VoIP network architectures and impacts on costing

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Abstract

Purpose – *This paper aims to describe the effect of VoIP network architectures on the cost modelling of termination rates of VoIP services.*

Design/methodology/approach – *The study investigates and organises the arguments available in the technical and regulatory field related to VoIP networks and services in order to ascertain the possible impact of VoIP techniques, the provisioning of voice features in VoIP networks, and network interconnection issues on the cost of regulated VoIP services.*

Findings – *The information and analysis reveals how the provision of VoIP services is related to a number of issues that will have an effect on the cost of VoIP termination rates. In particular, the study analyses the impact on a cost model of the components of a VoIP network architecture, the usage factor of network elements, and the traffic volume generated by VoIP applications.*

Research limitations/implications – *The issues described in the article can be used in elaborating a cost model for termination rates in VoIP networks. For the present study, no cost model was built, and therefore no quantitative estimations were made of the specific impact of every cost parameter on the termination rates.*

Practical implications – *The findings of this study can be used by policy makers, voice operators, and researchers.*

Originality/value – *Most studies of VoIP that are available in the literature address, on the one hand, the costs of corporate VoIP networks and, on the other, the regulation of VoIP services. This article, however, presents a comprehensive study of the most relevant features of VoIP network architectures that should be considered when determining regulated termination rates.*

Keywords *Telecommunication networks, Internet, Mobile communication systems, Cost estimates*

Paper type *Research paper*

1. Introduction

The use of voice over internet protocol (VoIP) services has seen important growth over recent years[1]. People have become more aware of the benefits that VoIP offers in terms of cost and ubiquity, and many corporate and residential users are familiar with this technology. For example, as of December 2007, VoIP-originated calls in the European Union represented 8 percent of the traffic in the fixed sector (Commission of the European Communities, 2009). In The Netherlands and France, this rate was 32 percent and 27 percent, respectively[2]. For voice operators, the provisioning of VoIP services can be commercially, economically and technically beneficial. In an environment of convergence of telecommunications networks, data, voice and video signals are transmitted over the same physical link. Network operators can then save costs by employing only one network for different services. For instance, there can be CAPEX savings by using high-speed Gigabit Ethernet interfaces of 10 Gbps, instead of the E1 links of 2 Mbps. Moreover, there can be OPEX savings when using the well-known IP network, which will enable the management of only one network instead of the management of separate voice, video and data networks. VoIP, as a consequence, has been or will be implemented by current and

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prospective telephony operators. In this sense, several circuit-switched telephony operators have begun plans to migrate to VoIP. In many of these migration plans, VoIP is used initially in the core network due to the high investment associated with the deployment of broadband in the access network, which is one of the main prerequisites for the provisioning of VoIP services. Furthermore, for a number of new voice operators, VoIP is the underlying technology[3].

In many countries, the fixed and mobile telephony operators follow a calling party's network pays (CPNP) rule, which obliges the network operator that initiates the call to pay a termination fee to the network operators where the call is terminated. Competition and regulatory authorities consider that there is a form of market power, which is named a ''termination monopoly'', because a call can only be terminated by the service provider that controls the addressed telephone number. By regulating the termination fee, regulatory bodies have a tool to limit the inadequate use of this market power. Therefore, several National Regulatory Agencies (NRAs), government bodies and telephony operators are interested in determining the cost of the termination fee. It is foreseeable that the termination monopoly will continue in VoIP networks, and in any type of broadband network, such as the Next Generation Network (NGN), where the VoIP service is provided (Marcus and Elixmann, 2008). In this way, the termination fees will remain for a still longer period and, as a consequence, telecommunications regulators and voice operators will have to determine the cost of such rates in VoIP networks[4].

One of the first steps when calculating termination rates is the definition of the network architecture that will be used in the cost model. Cost modellers have wide experience in the cost elements that should be taken into account in legacy networks, such as fixed and mobile circuit-switched telephony networks. Nevertheless, VoIP is a disruptive technology and there are still several uncertainties about the cost elements typical for a VoIP network. A few authors have already described the cost elements of corporate VoIP networks (Hersent *et al.*, 2005; Kaza and Asadullah, 2005). However, a VoIP network operator that receives a license of public telephony services has to meet different requirements that will have an impact on the cost of the service, such as calls to emergency services, lawful interception, data retention or quality of the call. So far, the regulatory implications of VoIP have been described, but not from a cost modelling perspective (Elixmann *et al.*, 2008; Graham and Ure, 2005; Meisel and Needles, 2005). As there is a dearth of information on this subject in the existing literature, this article describes the impacts on costing of VoIP network architectures that provide public voice services. The information and the analysis provided in this article can be of interest to policy makers, telephony operators and researchers.

