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The Future of VoiP interconnection

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1 Introduction

The Dutch PTT used to have a commercial with the slogan, "If you can count to ten, you can call the world." This slogan captured how users saw telephony then and today. They enter the digits on their phone and can reach the world. But the telephony provider has to interconnect its network with other networks in order to deliver on the promise that the whole world can be reached.

Interconnection is what a telecommunication provider has to do to enable communications from one network to other networks and devices. Interconnection is necessary, either because the two networks are based on two different technologies (fixed to mobile or mobile to VoIP, for example) or because they are owned by two different entities. Interconnection therefore consists of two elements:

1 Technical: all the hardware, software and arrangements necessary to guarantee the uninterrupted flow of communications from one network to the other; and

2 Business: the commercial basis upon which the parties agree to connect their networks and exchange communications.

3 A third element needs to be added, as the telecommunication industry is heavily regulated:

4 Regulatory: The regulations that govern the conditions regarding the technical and business elements of interconnection.

In recent years one technology, Voice over IP (VoIP), has left regulators and businesses wondering what the future of communications will hold. Mobile telephony has swept the world, with more than 4.6 billion users expected by the end of 2009[1] and over a USD 1,000 billion (USD 1 trillion) in revenue,[2] but it has left the structure of the industry intact. VoIP can boast impressive growth, both in users as in revenues, but not as impressive as mobile telephony. Still, its impact is feared. Voice over IP does away with some of the truths that were held to be self-evident in the sector. The most striking change is probably the separation between providing voice as a service and providing the physical network infrastructure. Another possibility enabled by VoIP technology is that consumers can have voice telephony without actually getting it from a telephony service provider. What VoIP promises, therefore, is a reduced role for the traditional telephone companies in the provision of voice, and a declining role for traditional telephony in the revenue stream of the telecommunication industry.

The dramatic change that VoIP can bring with it also has its effects on interconnection. This chapter intends to illustrate the following:

1 VoIP by itself doesn't bring dramatic changes to the process of interconnection and its technical implementation. Many of the changes that VoIP is expected to bring might already be possible in circuit-switched networks.

2 VoIP interconnection can support the same business and regulatory practices that exist in the current market. Combined with changes to business models and regulatory regimes, VoIP can, however, greatly decrease the cost of voice service and increase its use and usefulness.

3 Interconnection in itself is easy to understand, but is made more difficult by business regimes and regulatory arrangements.

This chapter has been written with a view of interconnection in liberalized telecommunication markets, with multiple telephony service providers operating in a competitive market. There still are telecommunication markets where unlicensed use of VoIP is prohibited by law. The introduction of VoIP and VoIP interconnection for licensed telephony service providers doesn't necessarily have to involve changing laws that apply to unlicensed VoIP. Many of the points in this discussion paper still hold, whether unlicensed VoIP is allowed or prohibited. There also doesn't need to be a difference between developed and developing nations. It is even possible that interconnection based on IP technologies will be easier to implement in developing nations, where mobile service is more prevalent than traditional wireline technologies.[3]

It is important in this paper to make the distinction between:

- Telephony service providers: These are parties that provide a voice service either over IP-based networks, satellite systems, mobile or traditional copper based networks (or any other network); and

- Telecommunication network providers. These provide physical telecommunication networks and are in the business of carrying traffic, either in the form of circuit-switched telephony or as voice or data packets.

The two are fundamentally different, although telephony service can be provided by a telecommunication network provider. The traditional telephony companies that dominated telephony for a hundred years did just that. In fact, they weren't capable of separating the service from the network. But in recent years, such separation has become the norm in many countries around the world.

2 The Development of interconnection

Interconnection of telegraphy was the basis for the establishment of the ITU in 1865. Interconnection now is as much a necessary process as it was then. The actual interconnection of two networks is now mostly governed by business rules, although almost all governments are involved through rules and regulations. In some ways, the process of interconnecting telephony networks is essentially much the same as it was at the start of the ITU:

1. Looking up the network (segment) on which the recipient of the communication is located;

2. If necessary, applying regulations (i.e., for emergency services calls);

3. Signalling the recipient (which in itself may require a look-up to identify where the recipient is located on the network, or on what network the recipient is roaming);

4. Transferring the communication;

5. If necessary, transcoding to a different protocol; and

6. If necessary, applying billing rules.

Where telegraphy worked with names and addresses, telephony used telephone extension numbers, and the manual communication and interconnection process quickly became an automated one.

Box7.1: How Interconnection Came To Be: The Technology, the Business Case and the (Inevitable) Regulation

On 24 May 1844, Samuel Morse sent his first public message over a telegraph line between the U.S. cities of Washington, D.C. and Baltimore, and through that simple act, ushered in the telecommunication age.

Technology
Barely 10 years later, telegraphy was available as a service to the general public. In those days, however, telegraph lines did not cross national borders. Because each country used a different system, messages had to be transcribed, translated and handed over at frontiers, then re-transmitted over the telegraph network of the neighbouring country.

Business
Given the slow and unwieldy nature of this system, many countries eventually decided to establish arrangements which would facilitate interconnection of their national networks. However, because such arrangements were managed by each country at a national level, setting up telegraph links often required a huge number of separate agreements. In the case of Prussia, for example, no less than fifteen agreements were required for the link between the capital and the frontier localities bordering other German States. To simplify matters, countries began to develop bilateral or regional agreements, so that by 1864 there were several regional conventions in place.

Regulation
The continuing rapid expansion of telegraph networks in a growing number of countries finally prompted 20 European States to meet to develop a framework agreement covering international interconnection. At the same time, the group decided on common rules to standardize equipment and facilitate international interconnection, adopted uniform operating instructions that would apply to all countries, and laid down common international tariff and accounting rules.

ITU
On 17 May 1865, after two and a half months of arduous negotiation, the first International Telegraph Convention was signed in Paris by the 20 founding members, and the International Telegraph Union (ITU) was established to facilitate subsequent amendments to this initial agreement. Today, some 140 years later, the reasons which led to the establishment of ITU still apply, and the fundamental objectives of the organization remain basically unchanged.

Source : ITU (www.itu.int), headings by the author.

2.1 From the 19th Century to the 20th Century

Three things have impacted the process of interconnection through the years, since the early days of Alexander Graham Bell and, later, the state-owned PTT's.

1. Numbers became personal (names): The phone number, standardized in ITU-T Recommendation E.164, used to function as a code to identify a physical line on which a connection was made. Entering the number would program the automatic switches (mechanical or electrical) to connect to the right line. If you moved to another house, you had to get a new phone number. These days, with modern intelligent networks, the number is not the physical identification of the line, but rather is associated with a different number that represents a physical element in the network. The mapping between the two is done in a switching table in the switch, which will look up the E.164 number in a table and

associate it with a number of a physical location or the number identifying a mobile phone.[4] In theory, any number can be used on any kind of network, mobile, IP or PSTN.[5] This has allowed the users of numbers greater flexibility to retain their numbers and take them from one operator to another. As a result, numbers have become personal.

