

Rendon, Juan; Plückebaum, Thomas; Bösch, Iris

Working Paper

Relevant cost elements of VoIP networks

WIK Diskussionsbeitrag, No. 316

Provided in Cooperation with:

WIK Wissenschaftliches Institut für Infrastruktur und Kommunikationsdienste GmbH, Bad Honnef

Suggested Citation: Rendon, Juan; Plückebaum, Thomas; Bösch, Iris (2008) : Relevant cost elements of VoIP networks, WIK Diskussionsbeitrag, No. 316, WIK Wissenschaftliches Institut für Infrastruktur und Kommunikationsdienste, Bad Honnef

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Relevant cost elements of VoIP networks

Juan Rendon
Thomas Plückebaum
Iris Bösch

Bad Honnef, Dezember 2008

**WIK Wissenschaftliches Institut für
Infrastruktur und Kommunikationsdienste GmbH**

Rhöndorfer Str. 68, 53604 Bad Honnef

Postfach 20 00, 53588 Bad Honnef

Tel 02224-9225-0

Fax 02224-9225-63

Internet: <http://www.wik.org>

eMail info@wik.org

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ISSN 1865-8997

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Zusammenfassung

Im Zuge der Konvergenz von Telekommunikationsnetzen werden Sprache, Daten und Video über ein gemeinsames physikalisch wie logisches Netz übertragen. Die Integration sämtlicher Dienste auf ein Netz ermöglicht den Netzbetreibern Kosten einzusparen. Kosteneinsparungen können dabei sowohl bei den OPEX durch die Nutzung nur eines, standardisierten IP Netzes realisiert werden als auch bei den CAPEX. Bei letzteren können die traditionellen E1 Schnittstellen des SDH (2 Mbps) durch eine deutlich geringere Anzahl von Hochgeschwindigkeitsschnittstellen (10 GBit/s) ersetzt werden. Eine in der letzten Zeit zunehmend auch in der Öffentlichkeit an Bedeutung gewinnende Technologie stellt Voice over IP (VoIP) dar. Auf Seiten der Netzbetreiber unterscheidet sich der Einsatz von VoIP grundlegend. Manche Netzbetreiber setzen VoIP lediglich im Backbone ein, andere hingegen realisieren den Transport für ihren VoIP Dienst über das öffentliche Internet (Skype).

Um die Terminierungsentgelte von VoIP Netzen bestimmen zu können, müssen in einem ersten Schritt die relevanten Kostenelemente identifiziert werden. Für die herkömmlichen Netze (PSTN und GSM) sind diese hinreichend bekannt. Mit Blick auf VoIP bestehen hier jedoch nach wie vor eine Vielzahl von Unsicherheiten bezüglich der genauen Diensteausprägung sowie der dazu erforderlichen Netzelemente. Dieser Bericht soll dazu beitragen, die bei VoIP noch bestehenden Informationslücken zu schließen.

Ziel dieses Berichtes ist es, die relevanten Kostenelemente von VoIP Netzen, wie sie von Sprachnetzbetreibern realisiert werden, zu beschreiben. Um diese Analyse durchzuführen, beschäftigt sich der Bericht mit den Besonderheit von VoIP und ihrer Bedeutung für die Kosten. Zu Beginn werden die charakteristischen Merkmale von IP Netzen beschrieben: Diese umfassen technische Anforderungen an den Endkunden, das Anschlussnetz, Konzentrationsnetz, das Kernnetz, Netzmanagement und Quality of Service in IP Netzen. Im Anschluss daran werden die bedeutendsten VoIP Realisierungsformen vorgestellt: Alternativen der Sprachcodierung und –decodierung, VoIP Architekturen und Protokolle (H.323, SIP, and Skype), Netzknoten und Systeme, wie beispielsweise Softswitches und IMS sowie Eigenschaften von herkömmlichen Sprachnetzen (Nummerierung, Notruf sowie Sicherheitsaspekte). Dabei werden verschiedene VoIP Architekturen miteinander verglichen, um die effizienteste Architektur zu identifizieren und somit ein Referenzmodell abzuleiten. An späterer Stelle des Berichtes werden Aspekte der Zusammenschaltung von VoIP Netzen angesprochen: Zusammenschaltung zwischen IP Netzen, IP Netzen und dem PSTN sowie Zusammenschaltung in einem NGN Umfeld unter Berücksichtigung von Kabel- und Mobilfunknetzen. Abschließend werden in diesem Bericht Besonderheiten von VoIP dargestellt wie beispielsweise Quality of Service in IP Netzen oder Aspekte der Zusammenschaltung zwischen Sprachnetzbetreibern.

Die zentralen Schlussfolgerungen des Berichtes sind nachfolgend dargestellt:

- *Verschiedene Realisierungsformen von VoIP:* Im Unterschied zum PSTN, das über eine überschaubare Zahl von Standards verfügt (z.B. PCM64 and SS7), die von den meisten Netzbetreibern realisiert werden, besteht in IP Netzen eine breite Palette an VoIP Systemen und Protokollen. Dabei variieren die zugrunde liegenden Realisierungsform von VoIP mit den Geschäftsmodellen der VoIP Anbieter. Die Anwendung eines SIP Softswitches konnte als 'state-of-the-art' Architektur identifiziert werden und wird als effiziente Architektur für ein Referenzmodell betrachtet.
- *Die Eigenschaften von VoIP Netzen erfordern den Betrieb von geeigneten Systemen und Speichermöglichkeiten:* Die zu berücksichtigenden Kostentreiber sind Nummern Portabilität (ENUM), Notruf, Billing Systeme, Kundenbetreuungssysteme (Customer Care Systems), Abhören sowie Vorratsdatenspeicherung.
- *Verschiedene Zusammenschaltungspunkte für die jeweiligen Typen von NGNs:* Es lassen sich 3 Basistypen von NGNs unterscheiden: Festnetze (PSTN oder NGN), Kabelnetze und Funknetze. Die Standortwahl für die Zusammenschaltungspunkte hat Auswirkungen auf die Terminierungskosten.
- *Quality of Service:* VoIP sind Schwankungen in der Dienstqualität inhärent und durch die charakteristischen Merkmale des IP bestimmt. Um Vorgaben beispielsweise hinsichtlich der Verzögerung strikt einzuhalten, kann der VoIP Anbieter verschiedene Techniken verwenden. So kann er Router oder Switches einsetzen, die QoS Mechanismen unterstützen. Alternativ kann er eine Überdimensionierung von Knoten und Kanten vornehmen. Welche Strategie gewählt wird, hat nicht zuletzt Auswirkung auf die Kosten von VoIP.
- *Zusammenschaltung zweier Netzbetreiber:* Zwei Aspekte sollen betont werden. Erstens, es ist noch unklar, über welche Standards und Schnittstellen im NGN die Signalisierungsinformationen ausgetauscht werden: auf Basis von IP oder SS7? Zweitens, die Zusammenschaltungspartner können ihre Leistungen untereinander auf der Basis von Calling Party Pays, Bill & Keep, oder einer Mischform aus beiden abrechnen.

Gemeinschaftliche Nutzung des Anschlussnetzes: Werden die Dienste über ein NGA realisiert, so sind Kosten des Anschlussnetzes ebenfalls anteilig zu berücksichtigen. Dabei muss jedoch beachtet werden, dass nur die dem Sprachverkehr zurechenbaren Kosten des NGA für die Terminierungsentgelte heranzuziehen sind.

Summary

With the convergence of telecommunications networks, the voice, data and video signals are transmitted over the same physical link. By using only one network for different services it is possible for the network operators to save costs. There can be OPEX savings by using the well-known and standardised IP network, and CAPEX savings by using high speed interfaces (10 GBit/s) instead of many E1 (2 Mbps) links. One of the technologies that has gained more acceptance among the public and that has been or will be implemented by current and prospective telephony operators is Voice over IP (VoIP). There are telephony operators that use VoIP only at the core network layer, whereas for other operators VoIP is the underlying technology (e.g. Skype).

In order to determine the termination rates in VoIP networks the relevant cost elements have to be identified first. For legacy networks such as fixed and mobile telephony networks there is wide experience about the precise cost elements that must be considered for calculating the cost of voice calls. However, VoIP is a disruptive technology and there are still several uncertainties about the precise services and network elements typical for a VoIP network. As there is a lack of information about this subject on the existing literature, the report describes this topic in detail.

The objective of this report is to determine the relevant cost elements of VoIP networks deployed by voice operators. To carry out the analysis, the report contains a discussion about the importance of several aspects of VoIP networks as cost elements. The report starts with a description of the following characteristics of IP networks: end-user requirements, access networks, aggregation network, IP core network, network and traffic management, and quality of service in IP networks. Then, the most relevant VoIP techniques are presented: voice codecs, VoIP architectures and protocols (H.323, SIP, and Skype), nodes and systems such as the softswitch and the IMS, and features of voice networks (numbering and addressing, emergency services, and security issues). This section compares different VoIP architectures in order to determine the most efficient technology for the definition of a reference model. Later the report touches aspects of interconnection of VoIP networks: interconnection between IP networks, interconnection of IP and PSTN networks, and interconnection in an NGN context of fixed, cable, and wireless networks. Finally, the report describes a few particularities of VoIP networks such as quality of service in VoIP networks, and interconnection issues between voice operators.

The main conclusions of the report are the following:

- *Different VoIP techniques:* Unlike what occurs in PSTN networks where there are a limited number of standards (e.g., PCM64 and SS7) that are followed by most operators, in IP networks there is a wide range of VoIP systems and protocols. Depending on the business model of the operator a specific VoIP system

will be used. The SIP softswitch is the actual state-of-the-art architecture and it can be considered as efficient reference model architecture.

- *The features of VoIP networks require the deployment of appropriate systems and storage space:* Cost drivers that should be considered are number portability (ENUM), emergency calls, billing systems, customer care systems, lawful interception, and data retention.
- *Different interconnection points for each type of NGN network:* There are three basic types of NGN networks: fixed networks (PSTN or NGN), cable networks, and wireless networks. The location of the possible interconnection points will have an impact on the cost of the termination rate.
- *Quality of Service:* The quality of the voice provided by VoIP service providers is in many cases variable. To satisfy the strict time delay requirements required by VoIP connections the VoIP provider could deploy routers or switches that support QoS mechanisms, or it could expand the capacity of the systems and links. The deployment of QoS mechanism has an impact of the cost of VoIP network elements.
- *Interconnection between two operators:* Two aspects should be taken into account. First, for the case of NGN networks it is not clear whether the operators will exchange signalling traffic through the IP or the SS7 interface. Second, the type of interconnection agreement between two operators could be Calling Party Pays, Bill & Keep, or a mixture of both procedures.
- *Common use of the access network:* If a NGA is deployed, one has to consider the cost of the access network to some extent as well, but also has to determine the appropriate share of voice and the other services in the access network in order to allocate the cost appropriately.

1 Introduction

1.1 Goal of the study

In the last ten years people have witnessed the appearance of several ways to establish voice communications by means of IP networks. In general, in all the countries corporate and residential users have become familiar with this technology. Several PSTN operators have started plans to migrate to VoIP whereas many of the new voice operators only consider the possibility of using VoIP from the scratch.

Usually in Public Switched Telephone Networks (PSTNs) and in Public Land Mobile Networks (PLMNs) the operators follow a Calling Party's Network Pays (CPNP) rule, which means that the operator where the call is initiated pays the operator where the call is terminated on a wholesale basis. This payment is known as the termination fee. As the telephone call can only be terminated by the service provider which controls the addressed telephone number, there is a form of market power (the termination monopoly). One of the ways that governments have to avoid the termination monopoly is by regulating the termination fee. Therefore, National Regulatory Agencies, government's bodies and telephony operators have an interest in determining the appropriate costs of the termination fees. Moreover, as it is foreseeable that the termination monopoly will continue in Next Generation Networks (NGNs), the need for regulating the termination fees will remain still for a longer period.

The research question of this report is the following: *What are the relevant cost elements of the VoIP networks deployed by voice operators?*

To this end, the report addresses the main cost elements of VoIP networks while editing a draft questionnaire for collecting the relevant cost information from VoIP operators. The answers shall enable the regulator to develop and feed a bottom-up LRIC cost model for calculating the efficient termination fees. The VoIP network should have an efficient architecture and, in this sense, it is important to determine the most efficient VoIP architecture among the existing solutions.

1.2 Methodology

To answer the research question posed in Section 1.1, the report consists of a review of the existing literature on the subject. The report classifies the different cost elements of a VoIP network.

So far there is extensive literature about the definition of cost elements of VoIP corporate networks. However, there is a lack of information about cost elements of public VoIP network operators.

Chapter 2 of the report explains the basic technology of an IP network. Initially, the end-user requirement is explained. Then, following network aspects are described: access network, aggregation network, IP core network, network and traffic management, and quality of service in IP networks.