To carry out the analysis, the article contains a discussion of the importance of different aspects of VoIP network architectures. In several countries, Long-run incremental cost (LRIC) models based on an efficient network are used for the assessment of termination fees[5]. For this reason, it is necessary to identify which VoIP network architecture can be used as an efficient network. Section 2 explains the most relevant VoIP techniques: VoIP architectures and protocols, nodes and systems for the provisioning of VoIP services, the importance of VoIP traffic in shared channels, and quality of service (QoS) in VoIP networks. The purpose of this section is not to identify the most suitable VoIP technique, but rather to describe the technological options that are pondered by network operators when designing a VoIP network. Section describes the implementation of the following features in VoIP networks and their impact on cost modelling: provisioning of telephony features (e.g. voice mail and caller identification), number translation, number portability, access to emergency services, security issues and additional systems such as the network monitoring system. Section 4 deals with the following two issues related to the interconnection of VoIP networks: the interconnection interface, which can be a circuit-switched network interface or an IP-based interface, and the cost implications of the location of the point of interconnection. Finally, section 5 discusses the conclusions.

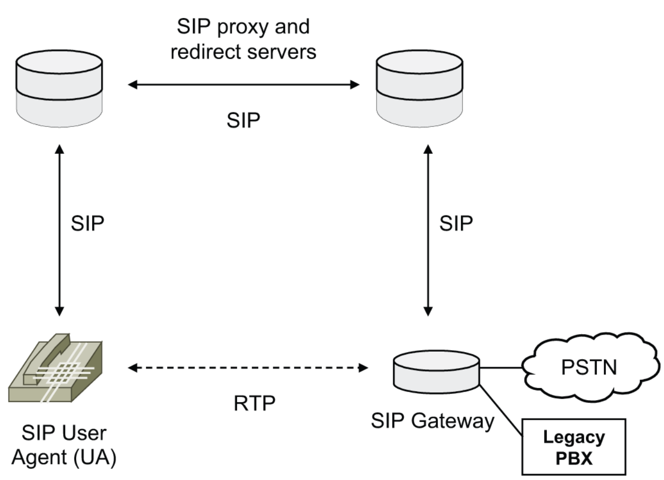
2. VoIP techniques

Unlike in the circuit-switched telephony world, where the technology used for the transmission and interconnection of telephony services was defined many years ago and is implemented on a national basis, VoIP operators have several possibilities at the moment of choosing the VoIP technique that will be implemented in their networks. So far, there is not a widely accepted standard for the provisioning of VoIP services. Standardization bodies such as the International Telecommunications Union (ITU) and the Internet Engineering Task Force (IETF) have defined VoIP architectures and protocols that have been deployed during the last two decades. A specific VoIP system will be used, depending on the business strategy of the voice operator. The importance of the definition of the VoIP technology that is implemented lies in the fact that LRIC cost models are based on efficient network architectures. This chapter describes a few of the most well-known technologies used by VoIP operators for the provisioning of voice services, as well as the relevance of VoIP traffic and quality of service in cost modelling of VoIP services.

2.1 VoIP architectures and protocols

The most relevant standardized VoIP network architectures and protocols are explained below, as well as a few proprietary VoIP systems[6]. The Session Initiation Protocol (SIP) was defined by the IETF for signalling and session management purposes (Internet Engineering Task Force, 2002). SIP is a peer-to-peer protocol that can be used to establish, maintain, and terminate calls. Figure 1 shows the basic SIP architecture. In the SIP architecture, a SIP endpoint can take the role of a User Agent Client (UAC), which is called a SIP client, or of a User Agent Server (UAS), which is called a SIP server. A SIP client can be a phone or a gateway: a SIP phone is an end user's terminal, whereas a gateway is used for translation functions between SIP terminals and different terminal types and for the interconnection with circuit-switched devices. Legacy PBXs and Public Switched Telephone Network (PSTN) switches can be connected to the SIP gateway. Proxy servers, redirect servers and registrar servers are SIP servers. A proxy server receives SIP messages and forwards them to another SIP server in the network. Proxy servers can also be used for authentication, routing, reliable request retransmission, network access control, and security. A redirect server provides the client with information about the next hop that can be taken by a message, and the registrar server is used for registration. SIP uses the Real Time Protocol (RTP) and Real Time Control Protocol (RTCP) for streaming media transmission, and the Session Description Protocol (SDP) to negotiate the participant capabilities, codification types, etc. SIP works with an end-to-end-oriented signalling methodology, which entails that the logic of the