2. Separation of Signalling and Media: Signalling used to be limited to the start of the conversation, ringing the phone. Through the years signalling was extended and separated from the conversation (or media) channel. The media and signalling could take different routes on the network. This freed up media channels and allowed for much more data to be exchanged for signalling, before, during and after the call. It made new services possible like call forwarding, call waiting, called number display, etc. The signalling protocol used today on the PSTN and mobile networks is known as Signalling System No. 7 (SS7) (specified in the ITU-T Q-series recommendations).

3. Nomadic/Mobile Use: Users and their phone numbers became mobile. Intelligent networks allow for different end-points, depending on the time of day, the geographical location of the caller or the called person. For interconnection this has had dramatic effects. A number isn't associated with a fixed geographic region anymore, it can be anywhere in the world. As an illustration, in mobile networks the Home Location Register (or equivalent) of the operator providing mobile telephony to the end-user will signal the location of the end-user and allow for the connection to be set up to that point. It is not necessary for the media stream to be routed via this operator. An American mobile phone user can be called on a mobile number with a geographic number of New York, while roaming in South Africa. (Unlike many countries, the United States doesn't use specific numbering ranges for mobile telephony. It uses geographic numbers for fixed as well as mobile telephony.) If called from a South African fixed line, the signalling will pass through the United States to validate and identify the user, but the actual conversation link will take a more direct route in South Africa and not pass through the United States.

2.2 VoIP Changes Basic Premises

Three premises seem to underlie the thinking around traditional voice interconnection:

1 The major premise that underlies interconnection today is that a telecom network operator is necessary to find a called party and to set up the call.

2 A second premise is that the originating telephony service provider needs to have a contract with the network the called party is on or with a transit provider in order to establish a telephony connection.

3 The third premise is that both parties need to use E.164 numbers in order to call each other.

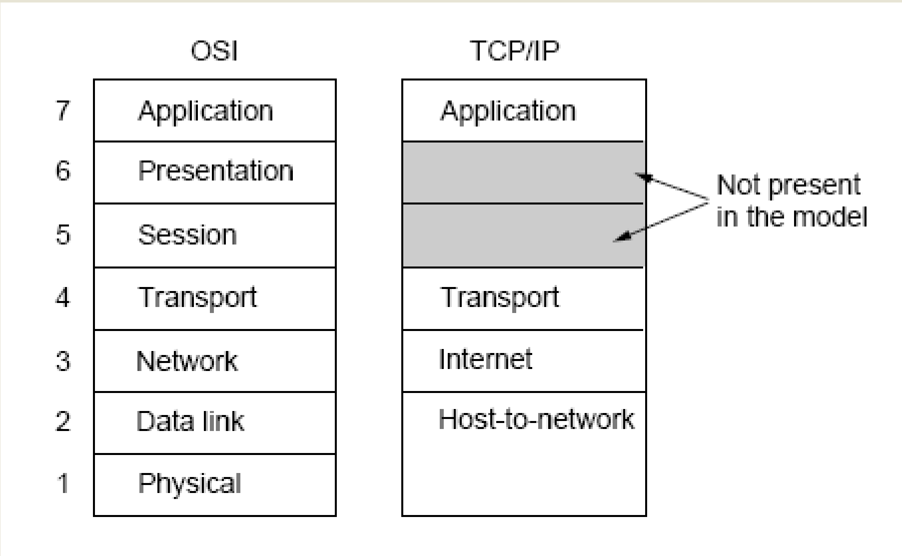
VoIP has shown that these premises don't necessarily need to be true. Voice is an application on IP networks. The application layer (layer 7 of the OSI-model[6]) doesn't need any interaction with the physical layers (layers 1 and 2) that the operator of the network controls. The IP layer (layer 3) abstracts the physical layer away and allows the application to function in the same way regardless of the physical network it is on.

IP allows two users to set up a call without the need for a telecommunication company to assist in signalling or routing of the media. All users need to know is the IP address of the other. Finding out the IP address of another user can be done through a website, an instant messaging application or peer-to-peer applications. Peer-to-peer applications (like Skype) don't even have a central authority to record what IP address a user is on. Instead, they make use of cryptographic certificates to uniquely identify end-users. They make the assumption that within six degrees of separation there always is a party that can find the called party. By contacting well-known nodes or nodes that have previously been contacted the entire network can be reached.

Because the first premise isn't true, the second one isn't true either. A user can subscribe to a VoIP provider and, as soon there is a functioning IP connection, have all calls routed over the internet. The VoIP provider doesn't need to have a contract in place with the network the user is on. It is the user that needs to have the connection. He or she needs a globally reachable IP address, but this can be on a corporate, academic or public telecommunication network.

The third premise then falls away, too. VoIP doesn't use a standard identifier for a call. Where traditional (mobile) telephony uses an E.164 number, VoIP can use any string as an identifier, as long as it is unique in the context in which it is used. This can be an email address, an E.164 Number, a digital certificate, or even a chat or game identifier. It can be part of a proprietary addressing scheme that is organized on a global scale and persistent over time, or it can be *ad hoc* -- lasting only for the duration of the call. This doesn't mean E.164 numbers have lost their significance -- the use of telephone numbers is much too embedded in our society for this to happen. It just means that it is possible to set up calls without them.

Figure 7.1: The TCP/IP reference model



Source: Andrew S. Tanenbaum, Computer networks (4th edition)

Because VoIP is an application, it can be integrated into a host of different applications. For example, video game consoles contain computers with integrated VoIP capabilities, which can be used to establish voice links among players. Most instant messaging applications also support VoIP, and now host millions of voice and video conversations per month. Skype is now named as the largest provider of international voice calls, hosting 8 per cent of all international calls, or 33 billion minutes of PC-to-PC communications and an unknown amount of PC-to-PSTN calls.[7] Skype's revenue (per minute and overall) is not as substantial as other market players, but this should be more of a warning to other market players of how much cheaper calls can be. It is not an indication that Skype is insignificant to the market.

For this paper however, the focus will be on telephony enabled by E.164 and leave the exotic versions of VoIP like those possible with using Instant Messaging, computer games and websites as they are often less of a regulatory concern.

3 The Technology behind Interconnection

There are three steps in routing traffic to an interconnected operator: look-up, signalling and transferring/transcoding. First, the destination number has to be looked up in a table in order to identify the receiving network. Then, the receiving network has to be signalled, after which the receiving party will be signalled. If the receiving party responds by accepting the call, this will be signalled back and the call will be transferred and transcoded. In this section, we will look at the technologies that support these three steps and how they might change with the move to VoIP. A note of caution: although there is considerable growth in VoIP, this doesn't mean that circuit-switched telephony is going away any time soon. The future may very well see a mixed assortment of circuit-switched and packet-switched networks that need interconnection.