As there are different types of VoIP networks, it is necessary to identify what are the VoIP systems that are being deployed currently. Chapter 3 touches initially the voice codecs, and then different types of VoIP architectures are explained: H.322, SIP, the P2P network Skype, the VoIP softswitch and the IMS. The realization of the following features of voice networks is also described: numbering and addressing, emergency services, and security issues.

Chapter 4 explains cases of interconnection of VoIP networks: VoIP basic scenarios such as interconnection of pure IP networks, and interconnection of IP and PSTN networks. Chapter 4 also explains the different interconnection points that could be considered inside three different types of NGN networks: fixed telephony networks, cable networks, and wireless networks.

VoIP has a few characteristics that make it different from other communications networks. Chapter 5 addresses therefore the following issues: cost elements of VoIP networks according to the specific service offered, quality of service, interconnection between operators, and common use of the access network. Chapter 6 contains the conclusions of the report.

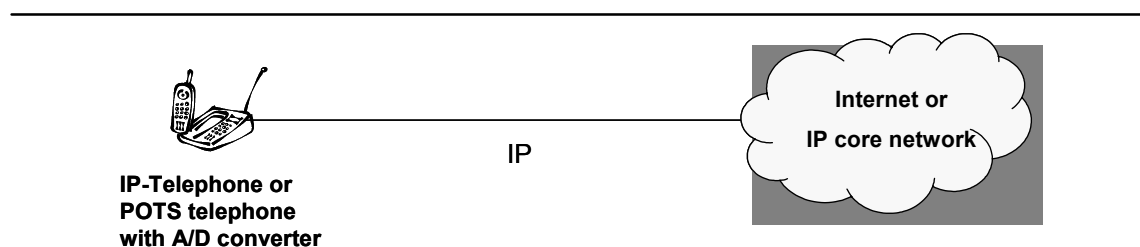
2 Transmissions over IP networks

For the provisioning of IP services the network components of the end-to-end transmission must have IP-compatible devices. In an end-to-end transmission there are two main sections to take into account: the end-user terminal and the network. Section 2.1 explains issues related to the end-user IP connectivity, whereas Section 2.2 describes several components of the network: access network, aggregation network, the IP network, and the Telecommunication Management network, and quality of service.

2.1 End-user requirements

This chapter describes scenarios where VoIP is employed by the end-user. A VoIP user has two possibilities of having a VoIP connection. In the first case the VoIP user has a direct access to the Internet or to an IP network, and the user requires an IP device. Figure 1 illustrates this case. In the second case, the end user can have access to the voice network through a PSTN network. Then, the PSTN voice signal can be converted by a Media Gateway into IP datagrams. Figure 2 shows this latter case. The end user is not using VoIP in the access, but the end-to-end communication employs VoIP in the core network. Several PSTN networks use IP (and VoIP) in the core network. The PSTN voice signal can be converted in a Media Gateway into IP datagrams.

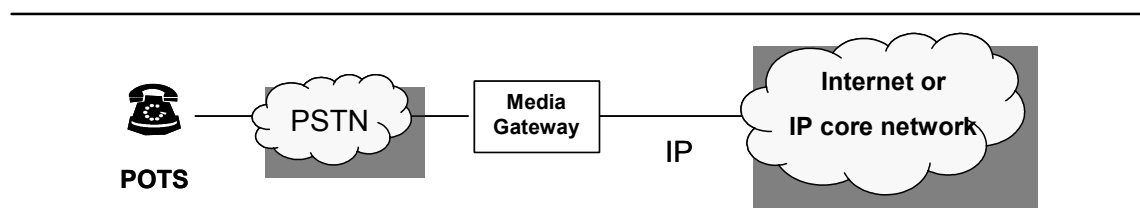
Figure 1: End-user with an IP device



Source: WIK

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Figure 2: End-user without an IP device

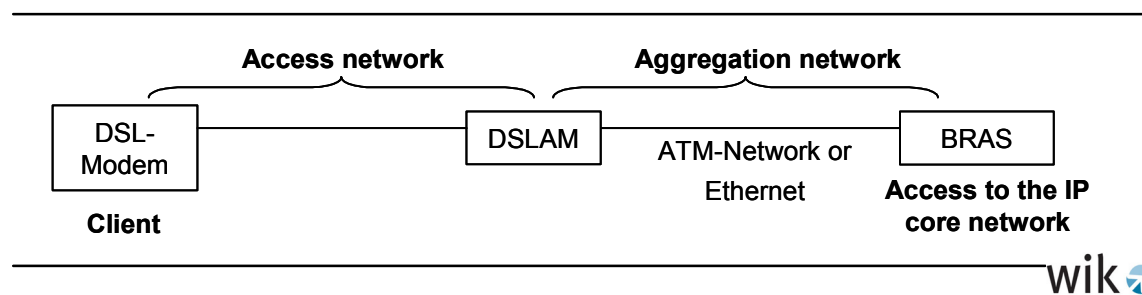


Source: WIK

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The end-user requires for the use of IP-based services access to the Internet, which can be a wired or a wireless access. The satellite links and the mobile and cellular access technologies are wireless technologies that can be employed. The cable TV, Power Line Communications (PLC), Digital Subscriber Line (DSL) and FTTx modalities belong to the category of wired access. The xDSL transmission technologies enable a high data transmission rate in comparison with the ISDN technique. For the use of the xDSL technique a DSL-Model is needed. Figure 3 shows the network access device required by the end-user.

Figure 3: Requirements for the provisioning of IP-based services



Source: WIK

The modem is connected directly to the PC (e.g. through the USB port) or to a local network through an Ethernet port or router, and it is necessary for the transmission of data to the subscriber line. The subscriber line can belong to the local network operator, which can directly charge the user for the use of its own facilities, or it can belong to another network operator, which charges the end-user's network operator for the use of the line.

The splitter is an element used in the xDSL access that separates the circuit-switched voice signal from the data signal. If VoIP is used on the data connection, then there are two voice streams present at the same time: the PSTN voice signal and the VoIP transmission over the data link. Currently there are cable operators that offer the voice service by means of the VoIP technology. In this case a splitter would not be necessary because there is no separation of frequency bands.

The DSL Access Multiplexer (DSLAM) is a node that is deployed between the access network and the aggregation network, and it will be described in the next section.

For the calculation of costs of VoIP termination rates it is necessary to calculate all the network elements of all the networks segments. Therefore, the definition of where VoIP really starts, which was explained in Figures 1 and 2, would not be relevant. On the other hand, the cost of the telephony device is not included in the calculation of the cost of voice calls.

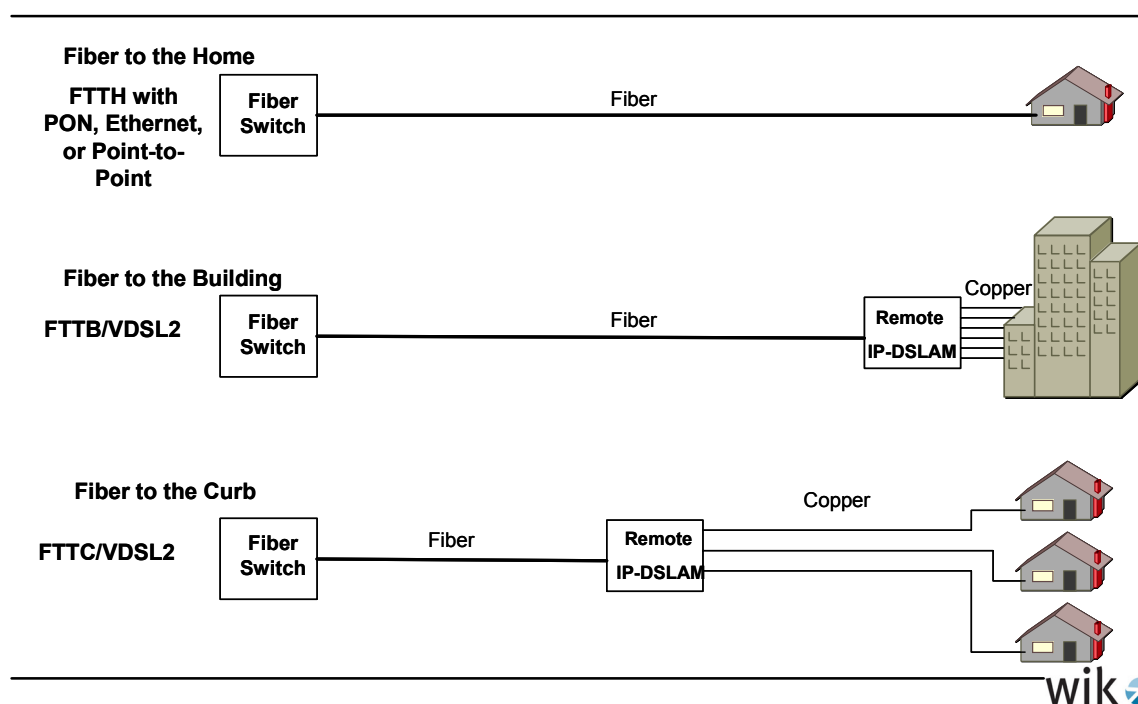
2.2 Network requirements

2.2.1 Access network

The most simple and conventional technique that can be used in the access is the technique based on the pair copper cable. The DSLAM is on the other side of the subscriber line and it can be deployed at the Main Distribution Frame (MDF) or at the street cabinets. The VDSL DSLAM in most cases is located in street cabinets close to the customer homes (Fiber to the Curb, which is explained below). The DSLAM contains slots for the line cards of the clients. The management of the aggregation network assigns a dynamic IP address to every user through the port of every user in the DSLAM.

Another possibility in the access consists in the deployment of Next Generation Networks (NGN) in the access (Next Generation Access Networks). A Next Generation Access Network in a fixed network implies in many cases the deployment of a fiber optic structure that supports IP-based high-capacity links. The following are the most famous FTTx technologies: Fiber to the Home (FTTH), Fiber to the Building (FTTB) and Fiber to the Curb (FTTC). FTTB and FTTC can work together with a VDSL2 link that reaches the customers' premises, creating thus the FTTB/VDSL2 and FTTC/VDSL2 access networks. Figure 4 shows the FTTx technologies. VoIP can start at the end-user premises, or at the remote IP-DSLAM.

Figure 4: Examples of FTTx technologies



Source: WIK

For the access there are different types of technologies: xDSL, FTTx technologies, cable and wireless networks, etc. As was mentioned in the previous section, the question of where VoIP really starts, i.e. at which precise node, is not relevant. All the network elements (nodes and links) should be included for the calculation of the cost of voice termination fees.

2.2.2 Aggregation network

The initial point of the aggregation network is the DSLAM. As transmission techniques Asynchronous Transfer Mode (ATM) or Ethernet can be used. ATM is a standardised packet-transmission oriented technology. It works with cells that are transmitted over a virtual link which was established before the transmission of cells start.

For the transmission of services such as voice and video ATM uses quality parameters such as processing delay, jitter, throughput and packet loss rate. A total delay of 15 ms is acceptable for the transmission of voice and video, but the delay oscillation can be critical. The ATM Adaptation Layer (AAL) is a layer used for the transmission of different types of services. In ATM the cells are numbered, and therefore it is possible to identify the cells that are lost during the transmission.

The Ethernet technologies are also used in the aggregation network and start to replace ATM. In comparison with ATM, Ethernet can support higher connection speeds, simpler provisioning, and multicast¹.

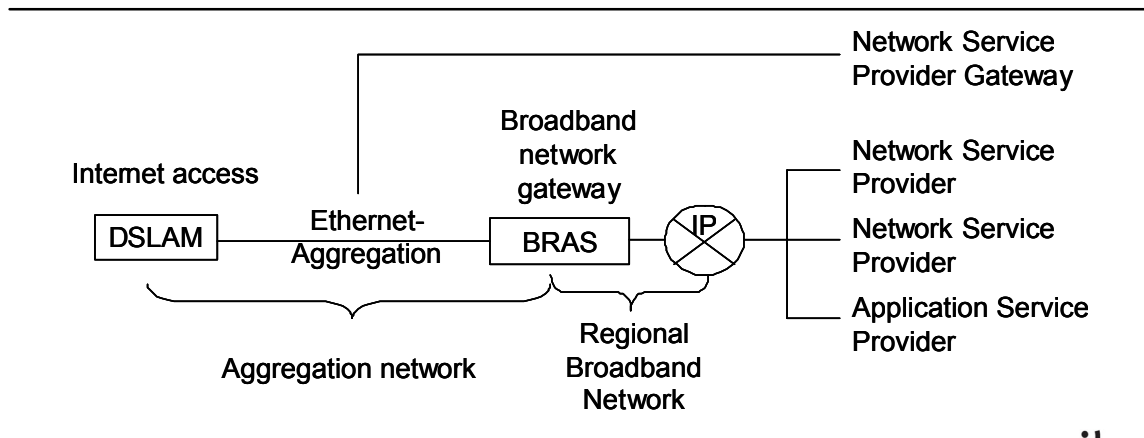
Ethernet is an OSI layer-2 protocol that in comparison with ATM enables the use of variable packets' size. For many years Ethernet was used basically in the Local Area Networks (LANs), but nowadays it tends to replace ATM as aggregation network technology. Figure 5 shows the network architecture of an Ethernet-based DSL aggregation network. The Broadband Remote Access Server (BRAS) is used in the regional broadband network.

