COST IMPLICATIONS OF DEPLOYING A VOIP NETWORK

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Abstract:
VoIP is a technology that is widely used worldwide and that has been and will be implemented by current and prospective telephony operators. As the call costs are regulated in many countries, it is important for a National Regulatory Agency and for operators to obtain information about the costs of deploying a VoIP network. This paper aims to address the main points that should be considered at the moment of determining the cost of a VoIP network. To this end, in the paper we describe several scenarios that reflect the current deployment of VoIP networks.

Keywords: VoIP, cost, deployment, termination charges

Paper submitted to the 19th European Regional ITS Conference, Rome, Italy, September 18-20, 2008

1 Introduction

For many years the cost implications of building and operating a Public Switched Telephony Network (PSTN) have been defined by using established procedures. However, with the introduction of new technologies and networks, the cost implications of the networks may change. The voice, video and data networks are converging into one network that in many cases uses IP as the main transmission protocol. In this sense, it is still unclear how the cost structure and proportion for the voice, video and data services will be defined.

For the case of the transmission of voice, probably the most dramatic change lately has been the appearance of the Voice over IP (VoIP) technology. There are operators such as Skype that use this technology as the cornerstone of their network nodes and for the transmission over the Internet whereas other telephony operators use IP partially in their transport networks. In many cases traditional PSTN networks are interconnected with VoIP operators, which leads to the definition of different interconnection arrangements. The implications of interconnecting IP and VoIP networks have been studied[1]. On the other hand, VoIP is largely used inside private IP networks by corporate users. To sum up, there are several ways of using the VoIP technology and, as the provisioning of voice could be subject to regulation, the cost implications of deploying a VoIP network is a matter of interest for current and potential VoIP providers.

As far as the authors know, there are a few studies about VoIP cost implications[2][3], but they have not addressed this topic in detail recently. In the paper, we describe several scenarios that reflect the current state of the art of VoIP networks, the network elements involved and the corresponding cost implications.

Section 2 of the paper describes the systems, protocols and additional technical aspects of a VoIP network. Section 3 contains the different scenarios that are currently employed to deploy VoIP networks and the corresponding cost implications. Finally, Section 4 addresses the conclusions.

2 VoIP Techniques

This section explains the technologies needed to deploy VoIP services. First of all, speech codes are described in Section 2.1. The next Section describes the end-to-end VoIP transmission protocols. Section 2.3 explains the main VoIP Architectures: SIP, H.323, Asterisk-based, Skype, etc. Finally, Section 2.4 depicts elements related to the provisioning

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1 Marcus, J. Scott and Elixmann, Dieter (2008), The Future of IP Interconnection, Technical, Economic, and Public Policy Aspects, WIK-Consult Report for the European Commission, January.
2 InfoDEV/ITU ICT Regulation Toolkit, 14.5.3 Implications of VoIP for Interconnection Pricing, available at http://www.ictregulationtoolkit.org/en.
3 Weiss, Martin B.H. and Kim, Hak-Ju (2001), Voice over IP in the Local Exchange: A case study, 29th TPRC Conference, USA.

of telephony services, such as numbering, number portability, access to emergency services, and security issues.

2.1 Voice codecs

The Voice over IP codecs are necessary to convert the analog voice signal into a digital signal that travels over IP networks. In VoIP networks different types of codecs are used and they vary in terms of bandwidth consumed, sound quality, computational effort, etc. Some of the most widely used and known codecs are the following ones: the ITU-T codec G.711 that uses PCM and needs 64 Kbps; the ITU-T codec G.726 that employs ADPCM and requires 16, 24, 32 and 40 Kbps; the ITU-T G.729 codec that uses CS-ACELP and needs 8 Kbps; Speex which is an open-source voice codec that requires 2.15 to 44.2 Kbps; and LPC10 that needs 2.5 Kbps and that is used for narrow bandwidth connections. Several VoIP equipment manufacturers and VoIP operators employ proprietary VoIP codecs. For example, Skype uses proprietary and encrypting codecs.

