The location server forwards the information received from one peer to the other peers. Hereby the location servers in one ITAD learn about gateways in the other ITADs. The location server selects the routes to use in its own domain, and the routes to forward to neighboring domain according to its local policies. The information can be modified according to the policies before it is forwarded. In this way, the provider can control the type of calls passing through the domain.

The location servers collect information and use it to reply to queries about routes to destinations. The query protocol is not defined by TRIP. Any directory access protocol can be used, for example LDAP [11].

4.1 Operation of TRIP

The TRIP protocol, the structure and operation of a node, and the implementation details are specified in the TRIP specification draft [9].

TRIP location servers process three types of routes:

1. External routes received from external peers.
2. Internal routes received from another location server in the same ITAD.
3. Local routes which are locally configured or received from another routing protocol.

The routes are stored in the Telephony Routing Information Base (TRIB), whose structure is depicted in Figure 1. The TRIB consists of four distinct parts.

1. The Adj-TRIBs-In store routing information that has been learned from other peers. These routes are the unprocessed routed that are given as input to the decision process. Routes learned from internal location servers and from external location servers are stored in separate Adj-TRIBs-In.
2. The Ext-TRIB stores the preferred route to each destination, as selected by the route selection algorithm.
3. The Loc-TRIB contains the routes selected by applying the local policies to the routes in the internal peers' Adj-TRIBs-In and Ext-TRIB.
4. The Adj-TRIBs-Out store the routes selected for advertisement to external peers.

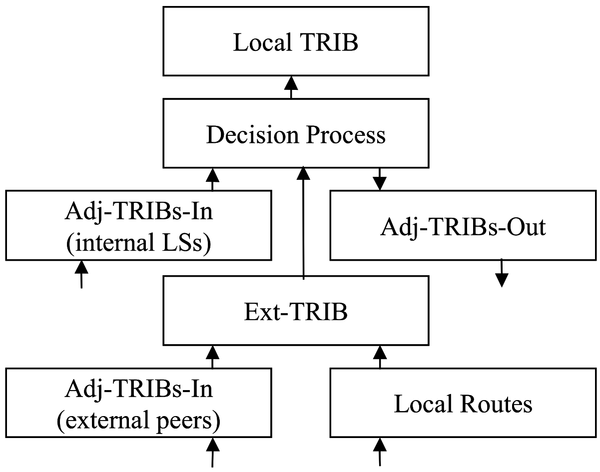

Figure 1: Structure of a TRIP node

TRIP uses the same state machine and the same messages as BGP-4. The messages are the OPEN message for establishing peer connection and exchanging capability information, the UPDATE message for exchanging route information, the NOTIFICATION message for informing about error conditions, and finally the KEEPALIVE message for ensuring that the peer node is running.

The routing information is transmitted in attributes of the TRIP messages. The specification includes a set of mandatory well-known attributes. In addition to the well-known and mandatory attributes, optional attributes can be added to allow for expansion. Gateways have many properties that may need to be advertised, so the expected large number of expansion attributes must be handled correctly. An attribute flag indicates how a location server handles a message that it does not recognize. The flag can take a combination of the values optional, transitive, dependent, partial and link-state encapsulated.

The specification [9] defines the basic set of attributes shown in Table 1. Additional attributes are defined in separate drafts. An authentication attribute is defined in [12] and a service code attribute is defined in [13].

Table 1: The basic set of TRIP attributes

Name	Description
Withdrawn routes	List of telephone numbers that are no longer available.
Reachable routes	List of reachable telephone numbers.
Next hop server	The next signaling server on the path towards the destination.
Advertisement path	The path that the route advertisement has traveled.
Routed path	The path that the signaling messages will travel.
Atomic aggregate	Indicates that the signaling may traverse ITADs not listed in the routed path attribute.
Local preference	The intra-domain preference of the location server.
Multi exit disc	The inter-domain preference of the route if several links are used.
Communities	For grouping destinations in groups with similar properties.
ITAD topology	For advertising the ITAD topology to other servers in the same ITAD.
Authentication	Authentication of selected attributes.