Quality of Service can be provisioned in the Ethernet MAC layer by using a prioritization mechanism. The IEEE 802.1Q standard defines the services offered in Virtual Bridged LANs, and the corresponding protocols and algorithms. Even though precise Quality of Service mechanisms are not defined in the standard, the standard defines the possibility of prioritizing received frames according to the priority given to the frame. The information about the priority can be found in the User Priority Regeneration Table of the reception port and in the frame itself².

¹ Architecture and Transport Working Group (2006): Migration to Ethernet-Based DSL Aggregation, Technical Report – DSL Forum TR 101, p. 11.

² Ek, Niclas (1999), IEEE 802.1 P,Q - QoS on the MAC level, Helsinki University of Technology, Research Paper.

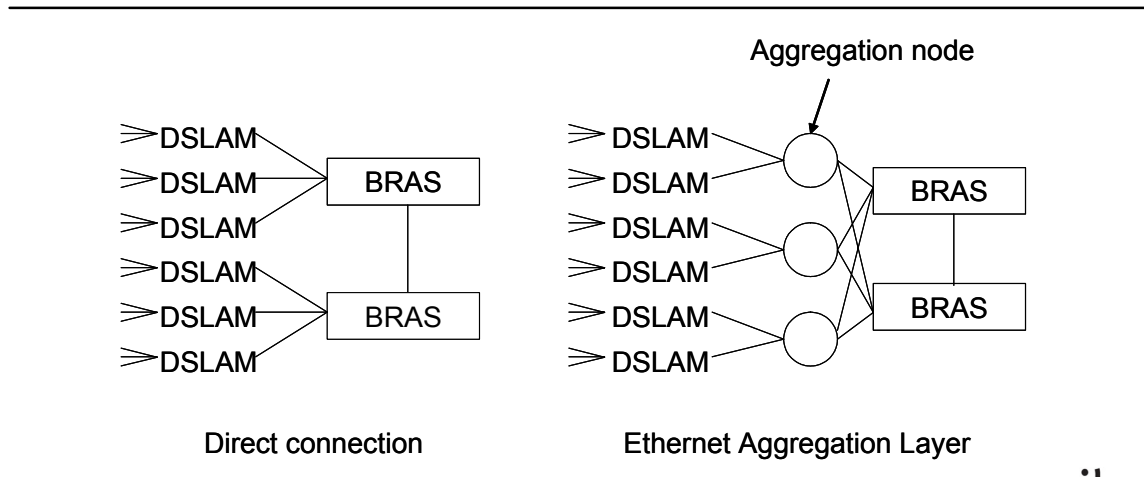
Figure 5: Network Architecture for Ethernet-based DSL aggregation



Source: Architecture and Transport Working Group (2006): Migration to Ethernet-Based DSL Aggregation, Technical Report – DSL Forum TR 101, p. 19

The Ethernet Aggregation layer is necessary for the aggregation of data traffic that is transmitted through the DSLAMs, for the traffic management, i.e. for the traffic prioritization, and also to guarantee “multicast deployments”. A characteristic of the Ethernet structure is that in the layer of the aggregation network, which is between the DSLAM and the BRAS, the DSLAMs can be directly connected to each other through the aggregation nodes. See Figure 6.

Figure 6: Alternative network architectures



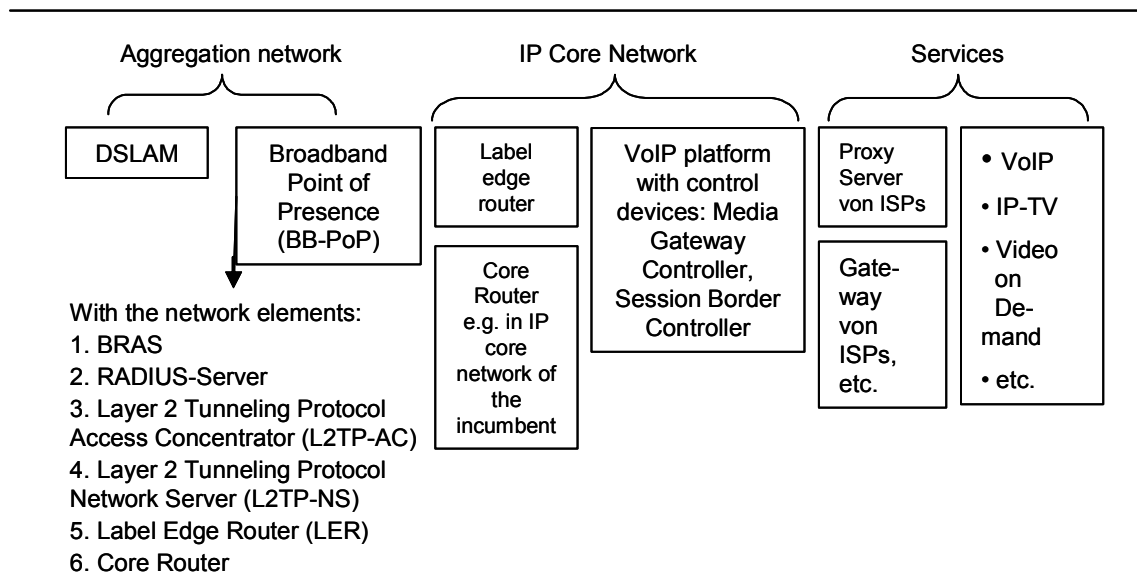
Source: Architecture and Transport Working Group (2006): Migration to Ethernet-Based DSL Aggregation, Technical Report – DSL Forum TR 101, p. 29

The network diagram on the left of Figure 6 shows that for a communication between two remote DSLAMs there can be a direct connection between two BRAS nodes. The network diagram on the right includes the aggregation nodes, and for a communication between two remote DSLAMs it is not necessary to go through the BRAS.

2.2.3 IP Core Network

At the BRAS or to be more precise at the Broadband Point of Presence (BB-PoP) the traffic data that arrives from the ATM-based or Ethernet-based aggregation network is transmitted to the IP Core Network (see Figure 7).

Figure 7: Network Elements at the aggregation and IP network planes



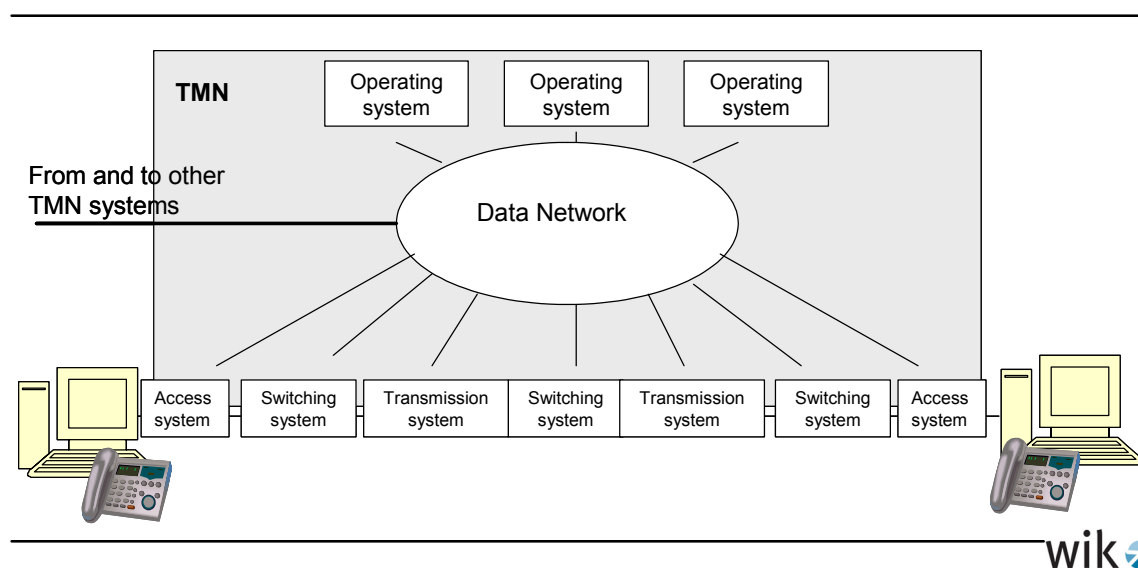
Source: WIK

The BB-PoP can contain the following elements. The BRAS is the bridge between the aggregation and the IP layer. It is necessary for the decoding of ATM traffic, and together with the Radius-Server it authenticates the consumers. The Layer-2 Tunnelling Protocol Access Concentrator (L2TP-AC) is necessary for the construction of tunnels to a L2TP network server. Label Edge Routers forward packets that come from the periphery of the IP Core network. The BRAS could have this function but a separation of network elements is better for security reasons. The core routers transmit high-speed data inside the operator's networks and also to other operators' networks.

2.2.4 Network and Traffic Management

A network operator can maintain the network with the help of a Telecommunications Management Network (TMN) system (see Figure 8). The TMN is necessary for the recollection of data about the traffic, for the communication between the network elements, and to manage the accounts of the customers.

Figure 8: Structure of a Telecommunications Management Network (TMN)



Source: Siegmund, Gerd (2002), p. 74

2.2.5 Quality of Service in IP networks

Real-time services like Voice over IP require strict levels of loss, latency and jitter delay. For the provisioning of Quality of service in IP networks the following techniques can be used:

- Prioritization of Data
- Reservation of transmission capacity
- Over-dimensioning of the transmission network

The prioritization of data can be done by means of the Differentiated Services (Diff-Serv) technique. The packets have information in the packet header about the type of service of the packets. This information is used by routers to give packets a corresponding priority at the moment of sending them to the network. There are three types of Services:

- Premium Services: The network operator guarantees a certain level of transmission quality.
- Assured Forwarding Services: For this type of service a minimum level of quality for the transmission of packets in the network is guaranteed.
- Best Effort Services: There is no guarantee for the transmission of packets in the network.

For the reservation of traffic capacity the following methods are used:

- Integrated Service (IntServ)
- Aggregation of traffic
- Multiprotocol Label Switching (MPLS)
- Virtual Private network (VPN)

Integrated Services usually works with the Resource Reservation Protocol (RSVP). Before a transmission takes place, an end-to-end path is established between the sender and the receiver and the necessary capacity is reserved.

The trend in industry is to use DiffServ instead of IntServ because IntServ has major problems of scalability in large networks.

The aggregation of traffic consists in that traffic with similar characteristics like video and voice is aggregated in the transmission plane and it is treated together by the intermediate nodes. Another procedure is MPLS. MPLS is a combination of routing and switching. With MPLS there is a classical routing to determine the best path through the network, and the packets are named with special labels to speed up their transmission at the data link layer. MPLS can also work with Virtual Private Networks. With the help of the labels private tunnels can be set up in the network. MPLS – Traffic Engineering (MPLS-TE) is a technique used to allocate sufficient capacity to a VPN. Traffic Engineering is employed to compute and select optimal paths such that the path is optimal according to some scalar metric and does not violate any constraints (bandwidth/administrative requirements)³.

By over-dimensioning it is considered that the operator will deploy more capacity in terms of additional nodes and links capacity.

³ Cisco (2002), Advance Topics in MPLS-TE Deployment, White Paper.

2.3 Conclusions

There are several requisites that should be fulfilled so that the end-user can work with IP-based services. If the end-user has direct access to the IP technology, then it can have an IP device. On the other hand, the cost of the telephony device is not included in the calculation of the cost of voice calls.

The access network could consist of fixed telephony networks, cable networks and wireless networks. In the aggregation network the ATM and Ethernet technologies are used. The IP core network consists of edge routers, core routers and VoIP devices. For network and traffic management a Telecommunications Management network is necessary. As the voice service requires strict levels of loss, delay and jitter, it is necessary to deploy quality of service mechanisms. There are three basic mechanisms: Prioritization, Capacity reservation, and over dimensioning. The provisioning of Quality of service in IP networks is a critical issue. However, there are several solutions to this problem. The network operator will decide whether it will implement QoS mechanisms or not. We believe it to be most efficient to use a prioritisation procedure like DiffServ⁴.

⁴ Jay, Stephan, Plückebaum, Thomas (2008), Strategien zur Realisierung von QoS in IP-Netzen, Wik-Consult Research Report for the Bundesnetzagentur, December, 2008, Germany.

3 VoIP techniques

During the last two decades VoIP devices have been used in private and corporate networks. As there are several types of VoIP technologies, this chapter explains the technologies needed to deploy the VoIP services. First of all, voice codes and protocols needed for an end-to-end transmission are described in Section 3.1. Section 3.2 explains the main VoIP Architectures: H.323, SIP, and Skype as P2P system. Section 3.3 touches aspects related to additional VoIP nodes and systems such as the softswitch and the IMS. Finally, Section 3.4 explains specific features of voice networks.