2.2 End-to-end VoIP transmission

In a packet-switched network, the packets can be delayed, lost, corrupted and transmitted in disorder. Therefore, for delay-sensitive applications such as voice, end-to-end techniques that ensure an acceptable level of transmission of voice packets are needed.

At the transport layer there are two main protocols employed for the transmission of data and signalling on the Internet: Transmission Control Protocol (TCP) and User Datagram Protocol (UDP). TCP is a robust protocol that ensures a reliable transmission; packets are retransmitted, when they got lost, before the next block of packets may be sent. On the other hand, the TCP headers are large and the transmission delay of TCP packets can be relatively high. Therefore, TCP is not used for the transmission of voice packets. But for the transmission of signalling information a few VoIP architectures use it due to is reliability. Thus for the transmission of voice packets the UDP protocol is employed by several VoIP architectures due to its low transmission delay. Lost packets then are not retransmitted but reduce the voice quality.

Real-Time Transport Protocol (RTP) and Real-Time Control Protocol (RTCP) are IETF standardized protocols suited for the transmission of multimedia services (voice and video). In many cases they are employed together. Two of the main important features of RTP are the sequence numbering and the time stamping possibilities. The RTCP protocol manages information about transmission parameters such as lost packets, jitter, round-trip delay, etc. This information can be used by applications that want to improve the Quality of Service. When the RTP protocol is used the voice packets carry IP/UDP/RTP headers.

To ensure the end-to-end Quality of Service there are several techniques that can be employed. The Integrated Services technique requires that in every switch or router along the end-to-end path there are enough resources available for the flow. In this sense, the

Resource Reservation Protocol (RSVP) is in charge of reserving the bandwidth for a specific flow. The Differentiated Services technique works by classifying the traffic in every switch or router according to the priority assigned to the packet, thus offering a relative quality. Another approach is not to use tools as described before but to dimension the capacity that large that in (probably) no case the transmission quality will be affected (Over-Dimensioning).

2.3 VoIP Architectures and Protocols

This Section describes the most common architectures and protocols used to deploy VoIP services. There are several standardized network architectures and protocols that could be used to deploy VoIP services, such as the ITU-T Rec. H.323 system, the IETF Session Initiation Protocol (SIP) protocol, the Media Gateway Control Protocol (MGCP), and ITU-T Rec. H.248 protocol (MEGACO). In addition, VoIP open standards such as IAX2, the Inter-Asterisk protocol that work with the Asterisk open source PBX server, are gaining importance in some VoIP deployments. Skype is a good example of a proprietary VoIP system.

On the other hand, the Softswitch can be used for the interconnection of a PSTN network with a VoIP network. The IP Multimedia Subsystem (IMS) is an architecture suited to create and deliver services easily.

2.3.1 H.323 Architecture

The H.323 system is an ITU standard. In this architecture a gateway is in charge of sending the VoIP packets to a destination gateway. For this purpose the Real Time Protocol (RTP) is used. The H.225 and H.245 protocols are used for controlling the call. Moreover, a Gatekeeper is needed to control the Gateways.

2.3.2 SIP Architecture

The SIP architecture was defined by the IETF standard. A SIP architecture consists of two basic elements: the SIP clients and the SIP servers. Additional elements are also needed to manage and provide additional services: billing and voice mail servers, media gateways, etc.

2.3.3 VoIP Open Standards – IAX

Whereas SIP is an IETF official standard, the Inter-Asterisk protocol (IAX2) was defined as part of a community effort. IAX2 is the open standard Asterisk PBX protocol and it enables connections between servers and clients. IAX2 transmits the payload and the signalling information on the same UDP data stream, which helps to enhance its performance. IAX2 is an alternative to SIP-based solutions[4].

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4 Escudero-Pascual, Alberto and Berthilson, Louise (2006) – VoIP-4D Primer. Building Voice infrastructure in developing regions

2.3.4 The Skype Architecture

Skype is a proprietary P2P software. The Skype architecture consists of nodes and supernodes[5]. In order to use the software a terminal with connectivity to the Internet is needed. The interconnection with the PSTN network is provided by means of the Skype-Out and Skype-In services.