3.1 Look-Up

Before telephony networks were digitized and telephony numbers became names, switches didn't perform actual look-up processes, they just mechanically followed the code they were given. With digital networks, it became necessary to do a look-up in a table to know where the called number resided in the network. In the PSTN network this look-up is performed through the SS7 protocol.[8] All telephone numbers are associated with a telephone switch that has its own identifying code in the network. Based on the number called, the telephone switch will either "know" which switch to forward the call to, or it will perform a look-up based on a database. The basis for this is a numbering plan, which details the geographic locations and uses of the numbers.

The complexity of look-ups has increased over the years.

- It is now very possible to route calls dialled to one telephone number (i.e., a toll-free number) to different actual end points thousands of kilometres apart, based on time of day, location of caller, busyness at call centre, etc.
- In wireless networks, the complexity is increased because users may be on the move or roaming, in their home countries or abroad.
- Liberalization has allowed multiple providers of telecommunication networks and telephony services to enter the market.
- Number portability has added another layer of complexity, as the user can change telephony service providers while retaining the same number.[9]

The look-up mechanism in the current PSTN is based upon the SS7 specification. There is no one centralized database that the operator can use to pinpoint the end point the dialled number is at, but the routing is incorporated in the switching tables of the switches. The switches will not only determine where the end point is located, but also what services that end point can use. This determination can only be performed by the switch, and is based on technical capabilities of the network and the services the end user has subscribed to. A new service, like SMS on fixed networks, or online chat, can only be introduced if the network and its look-up mechanisms support it.

On the Internet, look-ups are performed to translate domain names into IP addresses. For example, "www.itu.int" is translated into the IP numbers "156.106.202.5" by the Domain Name System (DNS). The use of alphanumeric names allows for an incredible diversity in domain names that are easy to remember for end users. It relieves them from having to remember a long string of digits, and it makes it possible to host multiple sites on one IP address. The DNS works as a multi-tiered hierarchy. The top of the hierarchy (or as it is formally known, the root) is managed by ICANN and served through a redundant infrastructure of 13 root servers. ICANN's DNS servers point to the location of country-code-level domains like ".nl" or ".lb" and generic top-level domains like ".org" and ".com." From this, the DNS name server of the registry for the top-level domain points to the DNS name server that is associated with the domain name. For the ITU the name server is "ns.itu.ch" at "156.106.192.121" and two others. These then can point to specific sub-domains like "mail.itu.int" or "voip.itu.int." Using a name and the "@" sign with the sub-domain creates a uniform resource identifier (URI).

The main difference between a look-up in the traditional PSTN/SS7 world and the DNS is that the DNS is an open system. Only a telephony operator's switch can perform a look-up on the PSTN. Similarly, only a telephone company can change or add data to the record of a particular number. By contrast, the DNS is maintained by the owner of the domain -- the person who has registered the domain name. Neither the telecommunication network that provides connectivity nor the registrar who has registered the domain name can influence what services and functions are operated by the domain or what sub-domains exist.

3.2 E.164 Number Mapping (ENUM)[10]

In order to use telephone numbers for VoIP and other services, it was necessary to map a telephone number to a Uniform Resource Identifier (URI), which could then be converted by the DNS to an IP address. ENUM allows the conversion of telephone numbers to domain names by reversing the telephone number +310123456789 to 0.8.7.6.5.4.3.2.1.0.1.3.e164.arpa. A look-up in the root of the DNS, managed by the Internet Assigned Numbers Authority (IANA), will then point to Dutch ENUM registry's DNS server, which holds the registry for +31. The DNS server will then be able to reply with the correct URI, depending on the service requested. From here the connection will be set up to establish a VoIP or instant messaging session or to deliver an e-mail. This way the telephone number can be used as a universal identifier to deliver any kind of service.

The initial version of ENUM, as standardized in RFC 3761,[11] is also known as "Public ENUM" or "User ENUM." What characterizes Public ENUM is not the look-up mechanism, but that it is a publicly accessible directory that gives the end user full control of the registration of the telephone number. In order to establish a registry, the ITU, the Internet Architecture Board and the corresponding ITU member state work together to register the country code in the global ENUM registry, which is maintained by the RIPE Network Coordination Centre (RIPE-NCC) in Amsterdam. The end user can register a phone number in the Public ENUM registry of the country of residence, and then can determine how he or she wants to be reached. This cuts out the telephony operator from the process. It changes the relationship between the subscriber and the telephone network into one resembling the DNS, where the owner of the domain name (or telephone number) determines what services are available and how they are routed.

There are currently nine ENUM registries active for nine countries.[12] Public ENUM hasn't been the success some thought it would be. Some of the reasons cited are:

- A lack of available services and a lack of commercial interest, because of a lack of customers. This creates a chicken-and-egg situation, where investment in commercial applications waits for the customers and the customers wait for commercial applications.

- Incumbent telephone service providers' mistrust based on privacy and security reasons. Telephony service providers don't want to communicate their interconnection points to the public internet for security reasons.

- End user mistakes in entering end-points and updating those end-points. These mistakes may seriously impact the perception of ENUM, because from a user perspective, telephony just has to work.

- Commercial concerns about losing customers. Public ENUM might empower subscribers to be independent of the telephony provider, as well as allowing the telephony provider's competitors to bypass it and deliver services directly to the customer, leading to a loss of revenue.

The viability of public ENUM, in which the consumer provides all the information and is solely and fully responsible for its accuracy, can be questioned. But the vision of users being empowered to control their communications is a powerful goal for regulators to consider.

Even though Public ENUM hasn't been a major success yet, a look-up mechanism based on ENUM is seen the end result that the technology industry will move to in coming years. It has been adopted by international standardization bodies such as the 3rd Generation Partnership Project (3GPP) to enable future networks and services, and it is in use on several commercial interconnection platforms. Its attractiveness lies in the reliability, flexibility and scalability cost- effectiveness of the DNS software used for ENUM.[13]

There are different forms of ENUM that might potentially be used. They are known as *infrastructure*, *carrier* and *private* ENUM. All of them all take away the public side of ENUM, but keep the look-up part:

- **Infrastructure ENUM** is the term used when telephony providers within a given geographic region use a central ENUM facility to find the telephony service provider that serves the subscriber. This may be used for VoIP-to-VoIP calls, but also for VoIP calls to and from fixed or mobile telephone networks. It can even be used for standard telephone calls without reference to VoIP. The idea is that this will also be in a standardized, top-level domain, with approval of national telecommunications authorities.

- **Carrier ENUM** is the term for when a group of telephony providers decide to use ENUM as a look-up mechanism for the numbers they are responsible for. It differs from Infrastructure ENUM in that Carrier ENUM does not have to be sanctioned by a national government. Participation by operators is voluntary, and membership can be global. For calls outside the group, the telephony providers resort to other Carrier ENUM structures, Infrastructure ENUM, or to SS7-based look-ups

- **Private ENUM** is when a telephony provider uses ENUM in its own network as the look-up mechanism. For calls outside the carrier's network, it will resorts to either another Carrier ENUM system, Infrastructure ENUM or SS7.