Figure 1 The SIP architecture



Source: Cisco (2009b)

communication is stored in the SIP end user's device. Network operators have implemented different versions of SIP.

The ITU-T standardized the H.323 architecture, which defines the protocols, procedures and components of devices for the provisioning of real-time audio, video and data communications (ITU-T, 2006b). The main elements of the H.323 architecture are the terminals, the gatekeepers, the gateways and the Multipoint Control Unit (MCU). An H.323 terminal requires the following components for interworking with other H.323 terminals: the H.245 protocol for the negotiation of channel usage and capabilities; the Q.931 protocol for call setup and signalling; the Registration/Admission/Status (RAS) protocol for communication with the gatekeeper; and RTP/RTCP for the delivery of audio and video packets. Within an H.323 zone, a gatekeeper is the central point of the call and is used to provide registered H.323 endpoints with call control services. The gatekeeper functions include address translation, bandwidth control and management, zone management, call-control signalling, and call authorization and management. The gatekeeper is needed to control the gateways, which are utilized for the interconnection of H.323 and non-H.323 networks. The MCU enables the provisioning of conferences of three or more H.323 terminals. In the H.323 architecture, the H.323 terminals exchange VoIP packets directly by using RTP and the User Datagram Protocol (UDP), whereas the H.225 and H.245 protocols are employed for controlling the call.

The Media Gateway Control Protocol (MGCP) is a VoIP signalling and call control protocol that was defined by the IETF (Internet Engineering Task Force, 2003b). The components of the MGCP architecture are the Media Gateway Controller (MGC), the Media Gateway (MG), and the Signalling Gateway (SG). Megaco is another call control and signalling protocol that resulted from the cooperation between the ITU (ITU-T, 2006a) and the IETF (Internet Engineering Task Force, 2003a).

Several VoIP manufacturers have implemented their own VoIP solutions based on proprietary network architectures and protocols. For example, the basic VoIP network architecture of the company Skype contains three basic nodes:

1. a Skype login server;
2. a super node; and
3. an ordinary host.

In the peer-to-peer Skype network, the Skype login server is the only central component (Baset and Schulzrinne, 2006). For cost modelling purposes, the Skype login server is the relevant cost element to be considered, since the super node and the ordinary host are not paid by the Skype company, but by the customers.

Another example of a proprietary protocol is the Skinny Call Control Protocol (SCCP), which is used by Cisco in its VoIP Call Manager solution. SCCP is a network terminal protocol used as a messaging system between the Cisco Call Manager and a Cisco terminal such as the Cisco 7900 series IP phone (Cisco, 2009a).

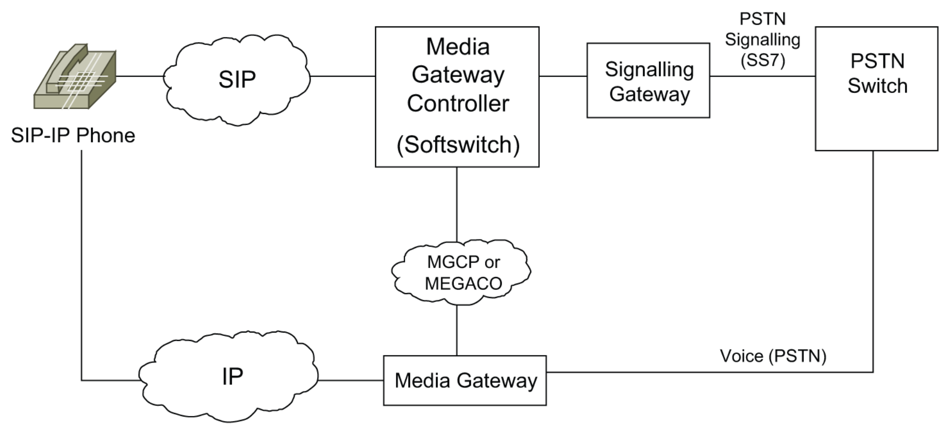
As has been explained, there are several VoIP network architectures and each one has different hardware and software requirements. For cost modelling purposes, it is necessary to define an efficient network architecture and calculate the usage factor of all the network elements involved in a VoIP call[7].