2.3.5 Softswitch

The softswitch is a piece of software that switches calls by means of software instead of switching them through a hardware device. The softswitch architecture requires a Media Gateway Controller (MGC), which is the softswitch itself, and Media Gateways (MGs) and Signalling Gateways (SGs). The Media Gateways convert PSTN voice calls into IP packets and vice versa, whereas the Signalling Gateways adapt the SS7 protocols to the signalling protocols used in the IP network.

The softswitch technology separates service access from service control; at the same time it uses an IP-based core layer in the switching network[6].

2.3.6 The IP Multimedia Subsystem (IMS)

IMS is an architecture designed to provide services in a converged multimedia environment. As the voice is becoming just another service that can be offered in a telecommunications network, it is necessary to work with a platform that manages multimedia services. The IMS defines a horizontal architecture where the application layer, the control layer, and the access/transport layer are clearly separated and do not necessarily belong to the same operator.

Regarding the voice service, it is still unclear whether operators are choosing the IMS or the softswitch solution. If an operator wants to offer end-customers a basic voice service with a few related applications, then probably the softswitch is the better alternative. On the other hand, if the operator is planning to offer multimedia services, the IMS alternative takes relevance and probably it could be the better alternative.

2.3.7 Next Generation Networks (NGN)

A Next Generation Network is a packet-switched network capable of offering multimedia services with QoS features. One of the significant characteristics of this network is that the application services layer is totally independent of the transport layer[7]. The PSTN network

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5 Baset, Salman A. and Schulzrinne, Henning (2006): An Analysis of the Skype Peer-to-Peer Internet Telephony Protocol, Proceedings of the INFOCOM ´'06, Barcelona, Spain, April

6 Shichang, Xiao (2007), IMS Soft Landing, The integration of softswitch and IMS, Huawei Technologies, Issue 31, June.

7 ITU (2004) International Telecommunications Union, NGN Working Definition, available at http://www.itu.int/ITU-T/studygroups/com13/ngn2004/working_definition.html.

will sooner or later migrate to a NGN network, and one of the open issues is how the migration is going to take place.

The NGN network has basically two components in the transport layer: the core network and the access network. In the core network there is a high-speed IP network with QoS features. The deployment of a core network does not pose a problem to operators, whereas the access network is critical due to the high costs associated with the deployment of the last-mile. The PSTN operators intend to continue making profit of their existing infrastructure (tandem switches, local exchanges, copper wires, etc.), but at the same time they are conscious that due to the competence and to the bandwidth-hungry applications demanded by users a new broadband infrastructure should be installed.

So far, it is unknown what type of infrastructure will be deployed in the last mile. For the access the operators can choose between several solutions: FTTN (VDSL2), FTTB (GPON, VDSL2), FTTH (GPON, Fiber P2P), etc. For some of these access networks the copper cable can be kept in the very last end, whereas for other types of architecture it is not possible. In Europe, a few countries have launched public consultations about NGN access networks and, even though the regulatory framework is valid for all the EU member countries, the specific remedies applied in every country tend to vary significantly.

It is clear that the softswitch is a necessary element to interconnect legacy PSTN networks with IP networks. The Softswitch can replace the Class 4 TDM and Class 5 TDM switches. Furthermore, the IMS architecture is helpful to offer different types of applications. Therefore, it is up to the operator to decide whether the two platforms (softswitch/IMS) will coexist and for how long.

2.4 Additional Elements

2.4.1 Numbering - Number Translation - ENUM

ENUM is an IETF standard used to transform PSTN E.164 telephone numbers into correspondent VoIP addresses and vice versa. Without the mapping function it would be impossible to make a call from an IP phone to a PSTN telephone. DNS servers are used for setting up the call.

2.4.2 Number portability

Number portability is a feature that permits telephony subscribers to maintain the telephone numbers when they change the telephony provider or move to a new location[8]. In the PSTN world the Number Portability is provided by means of a database that belongs to the Intelligent Network. When it is detected that a destination number has been ported a query is

8 Ivcek, Mario (2007), ENUM based Number Portability in VoIP and IMS Networks, Conference on Telecommunications and Information (MIPRO 07), Opatija, Croatia, May 23-25

submitted to the Number Portability Database (NPDB), which knows the current location (operator) of the destination number.