These three ENUM options can also be used together, so a telephony provider may use Private ENUM in its own network, Carrier ENUM for its global alliances and Infrastructure ENUM for national calls. The main difference, therefore, between ENUM and SS7-based lookups is that SS7 is rigid and limited in its implementation. ENUM can be used functionally the same way as SS7, but it can just as easily be used in a more flexible and open way to allow new services to be added quickly.

The vision of Public ENUM could partially be realized by giving the user control over some elements, for instance by allowing the user to enter an e-mail address next to a telephone number. The telephone number and its routing would then still be under control of the telephony provider

3.3 Signalling

The ITU-T defines signalling as "the exchange of information (other than by speech) specifically concerned with the establishment, release and other control of calls, and network management, in automatic telecommunications operation."[14] On telephony networks, the dominant protocol for interconnection and signalling is Signalling System No. 7, known more commonly in North America as “SS7” and elsewhere as “C7.” SS7 is both a network architecture and a series of protocols that provide telecommunication signalling. The signalling network can be considered the telecommunication network's nervous system.

At this moment all traditional telephony networks, whether they primarily provide VoIP, mobile cellular or PSTN services, use SS7 as their main signalling protocol for interconnection with other networks. SS7 is often used even between VoIP networks. Within their networks, however, VoIP service providers use different protocols, which are increasingly being adopted for interconnection between VoIP networks.[15] The dominant signalling protocol within a VoIP provider’s network is the Session Initiation Protocol (SIP), although H.323 is used frequently, as well.[16] Specified by the Internet Engineering Task Force (IETF), SIP is a protocol that can be used to set up any kind of session between two or more end-points. It is useful not only for telephony, but also for video, games, chat etc. It has also been accepted by the 3GPP as a signalling protocol and permanent element of the IP Multimedia Subsystem (IMS) architecture for IP-based streaming multimedia in mobile cellular systems. SIP has been designed so that it can execute all familiar SS7 functionalities. Functionally it is therefore equivalent to SS7.

One of the main differences between SS7 and SIP is that SS7 handles communication between switches. SIP can operate between clients and/or switches and can be completely peer-to-peer, requiring no interaction with a centralized telephony service provider. This makes it more flexible to introduce new features to a communication as there is no requirement for the network to support the service, only the clients need to be able to. The flexibility makes it more scalable and more “future-proof” than SS7, in the context of the transition to IP networks. This doesn’t mean that a telephony provider using SIP has to allow end users control or allow peer-to-peer SIP communication. It can work in that respect just like SS7, but this is optional and not required.

SIP and ENUM integrate very well, so much so that they are often named together as SIP/ENUM. They both support the same kind of openness and flexibility. There are different flavours of SIP available as a result of different vendor implementations or further standardization in different standardization bodies (like SIP-I standardized by the 3GPP).

Some experts believe that in the next five to 10 years, the current signalling systems based on national and international versions of SS7 will be replaced by a combination of ENUM and SIP-based signalling, look-up and interconnection. Among the main reasons cited for this move is the cost of SS7 equipment compared with a combination of SIP and ENUM.[17]

3.4 Transferring and Transcoding

After looking up a subscriber and signalling that there is a call, the transmission has to be transferred to the destination/receiving network. This requires a physical link between the two networks, and the call must be transcoded from the protocol used on the initiating network to the one used by the network on which the call is terminated. In both the physical interconnection and in the transcoding of protocols, we see changes that enable the move to VoIP interconnection.

For the physical interconnection, operators can choose to interconnect their networks either directly or through an intermediary network. Most current interconnections between major networks are over dedicated lines that are part of the telecommunication network. The move to all-IP networks makes it possible to mix voice traffic with other types of data traffic. This allows for more flexibility and scalability. Telephony providers may still choose, however, to use dedicated lines to allow for better manageability of the traffic streams.

One of the options that VoIP offers for transferring and transcoding is to avoid any carrier involvement. The carrier only needs to perform the look-up and signalling, then clients can route the traffic directly between the two points. This is essentially how the Internet currently works for most applications, but it is quite different from how the traditional telephony network operates. Routing over the "public Internet" is often regarded as less secure, and lower in terms of Quality of Service, than transferring and transcoding over a telephony network.[18] Whether or not this holds true depends on many variables. With advances in encryption, security, bandwidth and more widely interconnected networks, these objections may go away over time.

VoIP services make use of the same encoding as PSTN and mobile networks.[19] Transcoding of the communication may, therefore, not be necessary. Where VoIP is different from the PSTN and mobile networks, is its flexibility in the use of different *codecs*. The transcoding on PSTN and mobile network is limited to the codecs that those networks support/recognize. A practical example is *High Definition Voice*, which is now being introduced on some mobile networks. High Definition Voice requires active involvement of the telecommunications network. Two VoIP clients can negotiate any codec that both clients know and adapt it to the conditions that the communication is used in. The quality of the conversation can therefore vary from usable to CD-quality. With some codecs it is even possible to use a variable bit rate to allow the quality to be adapted during the conversation, based on throughput that the user is receiving from his connection.

4 The Business Side of Interconnection

As explained in the previous section, the move to VoIP and SIP/ENUM for interconnection means the move to a cheaper and more flexible technological platform. This platform can support all the requirements of existing business models, but it can also promote change to a new world. This is not because of technological changes *per se*; the flexibility and cost savings that the technology offers will lower barriers of entry and create new services and opportunities. The move to VoIP and VoIP interconnection promotes a fundamental rethink of the telephone business.

The most fundamental question the industry has to ask itself is whether it is useful to charge per-minute rates for use of a telecommunication network. For almost a century this has been common practice in almost all countries, so the general opinion is that it is the right way to charge for telephony. The majority (up to 80 per cent) of the cost of a telecommunications network (fixed or mobile) is in capital expenditure for lines, switches, rights of ways, licenses, and other expenses. These are fixed, upfront costs – not affected by the amount of use. If usage is not impacting much of the cost, than why are charges usage-based? Another way to cover the network costs would be to get as many potential users connected and spread the costs accordingly – without charging per-minute fees that limit usage. More and more telephony providers are moving to flat-rated, "unlimited" price structures. The next step could be to do away with charging and measuring for telephony at all, much like the current system for e-mail. Whether or not this happens will be a result of competition and of regulatory choices, which will be discussed later on.

This question has become all the more relevant now that the world is using more data than voice and a 1 megabit ADSL broadband connection can carry the equivalent of 12 standard telephone lines.[20] T-mobile Netherlands has reported that its iPhone users were downloading 640 MB of data per month, which is equal to 1,000 minutes of VoIP at 80 kilobits per second (Kbits/s).[21] Compared with an average telephony usage of between 200-400 minutes per month at 12.2 Kbit/s, it is clear that data uses much more resources than voice. But the cost of voice minutes on many Western European networks are significantly higher than that of data measured on a per bit basis (for example T-Mobile NL charges EUR 10 for an unlimited data subscription and EUR 30 for 250 minutes of voice calling).[22] An increasing number of telecommunication carriers (fixed-line, ISPs, etc.) have started offering unlimited calls on their networks, and some even provide unlimited calls to other networks.