3.1 Voice codecs and transmission of voice information

For the transmission of voice over the IP network, first of all the analogue voice should be converted into IP datagrams. There are different types of codecs. The standard G.711 of the International Telecommunications Union (ITU) is one of the most used codecs⁵. The quality of the codecs can be measured by means of the Mean Opinion Score (MOS) procedure. A value of 1 corresponds to the worst quality, whereas a value of 5 corresponds to the best quality. The codec G.711 does not work with data compression techniques, whereas the codec G.729 employs data compression techniques. There are other codecs such as GSM, iLBC, iSAC and Speex (see Table 1 for a comparison of voice codecs in terms of bandwidth, MOS, delay, and quality). The iSAC and Speex codecs can obtain a very good level of MOS (MOS > 4.0) if more bandwidth is used.

Table 1: Voice codecs

Name	Necessary bandwidth in Kbps	Mean Opinion Score (MOS) value, between 0 and 5	Delay (ms)	Quality
G.711	64	4,40	0,125	Good
G.729	8	3,92	10	Good
GSM	4-21	3,80	20	Satisfactory
iLBC	14,4	4,00	30	Good
iSAC	30-60	> 4,00	33-63	Very Good
Speex	2,15-44,2	> 4,00	30-34	Very Good

Source: Schilder, Hans-Jörg (2002), p. 16

⁵ ITU-T (1993): Pulse Code Modulation (PCM) of Voice Frequencies, ITU-T Recommendation G.711

Transmission of voice information

The Real Time Protocol (RTP) that was developed by the IETF is used for the transmission of real-time data. RTP works with the UDP protocol. The Real Time Control Protocol (RTCP) provides feedback about the Quality of service that is provided by RTP.

The bandwidth necessary for the transmission of VoIP packets is higher than the bandwidth of the codecs that appears in Table 1, because the headers of the additional protocols (e.g. RTP and UDP) should be added. VoIP network designers suggest that 90-128 Kbps are necessary per user. This value should be taken into account at the moment of calculating real bandwidth demand.

3.2 VoIP architectures and protocols

There are several standardized network architectures and protocols that could be used to deploy VoIP services, such as the ITU-T H.323 system, the IETF Session Initiation Protocol, the Media Gateway Control Protocol (MGCP), and ITU-T H.248 MEGACO (Media Gateway Control Protocol) protocol. In addition, VoIP open standards such as IAX2, the Inter-Asterisk protocol that work with the Asterisk open source PBX server, are used in some VoIP deployments⁶. Besides these VoIP architectures and protocols Skype is a good and widely spread example of a proprietary VoIP system.

3.2.1 H.323 Architecture

H.323 was standardized by the ITU-T and it defines the protocols, procedures and components of devices that provide real-time audio, video and data communications. Four different components were defined in the H.323 architecture: terminals, gatekeepers, gateways and the Multipoint Control Unit (MCU).

The terminal can be a personal computer or an H.323 terminal. An H.323 terminal must interwork with other H.323 terminals. In this sense, it needs to have the following components:

- The H.245 protocol, necessary for the negotiation of channel usage and capabilities.
- Q.931 protocol for call setup and signalling.

⁶ Whereas SIP is an IETF official standard, the IAX2 (Inter-Asterisk protocol) 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. IAX2 is suited for private VoIP networks with low-budget limitations, and it is not used by major VoIP service providers.

- The Registration/Admission/Status (RAS) protocol for the communication with the gatekeeper.
- The RTP/RTCP protocol for audio and video packets.

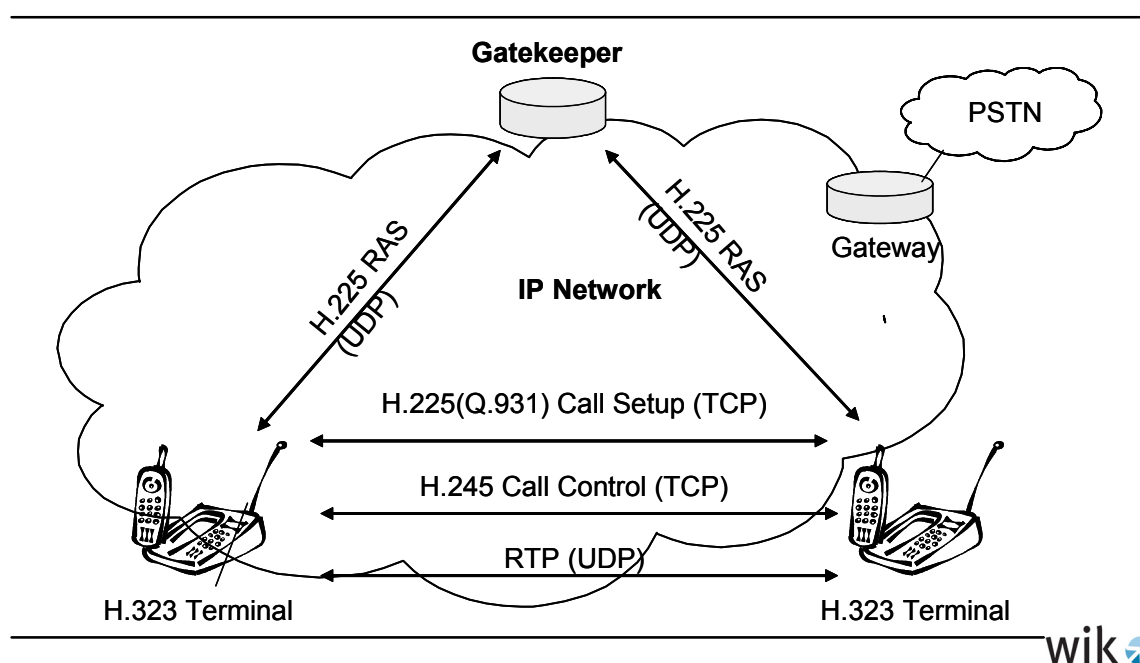
A gatekeeper is the central point of the calls within a zone and it offers call control services to registered H.323 endpoints. The following are the gatekeeper functions: address translation, bandwidth control, zone management, call-control signalling, call authorization, bandwidth management, and call management.

A gateway is used for the connectivity between H.323 and non-H.323 networks. The H.323 gateway uses the H.225 call signalling protocol for call setup and release, the H.245 control signalling protocol for exchanging capabilities, and the H.225 protocol for registration, admission, and status (RAS) with the gatekeeper.

The Multipoint Control Unit offers support for conferences of three or more H.323 terminals.

Figure 9 shows a basic scenario of an H.323 network. The H.323 terminals exchange VoIP packets directly by using the RTP/UDP protocols. The H.225 and H.245 protocols are used for controlling the call. Moreover, a Gatekeeper is needed to control the Gateways.

Figure 9: Basic nodes and protocols of an H.323 system



Source: Cisco (2006), Understanding H.323 Gatekeepers

3.2.2 SIP Architecture

SIP is an IETF VoIP protocol used for signalling and session management in a packet telephony network⁷. The definition of SIP appears in the RFC 3261⁸. SIP is used to establish, maintain, and terminate calls. SIP is a peer-to-peer protocol which is used by User Agent Clients (UACs) and User Agent Servers (UASs). A SIP end point can work as UAC and as UAS, but it only takes one role per transaction.

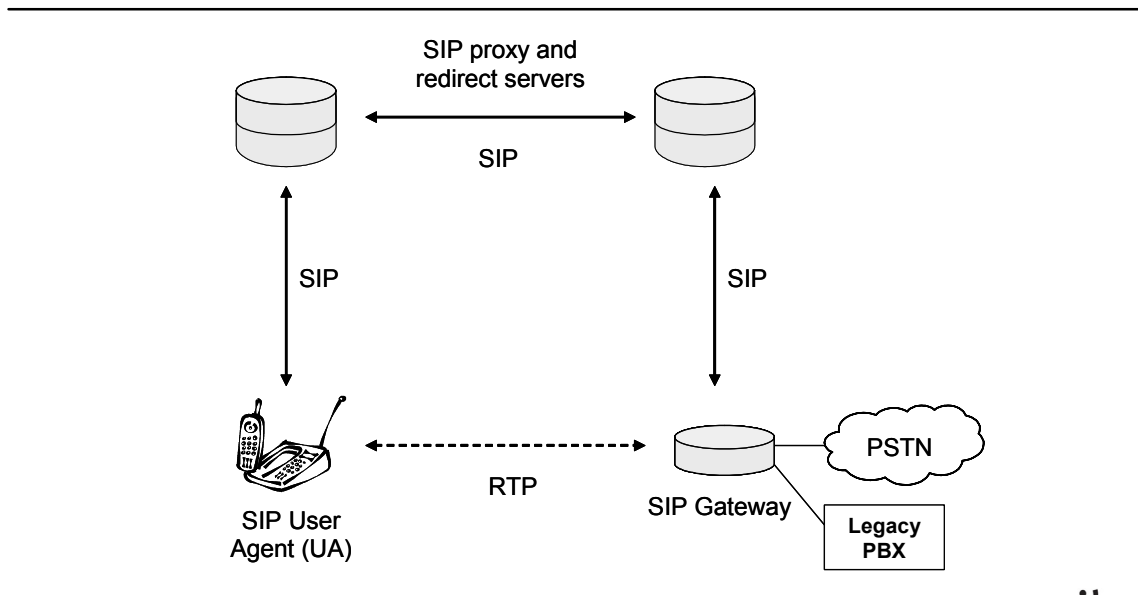
The SIP clients can be phones or gateways. The SIP phones are the end-user terminals, whereas the gateways are used for translation functions between SIP terminals and different terminal types. The SIP servers include proxy servers, redirect servers, and registrar servers. The proxy servers get SIP messages and send them to another SIP server in the network. They also are used for authentication, routing, reliable request retransmission, network access control, and security. The redirect server offers the client information about the next hop that can be taken by a message. The registrar server is used for registration.

In Figure 10 the basic SIP architecture is described. A SIP architecture consists of two basic elements: the SIP clients and the SIP servers. Figure 11 shows the location of the billing and voice mail servers in a SIP environment.

⁷ Cisco, Overview of the Session Initiation protocol.

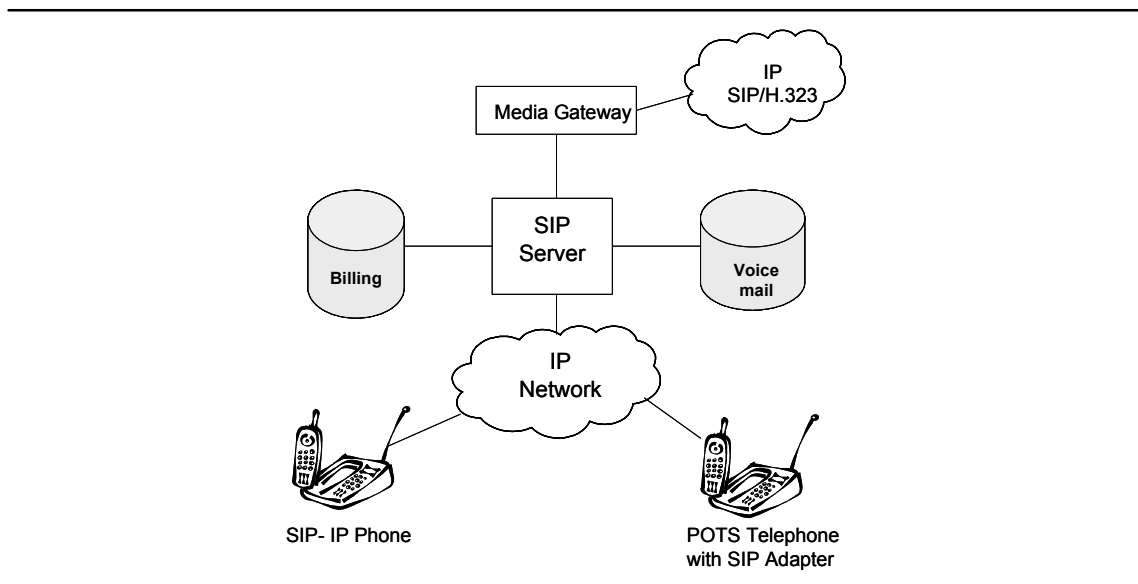
⁸ IETF, SIP: Session Initiation Protocol, RFC 3261, June 2002.