The advertisements represent routes toward a gateway through a number of signaling servers. A route must at least contain the following attributes: withdrawn routes, reachable routes, next hop server, advertisement path and routed path. For an advertised route, the withdrawn routes attribute is empty. The reachable routes attribute contains the list of telephone number ranges belonging to this route, and the corresponding application protocol. The next hop server is the next server that signaling messages are sent to. For the final hop, it contains the address of the gateway. The advertisement path is the path that this advertisement has traveled through and the routed path is the path for the signaling. These paths are lists of ITADs. They are mainly used by the policy to select routes containing, or not containing specific ITADs.

4.2 TRIP for gateways

The TRIP framework [1] leaves the question open, how the location servers learn about the gateways. Usually the register message of SIP has been suggested. However, the draft [14] points out the weaknesses of using the register message and suggests that a subset of TRIP could be used to export routing information from gateways and soft switches to location servers. TRIP manages the needed information transfer and keep-alives more efficiently than other protocols and can better describe the gateway properties. Two new attributes are proposed: circuit capacity for informing about the number of free PSTN circuits, and DSP capacity for informing about the amount of available DSP resources. Because of their dynamic nature, these are only transmitted to the location server that manages the gateway, and are not propagated.

A more lightweight version of TRIP can be used in the gateways. Since the gateway does not need to learn about other gateways, it operates in send-only mode. It neither needs to create any call routing databases. This stripped down version, called TRIP-GW, is still interoperable with normal TRIP nodes. Nevertheless, due to scalability problems it is recommended that location servers peering with gateways run a separate TRIP instance for TRIP-GW peers.

5 Telephone Number Mapping

While TRIP is carrying routes to destinations on the PSTN, a method for locating terminals on the IP network is still required. This problem is simpler than the gateway location problem, since the amount of information describing a terminal is less than the information about a gateway. TRIP could be used also for this purpose, but the complexity of it is not needed. A simpler directory can be used. It has been suggested that Domain Name System (DNS) [15], [16] could be used. An IETF working group called ENUM (tElephone NUmber Mapping) was established to specify the number mapping procedures.

DNS is used to map domain names into IP addresses. By constructing a domain name from the E.164 number, the DNS system can be used to map telephone numbers into IP addresses. More generally, the result of an ENUM lookup is a Uniform Resource Identifier (URI) [18], which contains the signaling protocol and the host name. An additional DNS lookup is thus required to map the host name to an IP address. The procedure is described in RFC 2916 [17], the main document specifying the ENUM service.

ENUM uses the domain “e164.arpa” to store the mapping. Numbers are converted to domain names using the scheme defined in [17]. The E.164 number must be in its full form, including the 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 +358-9-4515303 into the host name “3.0.3.5.1.5.4.9.8.5.3.e164.arpa”.

DNS stores information in different types of records. The Naming Authority Pointer (NAPTR) record [19] 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. The fields of the NAPTR record are shown in Table 2. 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.

Table 2: Fields of the NAPTR record

Name	Description
Order	The order in which records are processed if a response includes several records.
Preference	The order in which records are processed if the records have the same order value.
Service	The resolution protocol and resolution service that will be available if the rewrite of the regexp or replacement field is applied.
Flags	Modifiers for how the next DNS lookup is performed.
Regexp	Used for the rewrite rules.
Replacement	Used for the rewrite rules.

Figure 2 shows some example 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 [20]. The result of the rewrite of the NAPTR record is a URL, as indicated by the “u” flag. The own resolution methods of SIP and SMTP are used. In case of SIP, the result is a SIP URI, which is resolved as described in [1]. In case of the “tel” scheme, the procedure is restarted with a new E.164 number.

```
$ORIGIN 3.0.3.5.1.5.4.9.8.5.3.e164.arpa.
  IN NAPTR  10   10  "u"  "sip+E2U"
     "!^.*$!sip:nbeijar@tct.hut.fi!"    .
  IN NAPTR  100  10  "u"  "mailto+E2U"
     "!^.*$!mailto:nbeijar@tct.hut.fi!" .
  IN NAPTR  100  10  "u"  "tel+E2U"
     "!^.*$!tel:+35894515303!"     .
```

Figure 2: Example NAPTR records

The draft [21] describes a telephone number directory service based on ENUM. The model is divided into four levels.