The debate over flat-rated versus usage-based pricing is also taking place on the business side of voice interconnection. When two networks interconnect, they enter into a commercial agreement, arranging the "how" and the "what price" for interconnection. The "how" of interconnection covers:

- Location(s) for the interconnection;
- Technology used;
- Delivery method of the media (direct interconnection over a dedicated line, over an intermediate platform, or point-to-point over the Internet);
- Back-ups; and
- Service level agreements for repairs, upgrades etc.

Meanwhile, the "what price" agreement addresses the division of costs between the two interconnecting parties. The division of costs is handled completely differently in the Internet world than in the telephony world – a fact that posing one of the biggest challenges to traditional interconnection.

4.1 The « How » of Interconnection

The quality of the interconnection between networks determines, in part, the quality of services that can be delivered to customers. The more interconnected a network is, the more likely it will have shorter routes, better back-up routes and reduced outages. In an ideal world, each network would want a direct interconnection with all other networks for look-ups and interconnection of media. The introduction of VoIP interconnection simplifies the global interconnection of VoIP networks to such extent that it becomes a practical reality for almost all networks, regardless of their size. In the old PSTN-based world, global interconnections were much more cumbersome and therefore not as popular.

In reality, of course, it is not possible to connect directly with all networks. For international calls, national fixed telephony providers often turn to international transit providers, which connect their networks to the rest of the world. In liberalized markets, the same thing happens with national calls. The use of transit providers does add costs to the interconnection, so market players try to use transit as little as possible and try to get as many direct interconnects as they can. The international transit market has become very competitive, with the margins for the transit carrier squeezed by ability of other operators to shop around for the lowest transit rates.

The introduction of VoIP and interconnection based on SIP/ENUM will most likely increase the number of market players, due to significant reductions in market-entry barriers. The effect of the lower interconnection costs, compared with current SS7 interconnection costs, is that there will be more direct interconnections, bypassing transit parties. It will also make international interconnection between networks easier, further increasing the amount of possible interconnects. But having more market players also increases the complexity of directly interconnecting with all other market players. To simplify interconnection, telecommunication carriers can decide to establish an interconnection platform to which they all connect. The platform can be used either for look-ups, for the exchange of media, or both. Some examples of interconnection platforms are:

- Number portability platforms – Many countries have a number portability platform to simplify the announcement and look-up of telephone numbers that have been "ported" (that is, transferred) away from the original network. Participation is often mandatory for voice services providers in the country.
- Internet Exchange Points (IXPs) -- For the physical interconnection of IP-based networks there are other platforms: Internet exchange points. VoIP providers may interconnect over IXPs, but the IXP is indifferent to the type of traffic exchanged and doesn't play a role in the VoIP interconnection.
- SIP/ENUM platform -- Dedicated platforms that allow the interconnection of SIP/ENUM based networks are started either by cooperation among operators, acting in what's often known as a federation, or by commercial third-parties, whose business model is to simplify SIP/ENUM on a global scale.

Examples of the federation approach include:

- The GSMA's Carrier ENUM-platform known as Pathfinder, which provides routing information to members in a closed group.
- The GSMA's IPX, which is an IP-traffic exchange that is separate from the rest of the Internet, allowing the exchange of VoIP media and signalling packets between telephony service providers. It is promoted with openness, quality, cascading payments and efficient connectivity as its unique selling points.
- The Joint Cable Consortium SIP Exchange (JSX) initiative. It started out as an initiative between the largest cable companies in the Netherlands, but is now open to other telephony service providers. It uses the Amsterdam Internet Exchange as the platform for media exchange between providers.
- The Korean Internet Exchange (KINX) VoIP peering initiative is an initiative where 15 Korean providers of IP-telephony joined together to interconnect their VoIP-services over the KINX.

Examples of the commercial third-party approach, meanwhile, are:

- Xconnect, which provides a global platform based on SIP/ENUM, allowing VoIP providers to interconnect and find interconnection partners. It allows the creation of multiple federations with different agreements and payment structures between the telephony service providers. Its technology powers the KINX and JSX initiatives mentioned above.

- Neustar, which started as the operator of the North American number portability platform, provides a global SIP/ENUM exchange called SIP-IX. It is also the operator of the GSMA's Pathfinder service.

- The Voice Peering Fabric is another operator of a VoIP interconnection platform. Its unique selling point is that next to facilitating interconnection, it also facilitates the trade of roughly 1.5 billion voice minutes per day.[23]

The charging principles on the commercial third-party platforms can be anything the telephony service providers agree on and what is supported by the platform. They are open to all interested parties regardless of where they come from and what their status is in their home country (incumbent or new entrant).

4.2 The « How Much » of Interconnection

A practical view of interconnection would give the wrong idea that the costs of interconnection aren't a major issue -- certainly not in a VoIP-interconnected world. All a VoIP provider needs to do to enable interconnection to another VoIP provider is to point to the other provider in its look-up tables. The data packets can then be transmitted over an existing link or an Internet exchange point. You don't even need the VoIP providers if two parties both have a functioning IP-connection (as in the case of Skype). The cost, if any, wouldn't be much different from an ISP's cost to interconnect with another ISP over an Internet Exchange Point.

For the interconnection of IP networks, the situation is much as described above. Both interconnecting networks are responsible for their side of the interconnection. If one or more transit parties are necessary between the two end-point networks, the two networks will each pay their own transit costs. This arrangement is known as b*ill-and-keep*. There are two types of contracts for IP interconnection: Peering contracts and transit contracts.[24]

- **Peering** is when two or more autonomous networks interconnect directly with each other to exchange traffic. This is often done without charging for the interconnection or the traffic.

- **Transit** is when one autonomous network agrees to carry the traffic that flows between another autonomous network and all other networks. Since no network connects directly to all other networks, a network that provides transit will deliver some of the traffic indirectly via one or more other transit networks. A transit provider's routers will announce to other networks that they can carry traffic to the network that has bought transit. The transit provider receives a "transit fee" for the service.

With a per-port cost for 10 Gbit/s ≈ 150,000 simultaneous calls at around USD 2,500 month at major Internet Exchange Points and the total operating expense of about USD 10,000, the costs of interconnection can be as low as USD 0.06 per line, per month.[25] The wholesale costs of transit can be as low as USD 4 per mbit/s and per month, which is equivalent to USD 4 for 150,000 to 250,000 minutes of VoIP calls.[26]

Indeed, calculating the cost of traditional fixed-line and mobile telephony interconnection is much more difficult than what was just described. In most countries in the world, the receiving party's telephony service provider can apply a termination charge to the calling party "for the resources used." This is the charging mechanism known as *calling-party pays*.[27] The costs of such termination can range from USD 0.004 per minute in India for all national (fixed-mobile – fixed-fixed) calls to around USD 0.50 per minute or more for some international calls. It is clear that the attraction of calling via the Internet is also driven by a financial element. The differences between the interconnection charging mechanisms is worth a separate discussion, which is beyond the scope of this paper. However, VoIP interconnection can have implications for the way telephony service providers, regulators and consumers look at the costs of interconnection. If the costs of IP interconnection are so low, then why would the price of telephony interconnection be between 150 and 15,000 times more expensive? Some commentators argue that the move to VoIP-interconnection will automatically imply a move from calling-party pays to bill-and-keep. This isn't necessarily the case, as VoIP interconnection can support the same functionality and therefore the same business models as traditional interconnection mechanisms.