Figure 10: SIP Architecture



Source: Cisco, Overview of the Session Initiation Protocol

Figure 11: Basic architecture of a Sip Server provider



Source: Pulsewan, How to become a Voice over IP telephone company

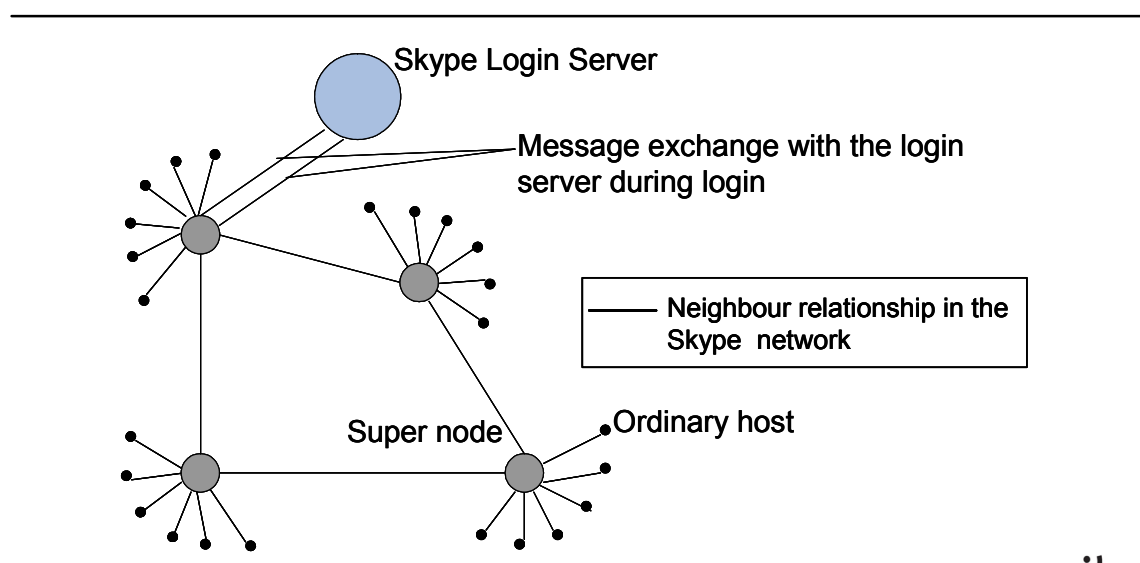
SIP uses the RTP/RTCP and the Session Description Protocol (SDP), which is necessary to negotiate the participant capabilities, codification type, etc. SIP follows an end-

to-end oriented signalling methodology, which means that the logic is stored in the SIP end-user's device.

3.2.3 The Skype Architecture

The company Skype works with proprietary software and its basic architecture contains three basic nodes: a Skype login server, a super node, and an ordinary host. The Skype login server is the only central component in the Skype network. Skype uses wideband codecs with a bandwidth of 32 kbps. It uses TCP for signalling and UDP and TCP for media traffic⁹. It is known that Skype uses the iLBC and iSAC codecs. For cost modelling the Skype login server is the relevant cost element that should be considered, since the other systems are not paid by Skype but are customer systems working as Skype nodes, paid by the customers.

Figure 12: A Skype peer-to-peer system



Source: Baset and Schulzrinne (2004), An Analysis of the Skype peer-to-peer Internet Telephony Protocol

3.2.4 More VoIP Architectures

Another VoIP signalling and call control protocol is the Media Gateway Control Protocol (MGCP), which was defined in the RFC 3435¹⁰. In the MGCP architecture there is a Media Gateway Controller, a Media Gateway, and a Signalling Gateway. Megaco, a

⁹ Arora, Prateek, VoIP: Skype architecture & complete call setup, seminar 2.

¹⁰ IETF, Media Gateway Control Protocol (MGCP), version 1.0, RFC 3435, January 2003.

signalling and call control protocol which is a co-production of the ITU (Recommendation H.248.1) and of the IETF (RFC 3525) can also be used.

As a proprietary system the Cisco Skinny Client Control Protocol (SCCP) can be mentioned. This protocol is a network terminal control protocol and it is used as a messaging system between the Cisco Call Manager and a terminal with the Skinny protocol, which can be a Cisco 7900 series IP phone¹¹.

There are several VoIP architectures and protocols and nowadays there is not a dominant VoIP technology. The VoIP market trends indicate that even though there is not a precise answer about which is the best VoIP technology, SIP is the most relevant and future-proof VoIP architecture. Therefore, SIP can be considered in LRIC based cost models.

3.3 Additional VoIP nodes and systems

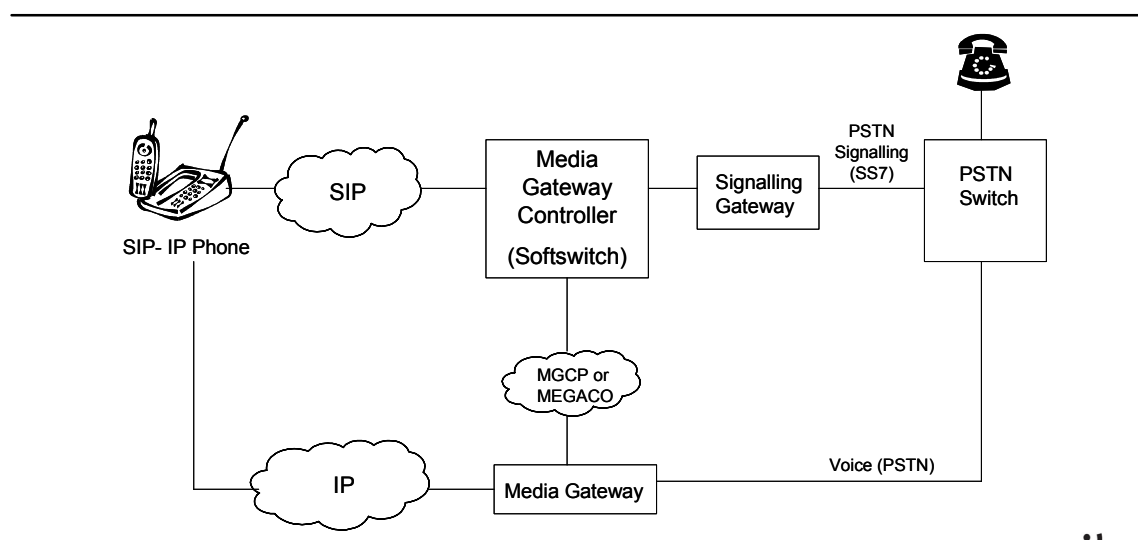
The softswitch can be used for the interconnection of a PSTN network with a VoIP network. The IMS (IP Multimedia Subsystem) is an architecture that helps to create and deliver services in an easy way.

3.3.1 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 (MG) and Signalling Gateways (SG). 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. Figure 13 shows the softswitch architecture. The softswitch can work with several signalling protocols, such as H.323, SIP, or MGCP. However, as SIP is being adopted by several network operators, the manufacturers tend to implement SIP in the softswitches.

¹¹ Cisco, Skinny Call Control Protocol.

Figure 13: The Softswitch Architecture



Source: Chang, Ming-Feng (2008), Course on Internet Telephony

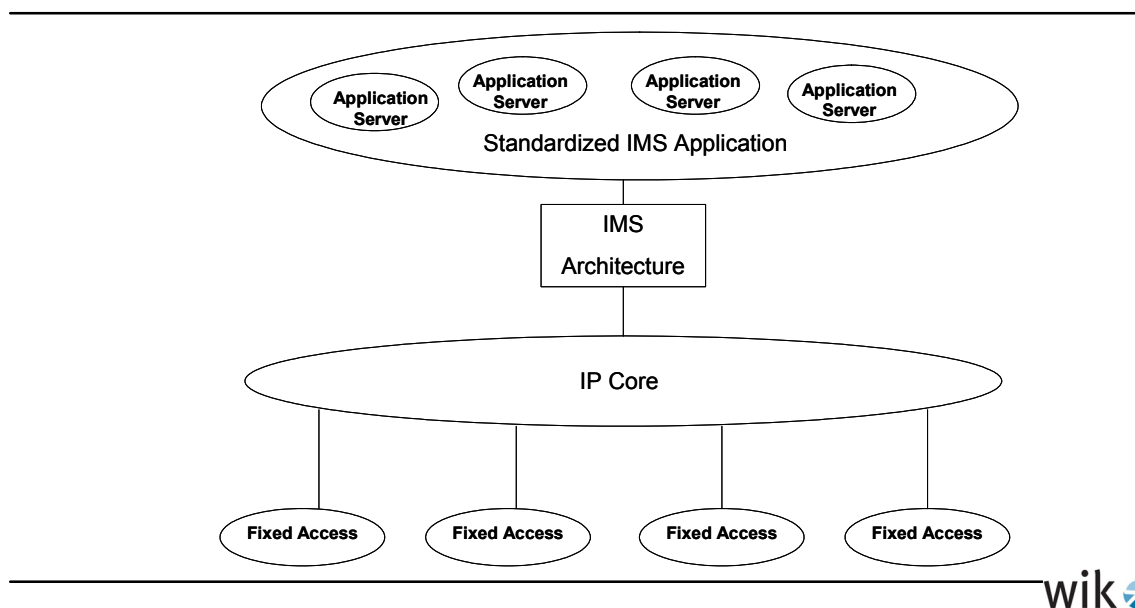
The softswitch technology separates service access from service control; at the same time it uses an IP-based core layer in the switching network¹². A discussion about the migration of PSTN to NGN with the softswitch is explained in Annex 1.

3.3.2 The IP Multimedia Subsystem (IMS)

The IP Multimedia Subsystem (IMS) is an architecture designed to offer 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. Figure 14 describes this concept.

¹² Shichang, Xiao (2007) IMS Soft Landing, The integration of softswitch and IMS, Huawei Technologies, Issue 31, June.

Figure 14: The horizontal architecture of IMS



Source: Ericsson White Paper, Introduction to IMS, March 2007

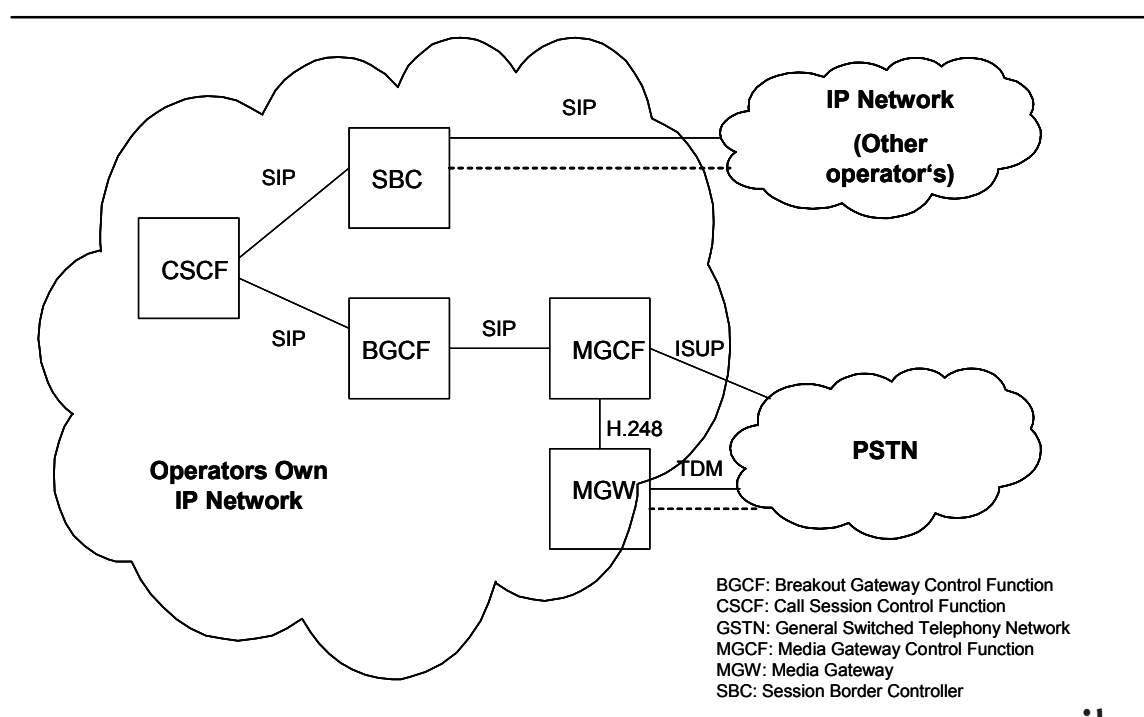
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The essential nodes of the IMS architecture are the Call Session Control Function (CSCF) and the Home Subscription Server (HSS). The CSCF node controls the signaling by using the SIP protocol. The HSS node contains the user profile database for authorization and authentication purposes¹³. The three roles of the CSCF node are the following: Serving-, Interrogating- and Pro-Call Session Control Function (S-, I- and P-CSCF).

For the interconnection with the General Switched Telephony Network (GSTN) – which can be the PSTN – two nodes are necessary: the Media Gateway Control Function (MGCF) and the Media Gateway (MGW). The MGCF is in charge of managing signaling information whereas the Media Gateway is controlled by the MGCF using H.248 and it translates RTP/UDP/IP packets into TDM signal streams. Figure 15 shows the nodes necessary to interconnect the IMS architecture to the GSTN architecture. The Breakout Gateway Control Function (BGCF) chooses the route of the Telephony session. The Session Border Controller (SBC) is an IP to IP gateway.