The first level is a mapping of the telephone number delegation tree into authorities, to which the number has been delegated. The hierarchical structure of DNS is used, and the mapping may involve one or several DNS queries, which are transparent from the user's point of view. The delegation maps the hierarchy of the E.164 number to the DNS hierarchy, using the country codes, area codes and other parts of the number. The first level mapping uses name server (NS) resource records in DNS.

The second level is the delegation from the authority, to which the number has been delegated, to the service registrar. The registrar maintains the set of service records for a given telephone number. Since there may be several service providers for a given number, the registrar has the role to manage service registrations and arbitrate conflicts between service providers. The second level uses the DNAME and CNAME records of DNS to provide redirection from the designated authority to the service registrar. The delegated authority and the service registrar can be the same entity, which is anticipated especially in the early stages of ENUM deployment.

The third level is the set of service records, which indicate what services are available for a specific telephone number. There can be multiple records for the same service, indicating competitive or redundant service providers. The NAPTR type of records is used at the third level. The response to a client's query is a set of NAPTR records, and the client is responsible for selecting the service to use for the intended action. A URI is obtained by rewriting the query using the rewrite rule. The URI can be an LDAP directory server, a H.323 gatekeeper, a SIP signaling server or a specific end point address.

Finally, a fourth level can be provided if necessary. This level provides specific attributes for the services that are only known by the provider of the service. Such attributes can be needed for placing calls, routing messages or validating capabilities. The attributes can be obtained through a SIP query to a signaling server or a LDAP query to a directory server. The level is service specific and dynamic, and should therefore be possible with minimal coordination between the directories of competing providers.

6 Scenarios

In this section, some scenarios based on ENUM and TRIP are presented. First the different types of resource records used by ENUM are presented through an example. Then two call setup situations are analyzed. The draft [22] describes how ENUM can be used in different call setup situations where interworking between the PSTN and IP-based networks is necessary. By additionally using the TRIP framework [4] and the ENUM model [17][21], we construct examples of how the protocols are used. Since any final call setup procedure is not defined, these examples only represent one possible approach for interworking between the networks.

6.1 Call setup using ENUM

To illustrate the use of ENUM, we will study a call setup situation, where the DNS records of Figure 3 are used. The figure shows the DNS configuration for the top level delegations, the national delegations, a service provider and a service registrar.

Sample top level delegations from ITU:

```
3.3.e164.arpa    IN NS ns.FR.phone.net.  ;France
8.5.3.e164.arpa  IN NS ns.FI.phone.net.  ;Finland
```

Sample national delegations:

```
5.4.9.8.5.3.e164.arpa.   IN NS ns.ServiceProviderX.net.
```

Sample service provider's configuration:

```
1.5.4.9.8.5.2.e164.arpa. DNAME 1.5.4.9.8.5.2.ns.hut.fi.
```

Sample service registrar configuration:

```
*.1.5.4.9.8.5.2.ns.hut.fi.
    IN NAPTR 100 10 "u" "ldap+E2U"\
    "$!ldap://ldap.hut.fi/cn=\1!" .
```

Figure 3: Configuration of DNS records

The described service enables an end-user to discover the various methods by which the recipient can be reached. The service is hosted by the recipient's corporation.

When a call is setup to the telephone number +35894515303, the number is first translated into the domain name "3.0.3.5.1.5.4.9.8.5.3.e164.arpa" according to the ENUM rules. Using the NS records in the top-level and national authorities' databases, the service provider is located. In this example, the number block +3589451xxxx is delegated to the service provider. The service provider provides a non-terminal redirection pointer to the corporation, which is the service registrar for the number block +35894515xxx. The query for the reachme service returns the NAPTR record. The client then applies the regular expression and gets an LDAP URI of "LDAP://ldap.hut.fi/cn=35894515303". The client uses LDAP with the reachme schema to determine the available communications methods.