2.2 The role of the softswitch and the IMS in VoIP deployments

For the provisioning of VoIP services, many operators have chosen the softswitch and/or the IP Multimedia Subsystem (IMS), which are explained below.

The softswitch is a piece of software that switches calls by using software instead of a hardware device. A softswitch can be used for controlling VoIP calls inside a VoIP network and also for the interconnection of a circuit-switched telephony network with a VoIP network. As is depicted in Figure 2, the softswitch architecture is composed of a

Figure 2 The Softswitch architecture



Source: Collins (2003)

Media Gateway Controller, which is the softswitch itself, and Media Gateways and Signalling Gateways. A Media Gateway helps convert circuit-switched voice calls into VoIP packets and vice versa, whereas the Signalling Gateway adapts the Signalling System 7 (SS7) circuit-switched signalling protocol to a VoIP signalling protocol. Several signalling protocols can be used by the softswitch, e.g. H.323, SIP or MGCP. However, as SIP is being adopted by a number of voice operators, the manufacturers tend to implement SIP in the softswitches.

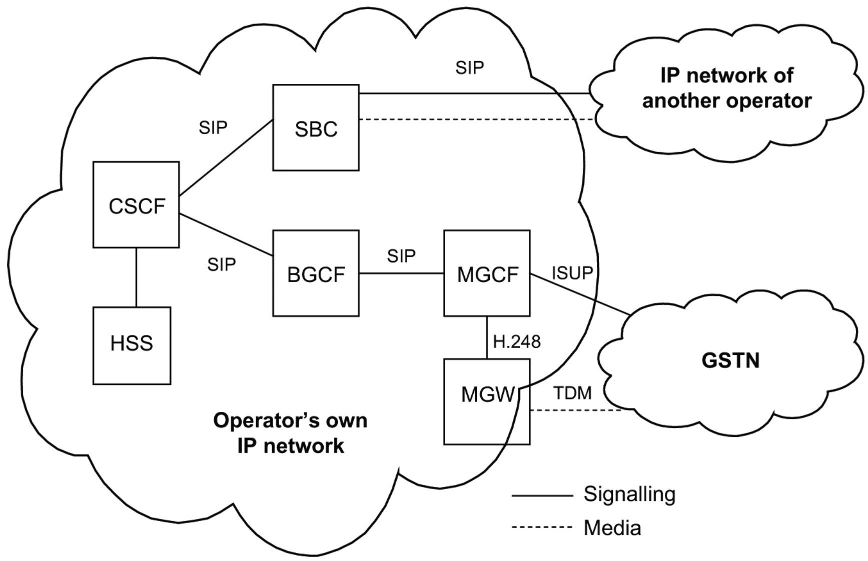
The IMS is a network architecture designed for the provisioning of services in a converged multimedia environment. It is a platform that manages multimedia services, among which voice is just another service that can be provided. SIP is one of the key protocols in the IMS network architecture. In the horizontal architecture of the IMS, the application, control and access/transport layers are clearly separated and they do not necessarily belong to the same operator. The IMS network architecture is shown in Figure 3. The fundamental nodes of the IMS architecture are the Home Subscription Server (HSS) and the Call Session Control Function (CSCF). The HSS contains the user profile database for authentication and authorization purposes (Poikselka and Mayer, 2009). The CSCF node controls the signalling by using the SIP protocol. The Serving-, Interrogating- and Pro-Call Session Control Functions (S-CSCF, I-CSCF and P-CSCF) are the roles of the CSCF node. For the interconnection with the General Switched Telephony Network (GSTN), two nodes are employed:

1. the Media Gateway Control Function (MGCF); and
2. the Media Gateway (MGW).