In a VoIP environment there is also a consultation to a database or a server. As there are several VoIP architectures, there are different ways of implementing the number portability function. If the ENUM number translation is used, there will be a consultation to a DNS.

2.4.3 Access to Emergency Services

According to the EU 2002/22/EC Directive (Universal Service Directive) of 7 March 2002, 112 is the single European emergency call number. The VoIP service can belong to one of the following two categories: Electronic Communication Service (ECS) and Publicly Available Telephony Service (PATS). The ECS category targets a wide range of services as it "consists wholly or mainly in the conveyance of signals on Electronic Communications Networks"[9]. A PATS service, among other requirements, must provide access to emergency services. The EU countries can define whether the VoIP service belongs to the PATS or to the ECS category.

The difficulty in providing access to emergency services depends on the location of the VoIP user[10][11]. If the VoIP user is non-nomadic and generates the emergency call from the address that was given to the VoIP operator, then it is technically feasible to route the call properly to the nearest Public Safety Answering Point. Conversely, if the VoIP user is nomadic and generates the emergency call from a location different to the address provided to the VoIP operator, then it may not be technically feasible to route the call properly to the closest Public Safety Answering Point (PSAP)[12]. Even though there are ongoing efforts to find a solution for this drawback, such as the one developed by the IETF ECRIT Working Group[13], nowadays this problem remains an open issue and it will pass some time until a standard is accepted by the ISPs and VoIP providers.

The nodes that provide the VoIP service, such as the routers, softwitches, etc. have to be able to route emergency calls to the proper destination number.

2.4.4 Security issues

Two aspects are related to the provisioning of call information to security forces: lawful Interception and data retention. Lawful interception is the obligation of providing access to

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9 Directive 2002/21/EC of the European Parliament and Council (Framework Directive), 7 March, 2002
10 ERG Common Position on VoIP, ERG(07)56rev2, December, 2007.
11 Elixmann, Dieter, Marcus, J. Scott, and Wernick, Christian (2008), The Regulation of Voice over IP (VoIP) in Europe, WIK-Consult study for the European Commission.
12 The VoIP network may be able to route the call to a PSAP, but the PSAP has to be the closest to the actual location of the VoIP user. Therefore, the actual location of the VoIP user should be transmitted by the VoIP user.
13 IETF ECRIT Working Group Webpage, available at http://www.ietf.org/html.charters/ecrit-charter.html

telephony calls for law enforcement agencies and intelligence services, whereas data retention refers to the storage of call detail records of telephony traffic for a period of time.

At the moment of implementing the lawful interception mechanism one of the facts that has to be considered is that usually VoIP signalling and data take different paths. However, there are technical solutions to overcome this situation. On the other hand, if a nomadic user makes a VoIP call from an IP network that is not under the coverage of the security forces, it could be not feasible to intercept the call.

In the European Union the Directive 2006/24/EC of March 2006 on Data Retention requires member states to storage the following information for a period of between 6 months and 2 years[14]. It is necessary to retain data:

- to trace and identify the source of a communication;
- to identify the destination of a communication;
- to identify the date, time and duration of a communication;
- to identify the type of communication;
- to identify user's communication equipment;
- to identify the location of mobile communication equipment.

This Directive is currently under implementation in the member states. For the case of VoIP it is possible to find information about several of the above-mentioned items in the softswitch.

3 VoIP scenarios

In the last ten years people have witnessed the appearance of several ways to establish voice communications through IP networks. Overall, in all the countries corporative and residential users have become familiar with this technology. In this section different ways of establishing a Voice over IP connection are described.

3.1 VoIP calls with an end-to-end IP connection (IP end-to-end through several IP networks)

In this category, the end-to-end communication takes place exclusively over IP connections. There are no inbound nor outbound calls. The terminal can be an IP telephone; a softphone; or a POTS telephone that is connected to an Analogue Telephone Adapter (ATA) or to a VoIP card with A/D conversion capabilities. This case corresponds to a VoIP provider, not to a company that uses VoIP in several subsidiaries. This latter case will be explained in Section 3.2.