¹³ Ericsson White Paper (2007), Introduction to IMS, March 2007.

Figure 15: Interface IMS – IP Network/PSTN



Source: Ericsson White Paper, Introduction to IMS, March 2007

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 best alternative.

For cost modelling of current VoIP networks the softswitch can be considered as a relevant node. For the future, as it is expected that operators will offer multimedia or advanced services and fixed and mobile networks will converge, the IMS can be taken into account as the call control node.

3.3.3 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 layer¹⁴. The PSTN network will sooner or later migrate to a NGN network, and one of the open issues is how the migration is going to take place.

¹⁴ ITU (2004) International Telecommunications Union, NGN Working Definition: http://www.itu.int/ITU-T/studygroups/com13/ngn2004/working_definition.html.

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 the 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. Moreover, there are also savings when new equipment with more switching power is deployed and when high-bandwidth links between the switches are used (e.g. Gigabit Ethernet links instead of nxE1 interfaces).

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 infrastructures the copper cable can be kept in the very last end, whereas for other infrastructures it is not possible¹⁵.

3.4 Features of Voice Networks

3.4.1 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.

3.4.2 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 location¹⁶. In the PSTN world the Number Portability is provided by means of databases that belong to the Intelligent Networks of the operators. When it is detected that a destination number is not in the operator's own network but has been ported a query is submitted to the Number Portability Database (NPDB), which knows the current location/ network operator of the destination number.

¹⁵ Elixmann, Dieter, Ilic, Dragan, Neumann, Karl-Heinz, and Plückebaum, Thomas (2008), The Economics of Next Generation Access, Wik-Consult Study for ECTA, June.

¹⁶ Ivcek, Mario (2007), ENUM based Number Portability in VoIP and IMS Networks, Mipro conference, Croatia.

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. We believe this to be the most appropriate solution.

3.4.3 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”¹⁷. 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^{18,19}. If the VoIP user is non-nomadic and generates the emergency call from the address that was registered at the VoIP operator, then it is technically feasible to route properly the call to the next Public Safety Answering Point (PSAP). 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. Even though there are ongoing efforts to find a solution for this drawback, such as the IETF ECRIT Working Group²⁰, nowadays this problem remains an open issue and it will pass some time until a standard is accepted by the ISPs and VoIP providers.

One additional feature of emergency calls is that the operators have to transmit the location information from where the emergency call has been set up to the PSAP to enable it to direct the rescue forces immediately even when the victim does not know its location or address or is unable to speak anymore. In the case of a resident user calling from the address registered the transmission of the registered address will not cause problems, but in the case of a nomadic user this may cause problems. In mobile networks the base stations allow to locate a caller. The IP-networks still have to be upgraded to the ability to locate the access port a user is actually using and by combining that with geocoordinates or address information.

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

¹⁷ Directive 2002/21/EC of the European Parliament and Council (Framework Directive), 07.03.2002.

¹⁸ ERG Common Position on VoIP, ERG(07)56rev2, December, 2007.

¹⁹ Elixmann, Dieter, Marcus, Scott, Wernick, Christian (2008), The Regulation of Voice over IP (VoIP) in Europe, Wik-Consult Study for the European Commission.

²⁰ IETF ECRIT Working Group Webpage: <http://www.ietf.org/html.charters/ecrit-charter.html>.

3.4.4 Security issues

Two aspects are related to the provisioning of call information to authorized security forces: lawful interception and data retention. Lawful interception is the obligation of providing access to 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 for this effect. 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. Similar as for the emergency calls there is a need to localize the callers, now those of an intercepted access.

In the European Union the Directive 2006/24/EC of March 2006 on Data Retention requires member states to store the following information for a period of between 6 months and 2 years²¹²²:

- to trace and identify the source of a communication;
- to trace and identify the destination of a communication;
- to identify the date, time and duration of a communication;
- to identify the type of communication;
- to identify the communication device;
- 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 this type of signalling information in the softswitch. However, for nomadic users that are outside of the home network it is feasible to keep track of the current IP address but it would not be easy to identify the precise location of the nomadic users.

Regarding the cost of the provisioning of lawful interception and data retention to authorized security forces the German court has not decided yet whether the cost for this equipment will be refunded by the public authorities (as the operators claim) or not (as the German Telecommunication law constitutes. In other countries in Europe the governments make a refund for the necessary investment to offer these two services. If the costs are refunded they of course may not be taken into account for the calculation of termination fees.

²¹ Directive 2006/24/EC of the European Parliament and Council, Art. 6, 15 March, 2006.

²² The German TKG restricts this period to 6 months.

3.5 Conclusions

The main conclusions of this chapter are the following:

- *Different VoIP techniques:* There is not a widely accepted standard for VoIP networks. Unlike what occurs in PSTN networks where there are a limited number of standards (e.g., PCM64 and SS7) that are followed by most operators, in IP networks there is a wide range of VoIP systems and protocols. Depending on the business model of the operator a specific VoIP system will be used. The SIP softswitch is the actual state-of-the-art architecture and it can be considered an efficient reference model architecture.
- *The features of VoIP networks require the deployment of appropriate systems and storage space:* Cost elements that should be considered are number portability (ENUM), emergency calls, billing systems, customer care systems, lawful interception, and data retention.

4 Interconnection of VoIP networks

In this chapter different scenarios used to establish a Voice over IP connection are described. Section 4.1 describes the basic VoIP scenarios, whereas section 4.2 addresses VoIP scenarios in an NGN context. Even though it is not the purpose of this report to match the different VoIP scenarios with the markets recommended by the European Commission, it has to be mentioned that a VoIP operator may have to satisfy a certain set of conditions according to the market it belongs to²³²⁴. And these conditions will have an impact on the investment and operating costs of the VoIP provider.

4.1 VoIP basic scenarios

This Section includes a taxonomy of current VoIP scenarios.

4.1.1 VoIP calls between IP networks

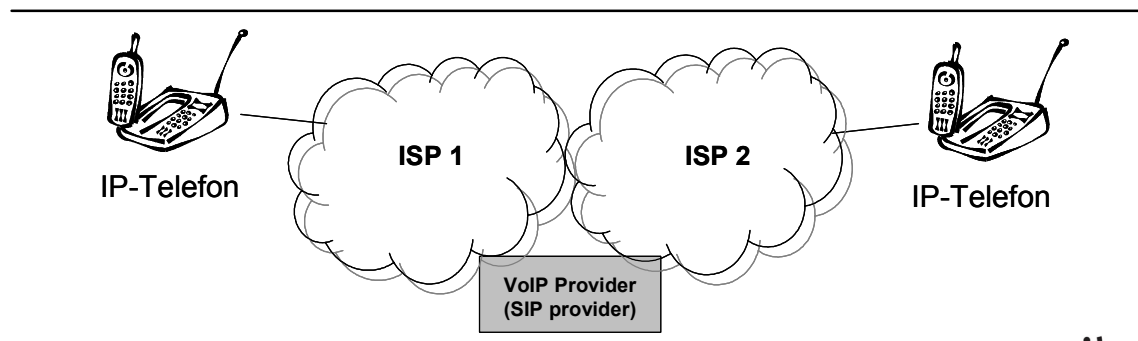
In this category, the end-to-end communication takes place exclusively over IP connections. There are no inbound nor outbound calls. The terminal can be an IP telephone, a softphone, or a POTS telephone that is connected to an ATA (Analogue Telephone Adapter) or to a VoIP card with A/D conversion capabilities.

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 of the end-to-end communication depends on the quality of service of the network nodes. Figure 16 shows how the end-to-end connection is achieved. It is not easy an easy task to guarantee interconnection with QoS parameters over network boundaries.

²³ For example the obligations that may be imposed on a VoIP provider such as Skype for its outbound service (SkypeOut) may not be the same as the ones imposed on Skype for its inbound service (SkypeIn).

²⁴ An explanation about the problematic of IP Interconnection appears in "J. Scott Marcus, D. Elixmann, The Future of IP Interconnection, Technical, Economic, and Public Policy Aspects, WIK-Consult Study for the European Commission, January 2008".

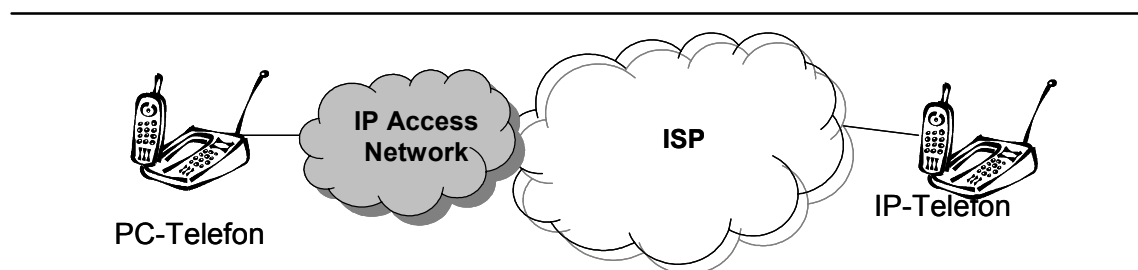
Figure 16: Scenario where the VoIP provider is not the owner of the access network



Source: WIK

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Figure 17: Scenario where the VoIP provider is the owner of the access network



Source: WIK

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Skype with its PC-to-PC option is one company that offers this type of connection. Furthermore, Skype uses a peer-to-peer network. This technology needs all the nodes to collaborate in the transmission of information. Some nodes assume the role of ordinary hosts, whereas others assume the role of supernodes. On the other hand, there are several equipment manufacturers that build equipment that uses several VoIP technologies: H.323, SIP, etc. Companies or organizations with limited budget can also use the IAX2 protocol.

This case refers to a VoIP provider, not to a private company that uses VoIP in several subsidiaries. Several VoIP providers that fall into this category specifically mention that their services can not be used for emergency service dialling.

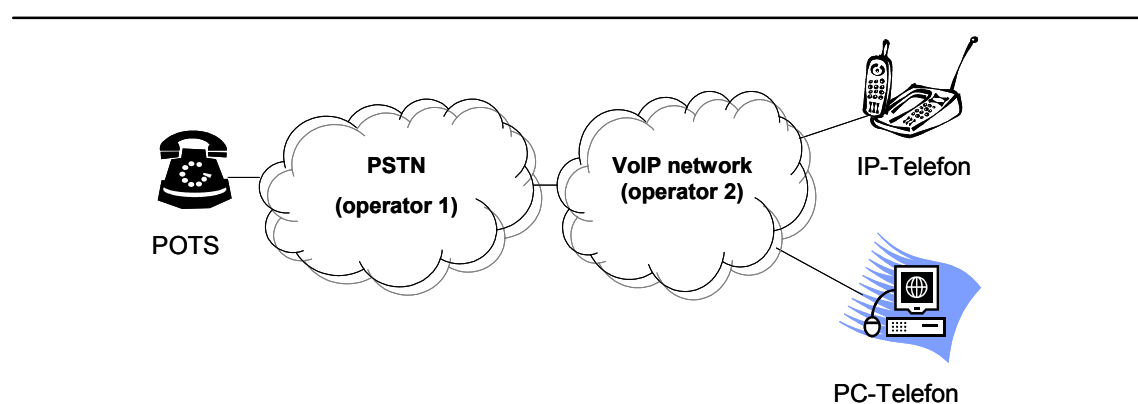
4.1.2 VoIP calls between an IP network and a PSTN network

This situation is much more complicated 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.

This case embraces the three cases defined by the European Regulators Group²⁵: Case 2 (outbound calls); Case 3 (Inbound calls); and Case 4 (Inbound & Outbound calls)²⁶. One question that arises when a PSTN operator and a VoIP operator exchange traffic is the definition of the type of interface that is used for the exchange of payload and signalling (SS7 or IP).

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 support E.164 numbers. The quality of the connection will strongly depend on the quality of the VoIP network. Figure 18 shows the depicted scenario. A media gateway that supports SS7 is necessary.

Figure 18: Calls between a PSTN and a VoIP network



Source: WIK

wik

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

²⁵ ERG Common Position on VoIP, ERG(07)56rev2, December, 2007.

²⁶ The specific services offered by the VoIP provider (inbound calls, outbound calls, location of the customer, etc.) could probably belong to the markets defined by the European Commission in 2007 (Official Journal of the European Union, L344/69, 28.12.2007).