The MGCF manages signalling information and controls the Media Gateway. The Media Gateway translates RTP/UDP/IP packets into TDM signal streams. The Breakout Gateway Control Function (BGCF) chooses the route of the telephony session and the Session Border Controller (SBC) is an IP-to-IP gateway.

The difference between the capabilities of the softswitch and the IMS lies in the type of service that will be provided. If an operator is interested in providing end-customers with a voice service with few related applications, then probably the softswitch is the better alternative. The IMS is an architecture that helps create and deliver different types of services in an easy way. If an operator is planning to offer multimedia services, then the IMS could be the best alternative. Current cost models of VoIP networks could consider the SIP softswitch as the relevant node for the management of VoIP communications. However, as it is expected that operators will provide customers with multimedia or advanced services in

Figure 3 The IMS network architecture



Source: Based on Ericsson (2007)

the future, and as fixed and mobile networks will converge, the IMS could be regarded as the future call control node in cost models.

2.3 VoIP traffic in shared channels

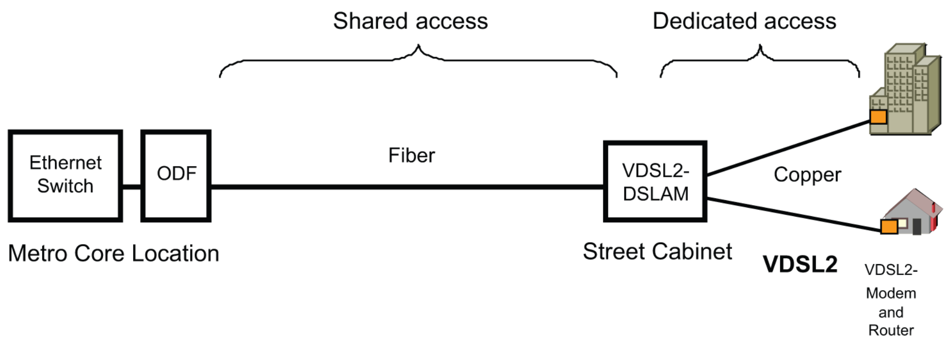
An important variable of cost models is the user-generated traffic volume that is delivered into the network. A traffic model uses as inputs the traffic rate and the generated packets' size. The size of a VoIP payload packet depends on the codec used for digitizing the analogue voice and on the headers that are added to the payload information necessary to deliver the packet. It would be a mistake to ignore the importance of the size of VoIP packets' headers because they could be the major component of the VoIP payload packet[8].

Before network convergence, every network was designed to deliver one type of service: television, telephony or data. Currently, however, networks carry several types of traffic. For that reason, one of the issues that should be included in VoIP cost models is the proportion of cost that corresponds to the VoIP service. In a Next Generation Access (NGA) network, the access line (fiber, cable or copper with xDSL) is shared among different services such as voice, internet data, video, etc. For example, in a Fibre to the Curb/Very High Speed Digital Subscriber Line 2 (FTTC/VDSL2) access network, the copper line will be used exclusively by every user (see Figure 4). In the fibre segment, all the traffic generated by the applications used by all the customers will be rivals for the common bandwidth. Thus, a cost model of a VoIP service should estimate the appropriate share of the voice service in order to allocate the costs properly.

2.4 Quality of service in VoIP networks

Another important technical issue is the assurance of Quality of Service in VoIP networks[9]. A real-time service like VoIP requires strict levels of packet loss, delay and jitter in an end-to-end communication. The values of these parameters have been specified in a few studies (ITU-T, 2002; Szigeti and Hattingh, 2004). In best-effort IP networks that do not have QoS techniques, sometimes it is not possible to meet these requirements. However, quality of service techniques improve the quality of experience (QoE) of the end user. There are three basic QoS mechanisms in IP and VoIP networks:

Figure 4 FTTC/VDSL2 access network



Notes: ODF: Optical Distribution Frame; DSLAM: Digital Subscriber Line Access Multiplexer
Source: Wik-Consult

1. prioritisation;
2. capacity reservation; and
3. over-dimensioning.