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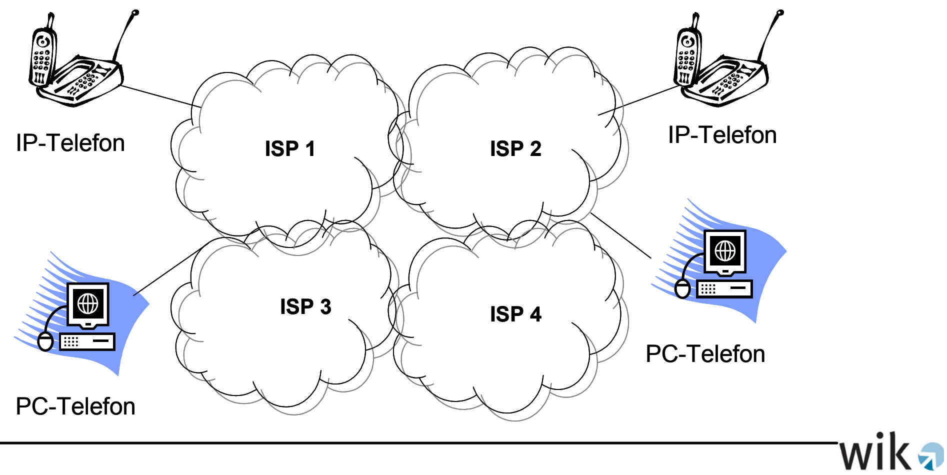
14 Directive 2006/24/EC of the European Parliament and Council, 15 March, 2006

The calls go through several IP networks that belong to different operators. The end-to-end quality is not guaranteed, unless every Internet Service Provider along the end-to-end path enables it. The Quality of Service depends on the quality of service of the network nodes. Figure 1 shows how the end-to-end connection is achieved.

Figure 1: VoIP calls through several IP networks



Skype with its PC-to-PC option is one company that uses this type of connection. Furthermore, Skype uses a peer-to-peer network. This technology requires that some nodes assume the role of ordinary hosts whereas others assume the role of supernodes.

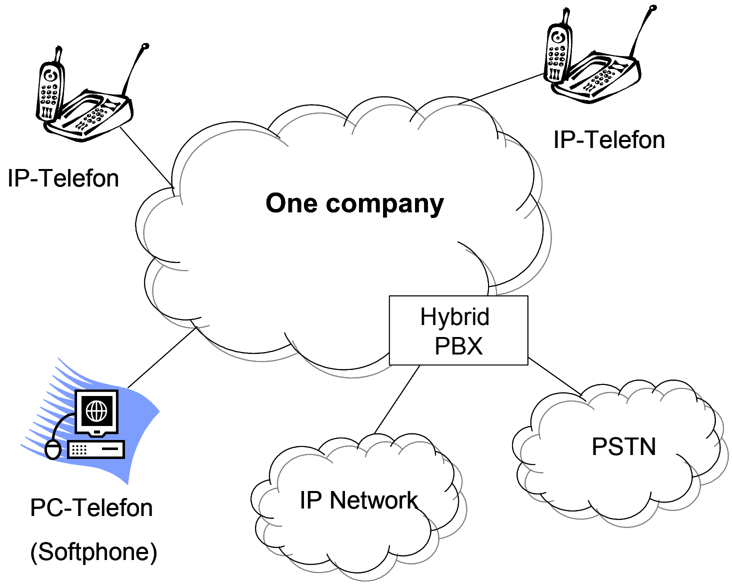
3.2 VoIP calls between an IP network and a PSTN network

This situation is more complicated than the previous one because the IP networking protocols should be adapted to exchange information with the PSTN nodes and protocols. Media Gateways are necessary to convert the IP packets into PSTN packets and, for the signalling layer, Signalling Gateways are needed.

3.2.1 VoIP calls between a private IP network and the PSTN network.

This modality comprises the calls made from IP phones or POTS phones with A/D converters to another VoIP terminal or to POTS telephones in the PSTN network. Figure 2 shows this basic scenario. The inbound and outbound calls pass through an Hybrid PBX.