What has become clear from interviews, as well as from analysis of the current interconnection issues, is that when companies move to VoIP peering, they are more inclined to move away from calling-party pays models, because it simplifies inter-carrier billing. Many telephony service providers see roughly symmetrical traffic flows to and from each other. When the termination charges are also symmetrical, the net effect of termination charges on profit and loss is zero, but there is also a downward effect on gross revenues, which makes some providers weary to move in this direction. There are, however, telephony service providers taking this step, as it allows them to simplify their interconnection processes.

The use of VoIP enables consumers and telephony service providers to bypass the termination charges or costs imposed by telecommunication network operators. Those operators cannot commonly distinguish VoIP from other traffic that is transmitted over a broadband network. It is this situation that has made VoIP feared by many incumbent network providers and is the reason it is still banned in many developing countries. Even on traditional telephony networks, it was possible to bypass a cost that a network would normally levy, by posing as different cheaper traffic – for example, local instead of national, or on-net instead of off-net traffic. Telephony providers still go to great lengths to bypass high interconnection charges for international calls, either by routing through a third country or by using call back systems.

Where VoIP interconnection differs is that it allows interconnection to or from a VoIP client over a fixed or mobile broadband connection, where the traffic is treated as any other packetized data, be it an e-mail attachment or YouTube movie. For international calls, which are often the most expensive, VoIP clients like Skype are used to bypass high international telephony charges. The effect of VoIP on traditional telephony providers could be a dramatic drop in income generated from interconnection. This income has supported the business case of the incumbents over decades, but when voice becomes just a service over the network instead of its *raison d'être*, minutes may not be the correct pricing unit any longer.

5 The Role of the Regulator

Regulators have been heavily involved in PSTN and mobile interconnection. As a result, any change to interconnection now, whether technological or in business models, is likely to result in regulatory changes. Regulators may have tried to be technologically neutral, but they have often codified existing technologies or business practices, either in precedent-setting decisions or *ex ante* rules. As one interviewee said, "You can't mention the word interconnection in a telecommunications company, without a lawyer standing up and saying, ‘That’s my responsibility.’” So, changes in the model also imply changes in regulation.

The role of the regulator is to guarantee the competitiveness of the telecommunication market and to ensure the consumer’s trust.[28] With regard to the future of VoIP interconnection, that role is to see:

- If VoIP interconnection will stimulate competition; and
- Whether the consumer’s trust in the telecommunication market can be upheld by VoIP interconnection. As explained in the first part of this discussion paper, for the end users it is irrelevant what technology is used, as long as they can dial any number and have a conversation.

When the answer to these two questions is yes, regulators will have to ask themselves:

- Are there any regulatory barriers to the introduction of VoIP interconnection that will need to be removed, or can the government enable the introduction of VoIP interconnection in other ways?
- Are there any anti-competitive barriers restricting the move to VoIP interconnection?
- Does the government have an active role in promoting VoIP interconnection, or should the move to VoIP interconnection be left to the market?
- How can consumers’ trust in VoIP interconnection be guaranteed?

The processes behind interconnection don’t change fundamentally, and the technology used for VoIP interconnection can support the current arrangements. The technology behind VoIP interconnection seems to be cheaper, more scalable and flexible, so more entrants can be expected to enter the market, which, in turn, will lead to more use, more uses (types of applications) and more usefulness of interconnection. So with regard to competition, moving to VoIP interconnection will lead to more competition in the market. The answer to the second question – whether consumers can trust VoIP interconnection -- is a qualified yes. Introducing Public ENUM would run the risk of consumers’ not properly configuring their entries in the ENUM database. But a more restricted version based on the Infrastructure ENUM model could offer a “best of both worlds,” in which the telephony provider would offer stability for some of the essential services such as voice, but could also offer flexibility to customers that require it.

The other four questions will serve as guidelines for the evaluation of the role of the regulator in each of the three main steps of interconnection: Lookup, signalling, transfer and transcoding.

5.1 Look-Up

Regarding the look-up process, regulators will have to ask themselves:

- Where will the look-up be performed?
- Who can perform the look-up (how open is access to the look-up function?)
- What will they find?
- Are there any special considerations with regards to the numbers found?

Before there can be a look-up, the source for the table in which the look-ups are done has to be determined. This source is normally the country’s numbering plan, which contains an overview of what numbers can be used for what purposes and/or geographic locations, what charges can be levied etc. The regulatory agency commonly is responsible for administering the numbering plan. Its main objective is that all the numbers can be reached, and under the correct set of rules.

In the SS7 world, every telephony service provider keeps some form of the numbering plan in a table, but there is no centralized database with all the numbers in them. This hasn't been a major problem in making every number reachable, although it has created problems with implementing a number portability solution. ENUM could very well take the role of a central database in some form, allowing the introduction of new features that are possible with ENUM. The resulting platform would be the national database to perform look-up functions for the provider associated with the number. It doesn't need to perform a role in the signalling and transfer and transcoding processes.

There are several things a regulator would need to keep in mind. First, a central ENUM platform, in which all telephone numbers and their respective routing information are contained, would also be the platform over which number portability could be executed using "All Call Query." With All Call Query the database gets checked for every number that resides outside of the operator's network. The benefit for the regulator would be to eliminate the involvement of donor networks in interconnecting with ported numbers. It would result in quicker porting between networks and would free the regulator from having to regulate the behaviour of the donor network.

Second, regulators should consider whether a national, centralized numbering platform would include the actual interconnection of signalling and media (data packets with voice/video), or whether this is not something the regulator should be involved in. It would be possible to provide a dedicated platform for interconnecting media as part of the numbering platform. But it may also be that the telephony service providers prefer the media interconnection to happen over a national Internet exchange or through completely separate systems and networks, like the GSMA's Pathfnder or Xconnect or a federation of their own design.

Third, regulators should consider whether to open the platform to non-traditional telephony service providers. ENUM and SIP are well supported on commercial and open-source PBX systems, which are similar or identical to the ones used by VoIP providers. Open platforms would allow medium and large enterprises to interconnect and manage their own telephone numbers, much as they now manage their own internal telephony networks or their DNS entries with a ccTLD. The effect would give companies and governments the ability to interconnect with telephony providers on the same level -- essentially becoming their own telephony providers.

Fourth, the regulator will have to consider whether the number database is accessible over the internet, a virtual private network (VPN), or only through a dedicated network. There are different economic and security considerations with each decision. As with openness, the easier it is to connect, the more market players will join.