For the prioritisation of data, the Differentiated Services (DiffServ) technique can be used (Internet Engineering Task Force, 1998). Packets with the DiffServ feature have information in the packet header about the type of service the packets belong to, which will be used by routers to give packets a corresponding priority at the moment of forwarding them to one of the links attached to the router. For the reservation of traffic capacity, the Integrated Services (IntServ) technique can be used. One of the most important techniques used in the Integrated Services architecture is the Resource ReserVation Protocol (RSVP) (Internet Engineering Task Force, 1997). This protocol reserves capacity along the path between the sender and the receiver before a transmission takes place. However, Integrated Services techniques have major problems of scalability in large networks[10]. The trend in the industry is to use DiffServ instead of IntServ. For the over-dimensioning of transmission capacity, the operator deploys more capacity in terms of additional links and nodes.

In many IP networks, it will take some time until VoIP catches up with the quality of voice of current circuit-switched networks. Probably, this enhancement of the VoIP quality will be achieved by deploying broadband networks with more capacity and by using QoS mechanisms. If the VoIP provider is the owner of the VoIP network, then it can manage the Quality of Service inside the VoIP network. The different characteristics of the services that are provided are detailed in the Service Level Agreement (SLA) that is signed by the VoIP provider and the end-customer[11]. To meet the strict time delay requirements required by VoIP connections, the VoIP operator could deploy routers or switches that support QoS mechanisms. The capacity of the links or systems could also be expanded. In any case, the deployment of QoS mechanisms, which could entail hardware and software acquisitions, will have an impact on the cost of the service and, therefore, cost models should calculate this additional cost. A few of the issues that should be addressed in cost models are the following:

- What is the cost (CAPEX and OPEX) of a high-quality voice service that uses QoS mechanisms?
- What is the cost of a VoIP service that is not reliable all the time and that is provided over a best-effort IP network?

The degree to which the implementation of QoS mechanisms affects the usage factor of network equipment should also be analysed.

3. Features of VoIP networks

A telephony operator needs to offer a set of services and meet a few legal requirements to be able to provide the telephony service. As will be explained, the provisioning of these services and features entails the deployment of additional equipment. The services and features described in this chapter need to be considered when calculating the costs of telephony services in IP networks.

3.1 Services provided

Besides the provisioning of the basic voice service, VoIP operators try to offer the same type of services that are available in circuit-switched telephony networks. Examples of these services are voice mail, caller identification, call waiting, call forwarding, call blocking, and three-party conferences. Moreover, IP networks enable the provisioning of advanced telephony services such as video telephony, which is a service that demands more bandwidth than the basic VoIP service. To provide these services, in some cases, additional hardware or software will be deployed, which will have an impact on costs. Another aspect that should be considered is the fact that each service requires a specific level of QoS, which will have an effect on the cost.

3.2 Telephone number translation

For the addressing inside the VoIP network, every VoIP operator can use the addressing system of the VoIP technology that was deployed. However, for the interconnection with a circuit-switched telephony network, it is necessary to translate the internal number or name of the VoIP user into a valid E.164 telephone number[12]. E.164 Number Mapping (ENUM) is an IETF standard that can be used to transform PSTN E.164 numbers into corresponding VoIP addresses, and vice versa (Internet Engineering Task Force, 2009c). The mapping function of ENUM enables calls from an IP phone to a PSTN phone. ENUM uses Domain Name System (DNS) servers for setting up the call. The usage of these servers, which is reflected in the usage factor, should be taken into account in cost models. Moreover, as more database queries are necessary, an increase in the signalling traffic volume is generated. This increase should also be considered in the cost model.

3.3 Number portability

Number portability is a feature that enables telephony subscribers to maintain the telephone number when changing the telephone provider or when moving to a new location. In circuit-switched networks, number portability is provided by using databases that belong to the intelligent network (IN) of an operator. When it is detected that a destination number is not in the operator's own network and that the number has been ported, a query is sent to the Number Portability Database (NPDB). This database knows the current location, i.e. the current network operator, of the destination number. In VoIP networks there is also a consultation with a server or a database. As there are different VoIP technologies and network architectures, operators can implement the number portability function in different ways. However, the ENUM number translation is gaining support in the industry and it could be used in LRIC cost models as a state-of-the-art number portability system. The mapping of telephone numbers and the provisioning of number portability then entails the deployment of servers that support these features. The usage factor of a few network elements and the possible increase in signalling traffic should be considered in the cost model.

