Introduction to VoIP

There's been a lot of discussion and a lot has been written about Voice over IP (VoIP). VoIP makes it possible to provide a telephone replacement service as additional functionality over a broadband internet connection. Due to the fact that VoIP does not require investment in costly telephone switches and because existing broadband connections are used, it becomes possible to offer the same services as those of traditional telephony infrastructure at a fraction of the cost. Furthermore, VoIP enables new services, such as video calls, presence and instant messaging.

This paper primarily addresses the pros and cons of providing SIP and ENUM-based telephony.

There are various VoIP standards, each with its own strengths and weaknesses. At the present time ENUM and SIP are the dominant de facto standards for VoIP. The reasons for this are as follows:

- SIP is future-proof: in addition to voice it can also be used as a protocol for video conferencing and instant messaging;
- SIP and ENUM are consistent with existing internet protocols;
- ENUM is the standard designed to effect the translation of a telephone number to an IP routable address;
- A wide range of SIP-compatible peripheral equipment is available;
- Major suppliers/vendors of telecom equipment, as well as open source communities supply different ENUM and SIP implementations.

What is SIP?

SIP is the acronym for Session Initiation Protocol and in a VoIP environment looks after the initiation and termination of calls. Figure 1 provides an overall overview of what is required for a VoIP service (for the sake of completeness, the traditional telephony protocol stack is also displayed for reference purposes).

The following elements are required:

1. Routing information: additions, deletions and changes of destinations. It is necessary to exchange destinations within and among operators. This generally takes place with the help of ENUM, which is where the link of the destination phone number to the SIP-URI (a type of e-mail address that identifies the SIP endpoint) is established.
2. Routing: the actual location of a destination when a session is to be initiated.
3. Signaling includes everything required to establish a session between endpoints. Is the destination online, is the session being established successfully, what type of session is to be established (e.g., voice or video), what are the capabilities of the destinations in terms of media?
4. Media flows: the call itself.

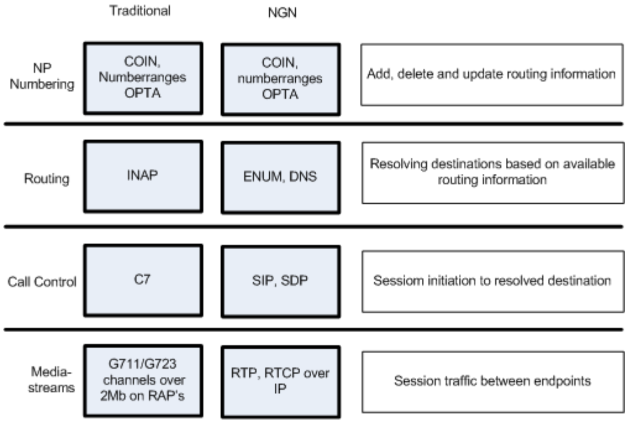


Figure 1: Overview of the protocols used in the Netherlands

SIP therefore is a component required for VoIP. SIP, in combination with SDP, ensures that calls are established within the VoIP system. SIP is defined by the Internet Engineering Taskforce (IETF) and is used by other standards as well, such as ETSI, IETF and 3GPP. SIP uses two types of mechanisms: User Agents (endpoints) and Proxies. A Proxy is a server that processes SIP signaling traffic and if necessary sends it to endpoints or other SIP Proxies.

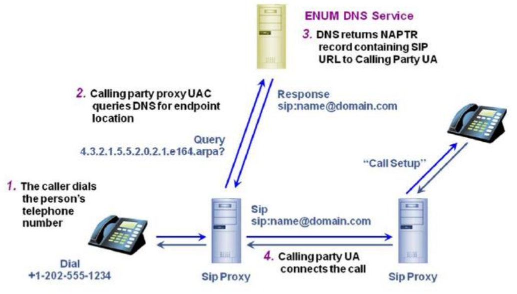
What is Carrier ENUM?