Fifth, the regulator will also have to consider if the database can contain entries beyond just references to telephony. As said earlier, the Public ENUM model may be difficult to attain, but its vision of putting consumers in control of their communications is a vision worth pursuing. The more flexibility is build into the system, the bigger the chance that competition will force market players to allow their customers to interconnect in new ways.

Finally, privacy is also a concern with a centralized database, based on All Call Query. Analysis of a centralized database could give unrivalled insight into the calling behaviour of a nation. Periodic updating of local caches at each of the telephony providers of the central database might prevent such a window being opened up into national calling patterns.

Part of the process of performing the look-up is determining whether any special considerations apply to the number. This most often applies to routing of an emergency call. However, when moving to VoIP interconnection there may be new services to which the regulation will also need to be applied. Examples that the regulator will need to consider are:

- How to deal with the interconnection of emergency calls. Quite a lot of work has been done in standardization commissions on making VoIP services comply with emergency number regulations. One can think of the work done on E911 in the United States, for example. Most of that work has been focussed on getting VoIP to work like traditional telephony -- for instance, to get the correct location of the IP address where the user was located. What has been less looked at is what new possibilities do VoIP-services offer that may be of use to emergency services. One could think of the transmission of video as part of an emergency call, for example.[29] This is technically possible and potentially beneficial to emergency services.

- How to deal with interconnection of other services. Because the ENUM database could be used to interconnect services that use no voice at all, some rules and regulations might not apply at all. For example, rules that apply to time and duration of a call may not apply to a service that uses text as its basis for communication.

5.2 Signalling, Transferring and Transcoding

Under normal circumstances market parties should be able to negotiate how to signal, transfer and transcode without regulatory intervention. The regulator could intervene only as a last resort, to settle disputes if parties couldn't agree. The flexibility that SIP and ENUM offer should allow the technology to take care of differences in standards and protocols. Regulators should try to protect the market from country-specific dialects of these standards. These only increase the barriers to entry in the market.

The use of federations or provider-specific platforms for interconnection can be positive for the market as long as they don't lead to anti-competitive behaviours or, worse, result in certain numbers becoming unreachable. Regulator will have to ensure that there is always a competitive way of reaching all numbers. The traditional regulatory toolkit is probably well equipped to ensure this by requiring that parties negotiate and by *ex post* imposition of federation rules if and when a federation becomes cartel-like.

An issue that regulators may be faced with is the issue of Net Neutrality. This is a highly contentious issue that is too wide and difficult to be discussed here. For VoIP interconnection, it concerns the question of whether a provider of a telecommunications network or service -- for instance, a mobile broadband Internet service provider -- is allowed to degrade or block the delivery of VoIP service offered by another telephony provider over its network. The mobile broadband Internet service provider certainly might be inclined to do so because the VoIP service competes with its own telephony offer.

Billing

Regulators have been very involved in overseeing billing for interconnection. There are two reasons:

- The need to regulate monopolies' interconnection charges; and
- The application of taxes on telephony services for specific policy purposes (such as universal service requirements).

The main reason for this is the natural monopoly that exists when a market player sets a termination charge. Either the calling party or receiving party pays a charge to the other network for sending or receiving the call. In setting such charges, regulators use models such as Long Run Incremental Costs (LRIC), as well as benchmarking. Essential in such a model is to define the costs that are to be calculated. The question regulators will have to ask themselves is whether, with VoIP, the traditional calculations still apply now that the telephony provider having the number may not actually have a contract with the network that is transmitting the call. If VoIP (or voice) is a service, would the costs for the network still be part of the model? This doesn't mean that VoIP comes at no cost to a telephony service provider, just that the average per-minute cost of adding more capacity to the platform is close to negligible, when the cost of running a national network aren't factored into the per-minute price. It may be that these costs don't need to be factored in, because they are paid for in other ways, such as through monthly charges to customers for the broadband connection, or by the VoIP telephony provider buying Internet traffic and voice transit.

Another question is how to charge for calls terminated on a private network if the private company self-provides its own telephony capability. For instance, if a country's largest bank became its own telephony provider, connecting directly to the national number database to perform look-ups, would it be allowed to charge other networks for incoming traffic?

Universal Service Obligations are another source of regulatory involvement in telephony interconnection. In many countries, the telecommunication network is provided below cost to special groups (the disabled, veterans, elderly). This is often financed by levying charges on the per-minute cost of telephony or on the cost of interconnection. A similar source of income stems from charges levied on incoming and outgoing international traffic. For developing nations, this is often a source of foreign currency inflows, partially financing the national telecommunication infrastructure. A move to VoIP interconnection is often seen as threatening the financing of the USO obligation or the income from international interconnection. As explained earlier, the technological change in itself doesn't warrant this fear. VoIP interconnection can support existing payment models. However, the underlying trend of a move to IP networks, making telephony just a service over a packet-switched network, is a clear threat to this source of income. Countries levying incoming telephony interconnection charges above the cost of an equivalent amount of IP-transit will continue to see citizens, foreign residents, travellers, companies and telephony service providers try to evade these charges.

6 Conclusion

This chapter focuses on VoIP interconnection and discusses some of the possible future developments in this area. One could say that the focus is on the future of voice interconnection in general, as it is widely agreed that VoIP will be of increasing importance in the coming years. This chapter has aimed to show three facts:

1 VoIP by itself doesn't bring dramatic changes to the process of interconnection and its technical implementation. Many of the changes that VoIP is expected to bring might already be possible in the current telephony networks.

2 VoIP interconnection can support the same business and regulatory practices that exist in the current market. Combined with changes to business models and regulatory regimes, however, VoIP can greatly decrease the cost of telephony and increase its use, uses and usefulness.

3 Interconnection in itself is an easy-to-understand process that is made more difficult by business regimes and regulatory arrangements.

The technology for the look-up process can be changed and improved dramatically with a move to ENUM. It would allow countries to move to a single centralized database that holds all numbers. It would also allow number portability without involving donor networks and, therefore, would result in less distortion to competition. Most likely, countries will not choose a Public ENUM database, which would require end users to be responsible for the accuracy of all data. But an Infrastructure ENUM database is a more realistic scenario, because it retains the telephony provider's responsibility but allows the user to provide some data, too. The signalling, transferring and transcoding can take place through a number of possible platforms and technologies, and this should be seen as benefitting competition in the market. If competition forces work properly, it shouldn't require the intervention of the regulator.

The analysis of the business of interconnection shows that voice has now become a service much like email or the World-Wide Web. Its costs, and the customers' willingness to pay, are converging to the same price level as other IP services. There is a great difference in the price of interconnecting IP traffic and the price for interconnection of a telephone call. This will lead to a continuing downward pressure on the price of interconnection and will continue to give opportunities for arbitrage.