3.4 Access to emergency services

In many countries there is a number assigned to emergency services calls. In Europe, for example, the Universal Service Directive of the European Union (EU) defines the number 112 as the single European emergency call number (The European Parliament and the Council of the European Union, 2002b). The VoIP service can belong in the EU to one of the following two classes: Publicly Available Telephony Service (PATS) or Electronic Communication Service (ECS). Among other requirements, a PATS service must provide access to emergency services. The ECS category targets a wide range of services, as it ‘‘consists wholly or mainly in the conveyance of signals on Electronic Communications Networks’’ (The European Parliament and the Council of the European Union, 2002a). A member state of the EU can define whether the VoIP service belongs to the PATS or ECS category.

The difficulty in providing access to emergency numbers lies in the fact that it might be technically complex to determine the location of the VoIP user (Elixmann *et al.*, 2008; European Regulators Group, 2007). When a customer contracts a VoIP telephony service, usually the VoIP user gives his/her address during the registration process. If the VoIP user is non-nomadic and generates an emergency call from the address that was registered with the VoIP operator, then it is technically feasible to route the call properly to the next Public Safety Answering Point (PSAP). Conversely, an emergency call generated by a nomadic user from a location different to the address provided to the VoIP operator may have technical difficulties in being routed properly. There are ongoing efforts that propose solutions to overcome this inconvenience. For instance, the IETF ECRIT Working Group has proposed mechanisms for the routing of emergency calls with Internet technologies (Internet Engineering Task Force, 2009a). However, for the moment, this matter remains an open issue and time will pass before a standard can be widely accepted by internet service providers (ISPs) and VoIP providers. For cost calculation issues, it should be considered that the network nodes that provide the VoIP service, such as the switches, routers and servers, have to be able to route emergency calls to the proper PSAP. The usage factor of these network elements, a possible increase in signalling traffic volume and the possible effect of QoS mechanisms will have an impact on costs[13].

3.5 Security issues

Two services related to the provisioning of call information to authorized security forces are lawful interception and data retention. Lawful interception entails the obligation of providing access to telephony calls to intelligence services and law enforcement agencies, whereas data retention is the obligation of network operators of storing call detail records of telephony traffic for a period of time.

One of the aspects that should be taken into account by lawful interception mechanisms is the fact that usually VoIP signalling and data take different paths. There are technical solutions to this matter, as long as the security forces have access to the network that carries the signalling and data traffic. Nevertheless, if a nomadic user makes a VoIP call from or to an IP network that is not under the coverage of the security forces, it might be difficult to intercept the call.

The Directive about Data Retention of the European Union requires member states to keep the following information for a period of time of between six months and two years (The European Parliament and the Council of the European Union, 2006). To:

- trace and identify the source of a communication;
- trace and identity the destination of a communication;
- identify the date, time and duration of a communication;
- identify the type of a communication;
- identify the communication device; and
- identify the location of mobile communication equipment.

In VoIP networks it is possible to obtain this type of signalling information in the softswitch. However, for nomadic users who are located outside the home network, it is possible to keep track of the current IP address, but it could be difficult to identify the precise location of the nomadic user.

In several European countries the governments make a refund of the necessary investment to provide lawful interception and data retention services, i.e. for the use of routers, switches, servers and databases[14]. If these costs are refunded, then they should not be taken into account in the calculation of termination rates.

3.6 Additional systems

Typical systems of any telephony network should also be taken into account when calculating the costs of the termination rates. The network monitoring system, for example, requires equipment and personnel. The databases of the billing systems and the cost of personnel and equipment for the customer care system may not be considered in the calculation of wholesale termination rates because they concern retail services costs.

4. Interconnection of VoIP networks

The interconnection of IP-based networks has profound technical, economical and regulatory implications, as has been explained in (Marcus and Elixmann, 2008). This section describes two issues that have implications for the cost of the VoIP service:

1. the definition of the interface for the interconnection; and
2. the location of the point of interconnection.

4.1 Interface for the interconnection

Nowadays, voice operators have two possibilities for the exchange of payload and signalling information: either they use a circuit-switched interface or an IP-based interface.