Figure 2: VoIP calls between a private IP network and the PSTN network

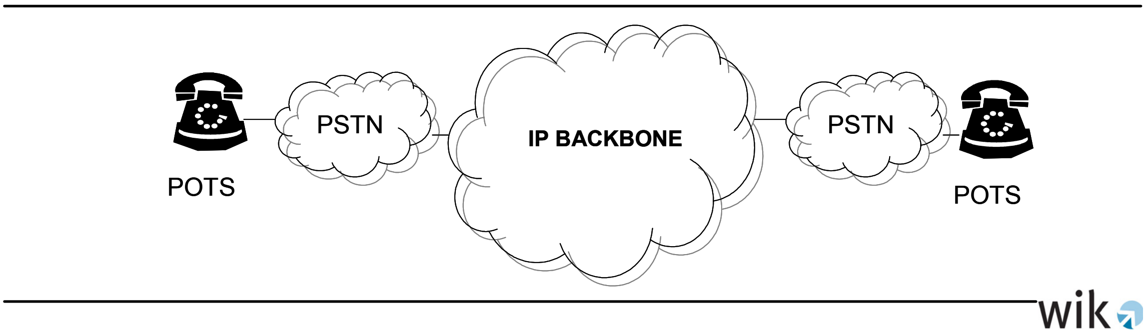


This case ranges from a small company that owns a private IP network with PCs working with a LAN network to a big company that has several subsidiaries inside the same city or in different cities. This case does not refer to a VoIP Provider and, therefore, it is no subject to regulation. It can be considered that the IP network belongs exclusively to the company. The QoS can be variable and depends on the priority given to the VoIP packets in the routers and switches.

3.2.2 The connection has an IP core network and a PSTN as access network (one operator).

In this scenario the PSTN network serves as access network. This situation, which is depicted in Figure 3, corresponds to what several telephony operators are doing at the moment. They use their deployed PSTN network as the last-mile connection whereas they work in the core network with a high-speed data connection that could be IP, Gigabit Ethernet, ATM, etc. Thus, they are indirectly VoIP providers. The PSTN access network can belong to the same operator of the IP backbone or not. For the interconnection between the PSTN and the IP backbone, several nodes and systems could be used, such as the Softswitch, Media Gateways, the IMS, etc.

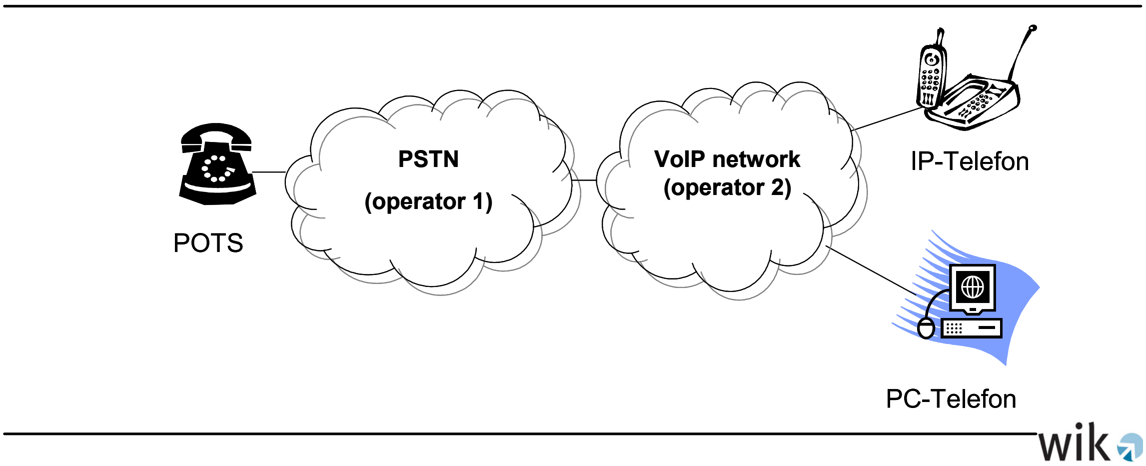
Figure 3: VoIP calls with PSTN as access network



3.2.3 A VoIP network has a connection with a PSTN (two operators).

In this scenario the PSTN operator is connected to a VoIP operator. The calls from the PSTN section are routed to the VoIP network. In this case the VoIP operator needs to have E.164 numbers. The quality of the connection will depend strongly on the quality of the VoIP network. Figure 4 shows the depicted scenario.