Regulators will appreciate the opportunities VoIP and VoIP interconnection offer to increase competition in the telephony service market, with new market players entering, new services being enabled and the possibility for cheaper services. ENUM offers possibilities to build a national number database with number portability integrated into it. In order to enable these possibilities, some regulatory reform may be necessary to remove the codification of old business practices that now block the introduction of new business practices. Technological and business model neutrality should be the norm.

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[1] ITU World Telecommunication Indicators Database 2009.

[2] Trillion with a T, the Newest Giant Industry has Arrived: the Money and Meaning of Mobile, Tomi Ahonen, http://communities-dominate.blogs.com/brands/2008/12/trillion-with-a.html

[3] The standardization organisatio for the future mobile networks 3GPP[3] has subscribed to two IP-technologies called Electronic Number Mapping (ENUM) and Session Initiation Protocol (SIP) as the technologies it will base part of its standards on. A move to for instance a national number portability database based on ENUM might then make sense.

[4] For GSM this is the IMEI number. Other mobile wireless telephony protocols may use different identifiers.

[5] Names are not fixed, mobile, geographical etc. Only regulation limits the application of names.

[6] It would go to far to explain the OSI layer model here. For an excellent explanation the reader is invited to read prof. Tanenbaum's book Computer Networks, of which the first chapter with the explanation can be read online at http://authors.phptr.com/tanenbaumcn4/

[7] Source Telegeography http://www.telegeography.com/products/tg/index.php

[8] Dryburgh, Lee; Jeff Hewitt (2004). *Signalling System No. 7 (SS7/C7): Protocol, Architecture, and Services*. Indianapolis: Cisco Press. ISBN 1-587-05040-4. This book can be read online at http://www.ss7-training.net/

[9] There are 5 different types of the so called “All Call Query”, where the telephony provider always checks a centralized database. Those are explained below:

- All Call Query (ACQ): The operator that originates the call always checks a centralized database and obtains the route to the call.
- Query on Release (QoR):The operator that originates the call first checks with the operator to which the number initially belonged, the donor operator. The donor operator verifies the call and informs that it no longer possesses the number. The operator that originates the call then checks the centralized database, as is done with ACQ.
- Call Dropback: Also known as Return to Pivot (RoP). The operator that originates the call first checks with the donor operator. The donor operator checks its own database and provides a new route. The operator that originates the call then uses this route to forward the call. No central database is consulted.
- Onward Routing (OR):The operator that originates the call checks with the donor operator. The donor operator checks its own database and obtains a new route. The operator to which the number was designated routes itself the call to the new operator. This model is called indirect routing.
- Broadcast changes: The central organisation that facilitates number portability broadcasts (or pushes) the changes to network operators who incorporate the changes in their routing tables.

Source: Dryburgh, Lee; Jeff Hewitt (2004). *Signalling System No. 7 (SS7/C7):* Broadcast Changes is specific to the Dutch situation and not mentioned in Dryburgh and Hewitt.

[10] As mentioned in the VoIP chapter of the 2004/2004 Edition of ITU's Trends in Telecommunications Reform report, ENUM is a protocol developed by the Internet Engineering Taskforce (IETF) that defines a domain name system (DNS)- based architecture and protocol aimed at using a telephone number to look up a list of IP service addresses (e-mail, IP phone addresses, URL, SMS, etc). http://www.itu.int/ITU-T/inr/enum/

[11] RFC (Request for Comment) are the standardization mechanism of the IETF. They specify all kinds of protocols, DNS, IP, ENUM, e-mail. The RFC’s can be found here http://www.ietf.org/rfc.html

[12] http://www.enumdata.org/ lists 9 active Austria, Czech Republic, Finland, Germany, Ireland, Poland, Romania, The Netherlands, United Kingdom. There have been more delegations.

[13] There are several open source DNS implementations that are of such quality that well known vendors like Siemens base their commercial ENUM platform on these programs (in this case BIND and PowerDNS).

[14] ITU-T Rec. Q.9 (11/88) Vocabulary of Switching and Signalling Terms.

[15] The H.323 is an ITU-T recommendation that was developed for VoIP sessions. However it has been succeeded in most implementations by SIP.

[16] H.323 was the First switching protocol for telephony over internet as such it is still widely supported. For the purpose of this article

[17] SS7 can also be handled over IP connnections using a protocol called SIGTRAN however it would still require the use of other SS7 particular functions and technology.

[18] There is no such thing as a public internet. When telecommunications networks use globally routable IP-adresses (Not the ones starting with eg. 192.168) to interconnect, this communication is part of the internet. They may have restricted access to these IP-adresses to specific other IP-adresses or even physical fibers, but it is part of the Internet. Whether or not other traffic like web, bittorrent, or Youtube shares that connection with VoIP is also irrelevant, because the network is indifferent to the types of traffic on it. What operators will need to do is engineer the network in such a way that it can handle peak traffic. This was the same in circuit switched networks.

[19] The digital encoding of voice on telephony networks generally uses either the ITU’s G.711 recommendation. This is a 64kbit/s encoding. In 2008 G.711.1 was introduced to enable higher quality voice at higher bitrates. For wireless networks different codecs, like AMR are used, which generally use up to 12kbits of encoding.

[20] There is no consensus on when this happened. A Google search mostly reveals late 1999 or early 2000-2001. Other sources indicate later dates.

[21] http://internetthought.blogspot.com/2009/03/calculation-of-how-iphone-uses-640mb-of.html

[22] Source website T-Mobile NL url?

[23] http://www.thevpf.com/ statistics are on the front page

[24] For an explanation of Peering and Transit see http://arstechnica.com/old/content/2008/09/peering-and-transit.ars

[25] The costs of a connection at AMS-IX or a similar internet exchange were used. The per port costs are as reported by experts in the industry the author communicated with.

[26] 1mbit/s can sustain 12 voip-lines at 80kbit/s. Given that transit is based on peak load and that there are day and night and week and weekend differences in call patterns, it would be fair to expect that the full amount of possible voice minutes per month will not be used. On a 30-50% utilization, a 1 mbit/s line would support 30 days * 24 hours * 60 minutes * 12 lines * 30/50%= 150.000-250.000 minutes. See also http://www.thevpf.com/solution/mpm

[27] Bill and Keep is the common system in countries in North America, Singapore and Sri Lanka. It is also the system used to interconnect IP-traffic between networks, where it is known as peering and transit. For a discussion of Bill and Keep vs RPP or CPP see the Discussion Paper "Interconnection in an IP-based NGN Environment" by Scott Marcus presented at the Global Symposium for Regulators (GSR) 2007, at: www.itu.int/ITU-D/treg/Events/Seminars/GSR/GSR07/discussion_papers/JScott_Marcus_Interconnection_IP-based.pdf.

[28] Taken from the mission statement of the Dutch Regulator OPTA

[29] Much of the focus with emergency services has been on how VoIP-providers support VoIP. If VoIP interconnection moves inevitably to ENUM/SIP, governments should pay attention to when their emergency services support VoIP and associated services natively. It is now possible with many 3G and VoIP services to use video. It would be useful for emergency services to receive video and other services, though many of them can't.