6.2 Calls from PSTN to IP-based network

The call setup scenario for a call from PSTN to an IP based network is depicted in Figure 4. The originating customer, who resides on the PSTN, dials an E.164 number. The PSTN operator forwards the call to an appropriate gateway. The selection of gateway depends on several factors. This is a gateway location problem similar to that on the IP network, but there are currently no corresponding solutions like TRIP. The draft [22] leaves the question open.

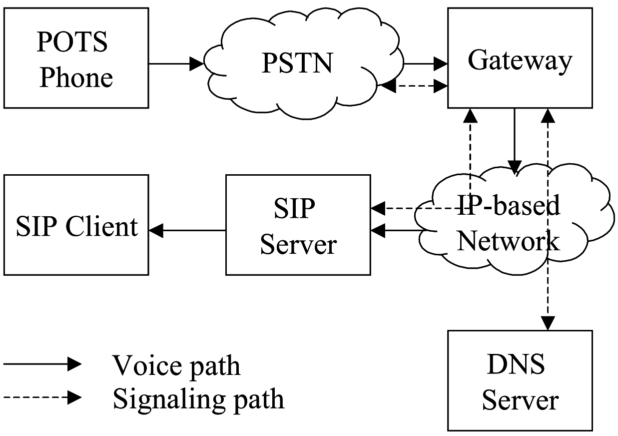

Figure 4: Call from PSTN to IP-based network

The gateway, which contains ENUM functionality, looks up the number in DNS. The dialed number is mapped into an URI. If necessary, the country and area codes are added to the number by the gateway. The DNS returns any service records that are associated with the URL. The record may be a SIP URI such as "sip:nbeijar@sipserver.hut.fi". The gateway makes a DNS query for the host, in this case "sipserver.hut.fi" to get the IP-address of the signaling server. The SIP call can then be established to the user agent of the given user.

6.3 Calls from IP-based network to PSTN

A call setup scenario for a call from an IP-based network to the PSTN is illustrated in Figure 5. The originating customer dials an E.164 number. Since a customer may dial a local number or a national number, the client must be capable of supply any missing digits. Here the caller uses a SIP client, but any other signaling protocol can be used. The client must contain ENUM functionality. A DNS request is constructed from the dialed digits according to the ENUM specification.

When the client looks up the name in DNS, the DNS returns any NAPTR service records associated with the URL. Since the destination is on the PSTN, the query only returns one record containing the URI in the "tel" format. For example the URI "tel:+35894515303" might be returned. The SIP client initiates an INVITE to the SIP server using the given URI. The SIP server queries a location server with LDAP or any other front end protocol suggested in the TRIP framework [4]. The location server has learned about available gateways using TRIP. The location server returns the IP address of a suitable gateway and the call is routed to the gateway by the SIP server. The gateway then completes the call through the PSTN.

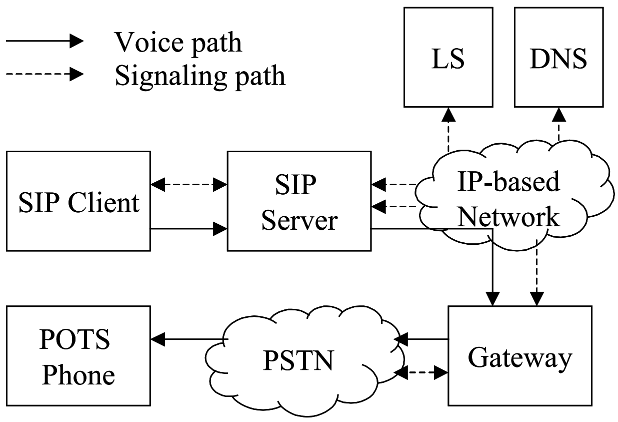

Figure 5: Call from IP-based network to PSTN

7 Solutions for number portability

Number portability requires a mapping between name and address. Generally numbers can be moved by changing the mapping. The described protocols TRIP and ENUM both provide a mapping between name (telephone number) and address (URI, next step signaling server or gateway).

7.1 ENUM and number portability

ENUM provides a solution for number portability for numbers moving within the IP network and to the PSTN. This allows users to change IP service providers without having to change their telephone number [22]. The directory service solution defined in [21] describes number portability on three of the conceptual levels of ENUM.