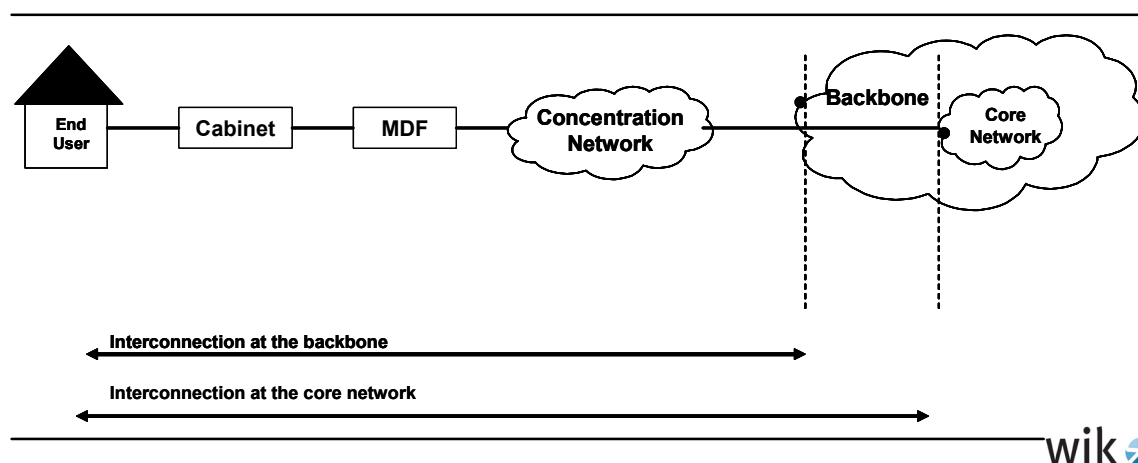
4.2 VoIP in an NGN context

There are three basic types of networks that can become Next Generation Networks: Fixed telephony networks (PLMN), cable networks, and mobile networks (PLMN). This section describes the three types of networks.

4.2.1 Fixed networks

Figure 19 shows the different interconnection points inside a fixed telephone network. There are two places where the interconnection can take place: on the outside section of the backbone and in the core network of the backbone. For the calculation of the termination rates the operators should take into account the cost of the elements that belong to the segment which is located between the end-user and the interconnection point.

Figure 19: Interconnection points in a fixed telephone network



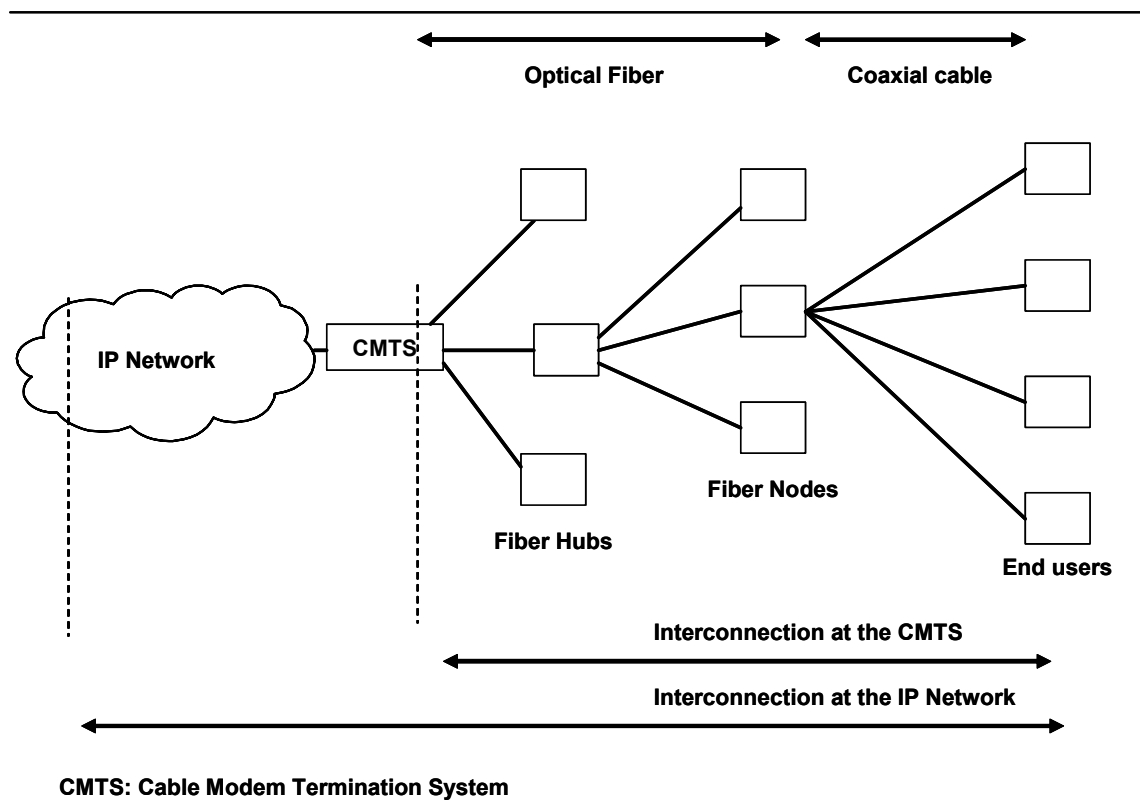
Source: WIK

A wholesale access service can be offered at the backbone and at the core network. Currently the access line is dedicated exclusively to the user until the MDF or to the street cabinet. With the deployment of FTTx technologies the user will have to share the line, and the available bandwidth, from a point that is closer to the end-user's premises. If FTTC is used, users share the infrastructure from the DSLAM located in the street cabinet. If FTTB is used the bandwidth is shared from the basement of the building. At the moment of calculating the cost of the voice call per user, this difference should be considered.

4.2.2 Cable networks

Cable networks have basically two main sections: the cable section and the fiber section. Figure 20 shows the two main types of interconnection that could take place in cable networks to transmit VoIP data. In the first case the interconnection takes place at the Cable Modem Termination System (CMTS), whereas in the second case the interconnection takes place at the IP network (e.g., if a bitstream service is offered).

Figure 20: Interconnection points in a cable network

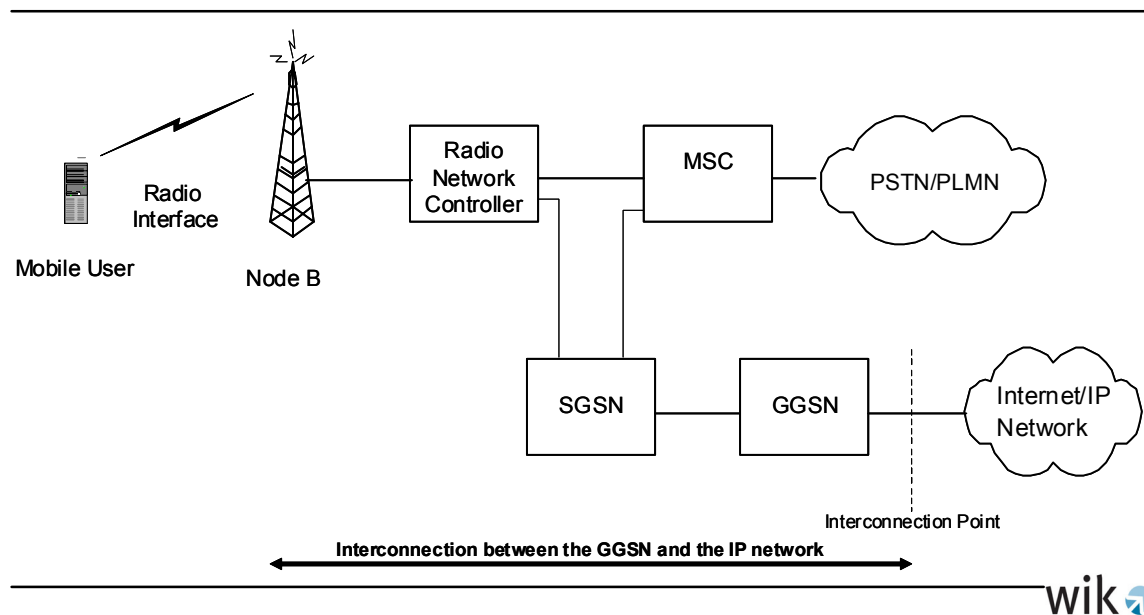


Source: WIK

4.2.3 Wireless networks

Figure 21 shows the possible interconnection point in a UMTS/HSDPA network. The interconnection point is located between the GGSN (Gateway GPRS Support Node) and the IP network.

Figure 21: Interconnection points in a UMTS wireless network

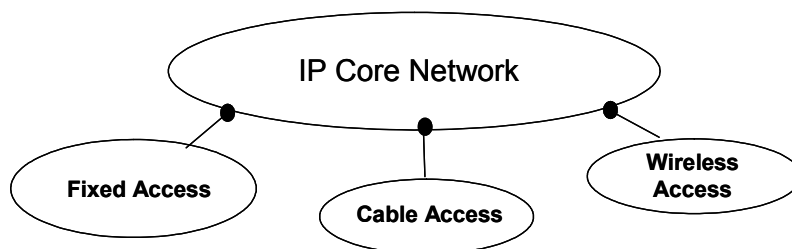


Source: WIK

4.2.4 Different interconnection points for different access networks

In the future, users will have an IP core network with different interconnection technologies. In the wireless networks field, this type of network is known as an heterogeneous network (see Figure 22).

Figure 22: Different interconnection points for different access networks



Source: WIK

wik

4.3 Conclusions

This chapter describes several scenarios about the realization of VoIP networks. The specific type of service offered by the VoIP provider will determine the market this service belongs to. And the investment and operating costs will change according to the obligations imposed on the VoIP provider. The main conclusion of this chapter is the following:

- *Different interconnection points for each type of NGN network:* There are three basic types of NGN networks: fixed networks (PSTN or NGN), cable networks, and wireless networks. The location of the possible interconnection points will have an impact on the cost of the termination rate.

5 Particularities of costs considerations of VoIP networks

This chapter discusses a few particularities that affect the determination of costs of VoIP networks. Section 5.1 explains the general considerations of the cost of the deployment of a VoIP network, whereas Section 5.2 addresses Quality of service issues. A few interconnection cases are described in Section 5.3. Finally, Section 5.4 addresses the implications on costs of the border between the access network and the voice switching network.

5.1 Costs consideration for deploying a VoIP network

The items that appear below should be considered at the moment of calculating the costs of a VoIP network. Annex 2 contains a basic questionnaire with the main points that should be considered at the moment of determining the cost 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, caller identification, call waiting, call forwarding, three-party conference, etc. If the bandwidth available is large enough, the video telephony 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, or at least to the voice service.

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 cost²⁷.

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 Access 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 the street cabinet (FTTC),

²⁷ Jay, Stephan, Anell, Patrick, Plückebaum, Thomas (2008), Netzzugang im NGN-Core, WIK study for the Bundesnetzagentur, Germany, June.

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:

- **Core Network:** The core network will consist of nodes connected through high speed links with fiber optic technology.
- **Access Network:** For the calculation of the termination charges the access network now should also be taken into account to a certain extent. The access network could use several technologies: copper cable or fiber optic. In the future 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.

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

Customer care systems: The personnel and equipment necessary to serve the clients should be considered.

Network operation systems: The equipment and personnel necessary to implement Operation and Management features must be taken into account.

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.

5.2 Quality of Service

It will take some time until VoIP catches up with the quality of voice of current PSTN networks. If the VoIP provider is the owner of the IP network, then it can manage the quality of service of VoIP inside its own network. In this sense, there are some issues that should be addressed:

- What is the cost of a service that currently is not 100% reliable? How much time will pass and which approach should be taken until VoIP is available and functional for 99.999% of the time (the PSTN's five nines)?

- What is the cost caused by high-quality voice service that uses QoS mechanisms?
- What is the cost of each one of these QoS mechanisms (prioritization, reservation, over-dimensioning)?
- How should the interconnection of two operators with QoS parameters be defined?

5.3 Interconnection between two operators

This section describes the types of interconnection agreements of two operators: interconnection agreements and the case of two networks that run in parallel.