ENUM is the acronym for tElephone NUmber Mapping System. Originally developed for linking end-user telephone numbers to various IP services, the protocol turns out to be highly suitable for linking the interconnecting VoIP networks of carriers. It does the following:

- Translates telephone numbers to URIs (IP routable addresses)
- Enables the routing of IP-based services on the basis of telephone numbers
- Is based on DNS (Domain Name System)
- Provides a private implementation of ENUM by Network Operators with an Operator opt-in.

ENUM therefore is a component that can be used to determine the operator where a number is active in the NGN. For traditional telephony, operators have assigned this responsibility to the COIN *(Communications Infrastructure)* Association that

administers the number activation and number porting processes, and the centralized numbers database (CRDB) for E164 numbers in the Netherlands. An investigation is currently ongoing to make this information available to COIN Association members via ENUM (DNS technology). VDVL employees are participants in this initiative.



Source: Richard Shockey[18]

Figure 2: ENUM and SIP-based call setup

Setting up an SIP service

An SIP service for end-users can be set up with very limited resources. This enables end-users to call each other using a SIP phone or SIP software on a PC, mobile telephone or another device. If the SIP Proxy can be accessed on the public internet, then it is also possible to reach the users located behind another SIP Proxy. Someone with some Linux experience can create a working SIP Proxy with several users that can call each other in a single day (see Figure 3).

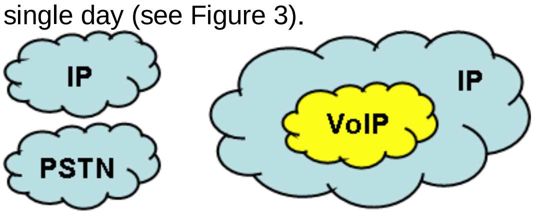


Figure 3: internal VoIP calling service

As soon as the VoIP service must also connect to a PSTN or IP network of another operator, its setup becomes far more complex, because in that case things such as interoperability, invoicing and protection against abuse become important (see Figure 4).

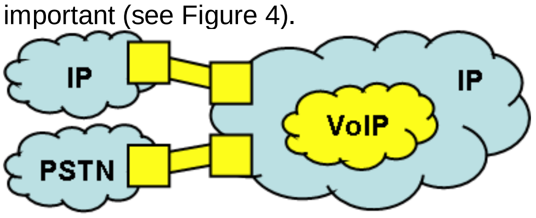


Figure 4: VoIP service connected to other operators

SIP and ENUM Challenges

Of course SIP does not only offer benefits. It has its drawbacks as well.

- SIP is developed on the basis of the internet architecture, whereby intelligence resides in the endpoints. Because VoIP services are often used to replace PSTN connections, the operator often does not want a connection to be accessible from the internet, because this can entail quality and/or security risks, and loss of revenue. The operator will have to take protective measures in this regard.
- Telephone numbers are used to interconnect PSTNs. SIP uses SIP URIs. Carrier ENUM enables the translation of telephone numbers to SIP-URIs and consequently provides a solution to potential interoperability problems. The establishment of Carrier ENUM requires collaboration among operators.
- Because the SIP standard lends itself to different interpretations, various SIP implementations are possible. For example, specific PSTN functions, such as DTMF tones for Voice Response Systems and Fax detection, or Calling Line Identification can operate differently for each party. The operator must take this into account. The interoperability between different SIP implementations consequently is not obvious. The standards require further work before **robust interoperability** will emerge like that developed in the traditional telephony sector over the last few decades. This area is being actively worked on in various forums, such as the IETF, GSMA and IPIA.

VDVL's experience with SIP and ENUM

VDVL has extensive experience in the following areas that play a role in SIP and ENUM implementations:

- Consumer and business SIP implementations
- Peering using ENUM and SIP-based platforms
- Number portability and ENUM
- Connections to network management and BSS/OSS systems

Global Telecoms Business Innovation Award

VDVL received a Global Telecoms Business Innovation Award in October 2008 for the development of a Proof-of-Concept for the peering platform of the members of the Joint Cable Campaign (JCC), such as UPC, Ziggo, ZeelandNet and CAIW.

Additional Information

Additional information concerning our references and knowledge areas is available on our website www.vdvl.nl. For additional information contact:

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