If the number is delegated to another authority, the corresponding update is performed in ENUM by changing the name server records to point to the new authority. The number is thus moved to another service provider or to a portability authority.

The service registrar can be reassigned, for example if the customer wants to also change the service registrar in conjunction with the change of service provider. The DNAME or CNAME records are then updated to point to the new service registrar. New service specific NAPTR records are created by the new service registrar.

Most frequently the movement of a number can be accomplished by changing the NAPTR records. This happens when a specific service is moved from one provider to another, for example when switching telephony providers. The service registrar coordinates the deletion of the old records and insertion of records for the new service provider.

7.2 Interworking and number portability

ENUM also solves number portability for hybrid PSTN-IP networks. The draft [22] separates three scenarios:

1. The number moves within the PSTN.
2. The number moves between PSTN and IP.
3. The number moves within the IP network.

For each scenario, the call setup procedure from both PSTN and the IP network is described.

For calls originating from the PSTN, the first scenario is already handled by today's number portability. The second scenario is solved by changing the number portability mapping to direct the call to a gateway. The third scenario is solved by ENUM as was described.

For calls established on the IP network, the first scenario may lead to inefficient routing. As the number moves, within the PSTN, the most suitable gateway probably changes. As a result, the DNS information must be updated. It is still not defined how this is done, and how it can be automated. Alternatively, if the DNS contains routing addresses (such as LRN) for PSTN destinations, these must be updated to point to the new operator and new gateway. Otherwise calls may be routed to the wrong operator. If the gateway do not have routing addresses available, an IN query must be performed by the gateway or at a later stage.

In scenario 2 for IP originated calls, it would be enough to update the type of URI returned by DNS. A "tel:"-based URI would be replaced by an URI for a SIP or H.323 terminal, or vice versa. Also the third scenario is solved by updating ENUM information.

The question about whether to store routing or directory numbers of PSTN terminals in DNS has been discussed in IETF working groups. It is also unclear how to know which terminals reside on the PSTN. In current plans, mainly mappings for IP terminals are stored in DNS. It is assumed that calls to unknown E.164 numbers are routed to the PSTN. This may create unnecessary traffic and gateway blocking due to wrongly dialed numbers.

7.3 CTRIP

As we saw in the two first scenarios, both the mappings in ENUM and in the IN databases on the PSTN must be updated in some cases. The question how the update is coordinated and how the information is transferred is still unresolved. Moreover, the information of TRIP must be updated in some cases. There is still no solution how to coordinate information in ENUM, TRIP and the IN databases.

It seems to be necessary to automate the distribution of numbering information between the network types and between the protocols. To solve the problem, a counterpart to TRIP is being developed in the Networking Laboratory at Helsinki University of Technology. The suggested protocol, named Circuit Telephony Routing Information Protocol (CTRIP), automates the distribution of routing information between operators and network elements. Information is exchanged with other protocols in Numbering Gateways. [24]

8 Conclusions

Although the signaling protocols provide basic mechanisms for locating terminals and gateways, new protocols are required for distributing routing information in order to make a global IP based telephone network possible. TRIP provides a solution for the gateway location problem by distributing information about gateways and reachable PSTN destinations between location servers. ENUM defines a directory of name to address mappings, which is used to locate terminals on the IP network. Both are based on tried solutions: TRIP is based on BGP-4 and ENUM uses the existing DNS system.

Number portability is generally implemented by modifying the mapping between name and address. In the PSTN the Intelligent Network implements the mapping functions. On the IP network the mappings of ENUM and TRIP can be modifying to realize number portability.

When the two network technologies, PSTN and IP, are interconnected new problems arise. The information in ENUM, TRIP and IN must be kept updated 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 the protocols is needed.

The protocols are still under development. The basic ENUM specification has reached RFC standards track stage but TRIP is still an Internet draft. Commercially available implementations are not available. The protocols' suitability for IP telephony in real networks is still not verified.

Yet, the need for standardized protocols for distributing IP telephony routes is high and the future for TRIP and ENUM seems promising. The signaling protocols alone cannot be used to form a global IP-based network, and the described protocols provide the required solution. However, as we have seen some new parts are required to make the architecture complete.

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