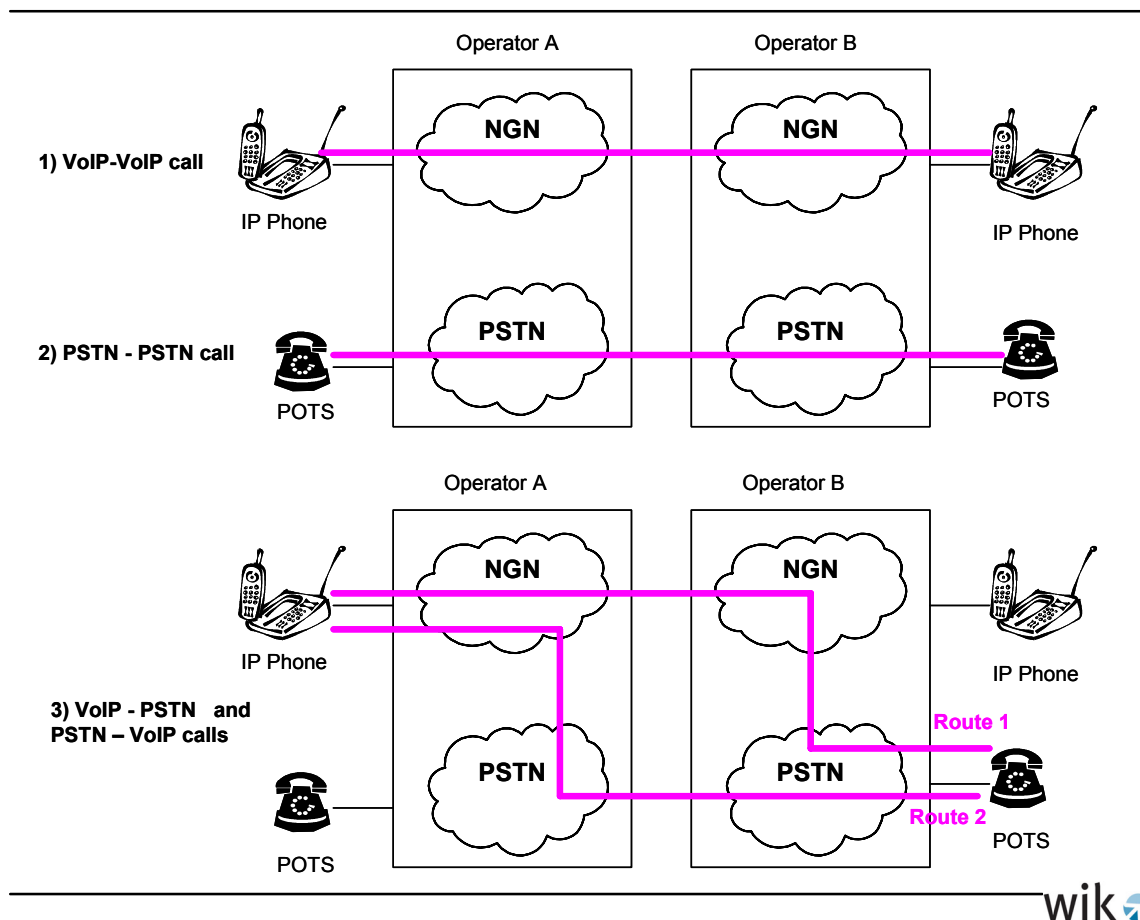
5.3.1 Two networks that run in parallel: PSTN and NGN

For a VoIP call between two NGN networks it makes sense to consider the cost of the interconnection between the NGN networks (there is only “one” path). As shown in Figure 23, there are different possibilities to route a call between two operators that own PSTN and NGN networks. It should then be determined whether the cost of the call corresponds to the actual path used for the call or to an average of the costs of all the paths that can be taken. Furthermore, the most efficient future-proof architecture should be considered.

When two operators exchange voice traffic, it should be clear which type of interface will be used for the payload and the signalling (SS7 or IP)²⁸. In mixed environments, where one operator is still using SS7, the other operator will have to deploy an appropriate Gateway, since SS7 is the history of voice networking. This view will change over time when the majority of networks is using IP instead of SS7.

²⁸ For example, many incumbents have IP and PSTN networks. Even though a telephony operator could be using IP internally to route the calls, in several cases it prefers to have the interconnection with other operators through SS7 interfaces.

Figure 23: Interconnection cases in the migration phase



Source: DTAG Presentation, „Diskussion zu den Auswirkungen des Technologiewandels auf die Mehrwertdienste, NGN-Interconnection“, Bonn, 14.07.2008. Published in the Bundesnetzagentur Amtsblatt 17/2008, p.2484-2499, 10.09.08

5.3.2 Interconnection agreements: Calling Party Pays and Bill & Keep

Usually in fixed and mobile networks the principle Calling Party Pays (CPP) is used to charge users for the calls. For the interconnection between Internet operators the principle used for charging is Bill & Keep. With VoIP the question that arises is the type of principle that will be used: Calling Party Pays, Bill & Keep, or a mixture of both²⁹. If the VoIP calls follow the CPP principle, then the operator of the caller will be the only one which will charge the caller. If the VoIP calls are charged according to the Bill & Keep principle, then the VoIP provider will charge the user for the originating and terminating calls. This is still a point with no precise answer and the answer will come from the interconnection agreements between the operators and from the decision taken by the

²⁹ Plückerbaum, Thomas, Abrechnen – aber wie? Mit VoIP werden gewohnte Abrechnungsverfahren in Frage gestellt, NET, Zeitschrift für Kommunikationsmanagement, 2007, NET 6.

regulatory authorities. CPP is relevant for cost modelling. That may change, but the approach now is to use CPP.

5.4 Common use of the access network: The border between the access network and the voice switching network

In the past the voice switching network which had been considered for the calculation of termination charges started at the locations where the traffic of different users had been combined and multiplexed on commonly used network elements, where they rival amongst each other for the common capacity. This normally happened in the MDF or local exchange locations, where the single (copper) access lines had been multiplexed to E1, E3 or STM-1 concentrators to transmit them to the PSTN switches. The other network elements had been part of the access network.

While the cost of the voice network had been shared by all users according to everybody's usage of the network, by calculating the cost per usage (per second or minute) and accounting the appropriate amount, the cost for the network access, the access line fee, had been calculated as a monthly fee, since it was provided for the voice service of a single end user only and could not be shared with others or by other services.

The ports of the PSTN network concentrators can be seen as endpoints of the PSTN core signalling network, between the concentrator (switch) ports and the telephones there is a terminal signalling. Within a NGN/NGA architecture the terminal signalling is restricted to the connection between the CPE/ CPE router and the telephones in the customer premises, if at all. Between the CPE and the softswitches/ IMS the core network SIP signalling is used. In addition to this criterion the access line in an NGA network is commonly used between different services, e.g. voice, business data, internet, IP-TV and video. Depending on the type of access network (e.g. Fibre to the Building, FTTB) several customers in the same building use the same access line with all the services they individually use, rivaling for the common bandwidth.

While on the one hand the border of the switched and commonly used network is migrating into the customer premises, the access and the core network are shared between several services, only one being telephony.

Thus, if a NGA is deployed, one has to consider the cost of the access network to some extent as well, but also has to determine the appropriate share of voice and the other services in the access network in order to allocate the cost appropriately. For this reason, a questionnaire for the access network should be prepared as well. This will be an area of future research.

5.5 Conclusions

This chapter has explained the following four aspects:

- *Quality of Service:* The quality of the voice provided by VoIP service providers is in many cases variable. To satisfy the strict time delay requirements required by VoIP connections the VoIP provider could deploy routers or switches that support QoS mechanisms, or it could expand the capacity of the systems or links. There are open issues such as the interconnection of VoIP providers with different QoS mechanisms.
- *Interconnection between two operators:* Two aspects should be taken into account. First, for the case of NGN networks it is not clear whether the operators will exchange signalling traffic through the IP or the SS7 interface. Second, the type of interconnection agreement between two operators could be Calling Party Pays, Bill & Keep, or a mixture of both procedures.
- *Common use of the access network:* If a NGA is deployed, one has to consider the cost of the access network to some extent as well, but also has to determine the appropriate share of voice and the other services in the access network in order to allocate the cost appropriately.

6 Final Conclusions

This report explains the main cost elements that should be considered at the moment of calculating the costs of voice calls of a VoIP provider. Chapter 2 explains how IP networks work and the components of an IP network. Chapter 3 describes the different VoIP techniques. In chapter 4 different interconnection scenarios of VoIP operators are explained. Chapter 5 addresses specific features of VoIP networks.

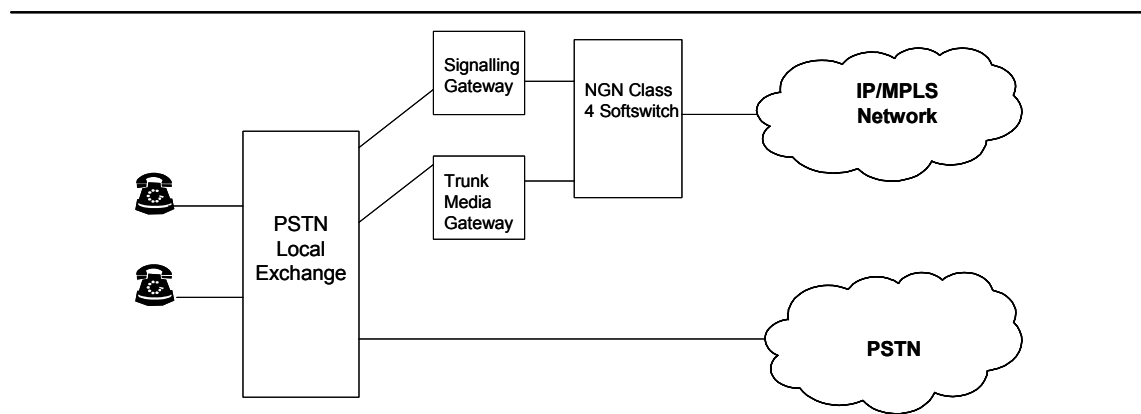
The main conclusions of the report are the following:

- *Different VoIP techniques:* Unlike what occurs in PSTN networks where there are a limited number of standards (e.g., PCM64 and SS7) that are followed by most operators, in IP networks there is a wide range of VoIP systems and protocols. Depending on the business model of the operator a specific VoIP system will be used. The SIP softswitch is the actual state-of-the-art architecture and it can be considered as efficient reference model architecture.
- *The features of VoIP networks require the deployment of appropriate systems and storage space:* Cost elements that should be considered are number portability (ENUM), emergency calls, billing systems, customer care systems, lawful interception, and data retention.
- *Different interconnection points for each type of NGN network:* There are three basic types of NGN networks: fixed networks (PSTN or NGN), cable networks, and wireless networks. The location of the possible interconnection points will have an impact on the cost of the termination rate.
- *Quality of Service:* The quality of the voice provided by VoIP service providers is in many cases variable. To satisfy the strict time delay requirements required by VoIP connections the VoIP provider could deploy routers or switches that support QoS mechanisms, or it could expand the capacity of the systems or links. The deployment of QoS mechanism has an impact of the cost of VoIP network elements.
- *Interconnection between two operators:* Two aspects should be taken into account. First, for the case of NGN networks it is not clear whether the operators will exchange signalling traffic through the IP or the SS7 interface. Second, the type of interconnection agreement between two operators could be Calling Party Pays, Bill & Keep, or a mixture of both procedures.
- *Common use of the access network:* If a NGA is deployed, one has to consider the cost of the access network to some extent as well, but also has to determine the appropriate share of voice and the other services in the access network in order to allocate the cost appropriately.

Annex 1: Softswitch for the migration from PSTN to NGN

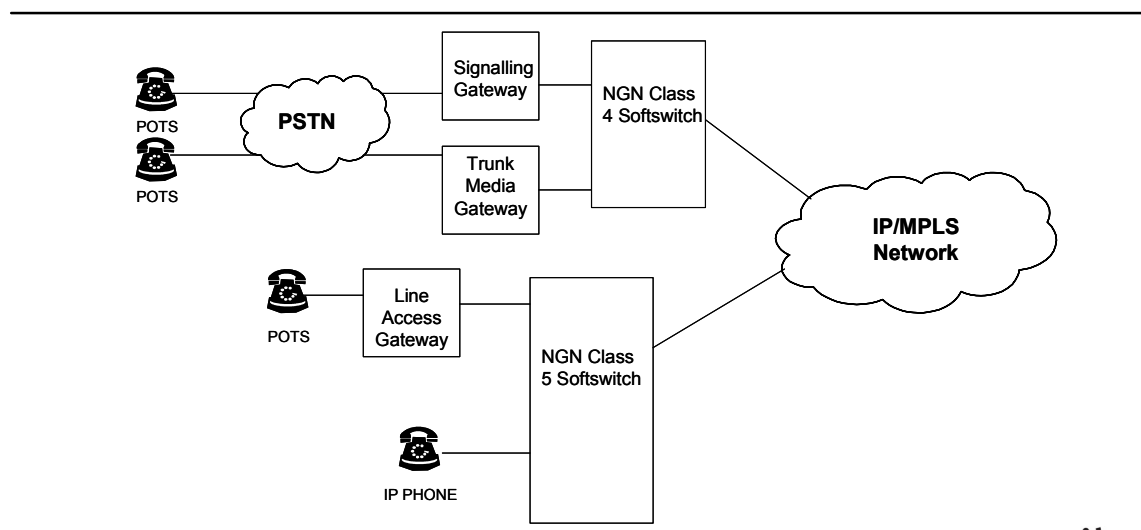
It is clear that the softswitch is a necessary element to interconnect legacy PSTN networks with IP networks. The softswitch can replace the Class 4 TDM and Class 5 TDM switches (see Figures A1.1 and A1.2). Furthermore, the IMS architecture is helpful to offer different types of applications. Therefore, it is up to the operator to decide whether the two platforms (softswitch/IMS) will coexist and for how long.

Figure A1.1: Migration of Trunk Exchanges (class 4)



Source: Migration of PSTN to NGN , Release 1, available at <http://www.tec.gov.in>

Figure A1.2: Migration of Local Exchanges (class 5)



Source: Migration of PSTN to NGN, Release 1, available at <http://www.tec.gov.in>

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