Figure 4: Calls between a PSTN and a VoIP network



Examples of VoIP providers are companies such as Vonage, Sipage, Skype with the Skype-in and Skype-out modalities, etc. Also callshops and prepaid calling card resellers that use VoIP belong to this category.

3.3 Costs considerations for the deployment of a VoIP network

As was explained in this Section, currently there are three basic scenarios that show how a telephony provider can offer the VoIP service:

1.- VoIP calls with an end-to-end IP connection (IP end-to-end through several IP networks)

2.- The connection has an IP core network and a PSTN as access network (one operator).

3.- A VoIP network has a connection with a PSTN (two operators).

In this analysis we consider the case of VoIP operators. We do not take into account companies or organizations that deploy private VoIP networks, which corresponds to the case explained in Section 3.2.1.

The following items should be considered at the moment of calculating the costs of a VoIP network:

Services provided: In addition to the provisioning of the voice service, a VoIP operator can offer additional services such as voice mail, call waiting, call forwarding, three-party conference, etc. If the bandwidth available is large enough, the videotelephony service could also be provided. For the provisioning of these services in some cases hardware or software elements are necessary.

Architecture of the network: In the past, every network was able to deliver one type of service: telephony, television or data. Currently in an increasing converged environment the networks carry several types of traffic. Therefore, one of the issues that should be addressed in detail is the proportion of the cost that corresponds to every type of traffic.

One of the possibilities consists in calculating the necessary bandwidth to deliver one service through one link and compare it to the total available bandwidth of the link. Another approach might be to take the traffic distribution of the different traffic types at the busiest hour as criterion to allocate the cost[15].

On the other hand, the transmission of voice is quite sensitive to delay requirements. Therefore, the intermediate nodes (routers and switches) should be able to manage QoS parameters, which increases the price of these nodes. Usually QoS is provided by means of priority and scheduling mechanisms.

With the deployment of Next Generation Networks the boundary between the transport and the access network is moving from the Main Distribution Frame (MDF) closer to the end customer. This boundary can be located in a node (FTTN), in the building (FTTB) and at the end user's office or household (FTTH). For NGNs the main components of the cost are the following ones:

- Transport Network: The transport network will consist in nodes connected through high speed links with fiber optic technology.

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15 Jay, Stephan, Anell, Patrick, and Plückebaum, Thomas (2008), "Netzzugang im NGN-Core", WIK study for the Bundesnetzagentur, Germany, June.

- Access Network: For the calculation of the termination charges the access network now should also be taken into account to a certain extend. The access network could use several technologies: copper cable or fiber optic. Cable networks and wireless networks could also be used.

Interconnection with other operators: The cost of the interconnection with IP and PSTN operators and the cost of the gateways should be considered.

Number portability – ENUM: The provisioning of number portability and of the translation of numbers implies the deployment of servers that support this feature.

Support of calls to emergency numbers: To implement this feature it is necessary to include a server that keeps the location of the VoIP user.

Lawful Interception: To implement lawful interception it could be necessary to allocate links, switches and databases.

Data Retention: The storage systems that keep information about customers' communications are mostly databases.

Billing systems: The prices of the databases that keep billing information need to be taken into account.

4 Conclusions

There are several ways of deploying the necessary infrastructure to offer a VoIP service. Depending on the business model of the VoIP operator, several alternatives can be considered. VoIP is a fairly mature technology, but it still needs to overcome a few drawbacks in order to be widely accepted. The main important problems of this technology are the necessity of minimum QoS levels in order to enable a good transmission of the voice, and issues related to the mobility of VoIP users (e.g. calls to emergency numbers and lawful interception). The cost items depend on the type of VoIP network that is deployed. Overall, it is considered that the following cost items should be taken into account: Services provided, architecture of the network (transport and access network), interconnection with other operators, number portability, calls to emergency numbers, lawful interception, data retention, and billing systems.

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