The network interfaces adapt the voice codecs and the signalling system of one operator into the codecs and signalling protocols employed by the other operator. The voice signals exchanged through the circuit-switched interfaces do, in many cases, work with the same version of the voice codec that is used (PCM64, ADPCM, etc.). Most public circuit-switched telephone calls use the Signalling System 7 for signalling. There can be a few differences between the national variants of the SS7 protocols implemented around the world. A VoIP operator that interconnects with a circuit-switched voice operator, or with any operator that is obliged to interconnect through SS7 interfaces, needs to adapt its VoIP protocols[15]. Therefore, it will be necessary to use a Media Gateway to convert IP packets into circuit-switched packets; Signalling Gateways are also required to convert circuit-switched SS7 signalling protocols into VoIP signalling protocols. Figure 2 shows the gateways of the softswitch architecture used for the interconnection with a circuit-switched network. For the interconnection through an IP-based interface, it is necessary to adapt the different VoIP signalling protocols. Therefore, signalling gateways will also be necessary.

From the point of view of cost calculation, the question that arises is the definition of the network elements needed to adapt the different VoIP payload and signalling variants. For example, if according to the country-specific regulation it is mandatory to exchange voice traffic through SS7 interfaces, then VoIP operators will need to deploy the corresponding media and signalling gateways. This will have an effect on the costs of the termination rates[16].

4.2 Location of the point of interconnection

Another aspect that has an effect on the cost of the termination rates is the location of the point of interconnection (PoI). Normally, a point of interconnection that is located far away from the VoIP network will entail an additional cost for the VoIP operator, because the VoIP operator will have to assume the cost of the transit service from its network to the point of interconnection of the destination network. It is up to the operators and/or to the regulatory authorities to define the most appropriate interconnection mechanism.

5. Conclusions

The deployment of VoIP networks raises a number of technological, economic and regulatory issues that affect the determination of voice termination rates. This article provides an overview of the features of VoIP network architectures that will have an impact on the cost modelling of termination rates. The analysis covers three aspects:

1. VoIP techniques;
2. features of telephony services in VoIP networks; and
3. interconnection of VoIP networks.

First, one of the aspects to be considered by network cost modellers is the definition of the network architecture. Unlike circuit-switched telephony networks, where there are a limited number of standards that are adopted by most operators, VoIP operators have several possibilities when selecting the VoIP systems and protocols (e.g. H.323, SIP, and MGCP). Even though there is no precise answer about which is the best VoIP technology, the SIP softswitch seems to be the current state-of-the-art architecture and it could be considered as an efficient reference model architecture. The usage factor of the network elements involved in a VoIP call should be calculated. The traffic volume generated by VoIP users should also be considered appropriately in cost models. As the VoIP service requires strict levels of loss, delay and jitter delay, Quality of Service techniques could be implemented. If this is the case, the implementation of QoS could lead to costs that should be taken into account.

Second, the features of a telephony service provided by a VoIP operator require the deployment of appropriate servers and databases. Examples of cost elements that should be considered are typical telephony services such as voice mail, caller identification, call waiting, etc.; telephone number translation; number portability; access to emergency services; security systems; and network monitoring systems. The provisioning of these features will have an effect on the usage factor of a few network elements and on the traffic volume, especially the signalling traffic, that will be generated.

Third, the definition of the point of interconnection is an issue that can entail costs for the VoIP provider. Depending on the regulatory framework of every country and on the agreements signed between telephony operators, it would be possible to use an IP-based or/and a circuit-switched SS7 interface for the interconnection. If SS7 is mandatory, then a VoIP operator will probably have to assume the costs of the corresponding signalling and media gateways. The location of the point of interconnection has implications for the costs because it could be necessary to consider the cost of the transit service from the point of interconnection of the originating network to the closest point of interconnection to the addressee.

In sum, the findings of this study could help cost modellers to reflect on the particularities of VoIP networks. A future study could entail the elaboration of a cost model that includes the cost elements described in this article.

Notes

1. In this article VoIP is considered a service that has to meet a few specific technical and regulatory requirements and that needs a license to operate in a country. Some authors prefer to use the term ‘‘telephony over IP’’ when they refer to the same type of service. In this study the terms ‘‘telephony over IP’’ and ‘‘voice over IP’’ refer to the same service.
2. Peer-to-peer (P2P) VoIP traffic was not considered in the VoIP statistics published in Commission of the European Communities (2009). Therefore, the figures of total VoIP traffic generated might have been higher.
3. For example, Skype with its SkypeIn and SkypeOut services.
4. An alternative to the CPNP rule is the bill & keep pricing principle. In a bill & keep arrangement, an operator does not have to pay another operator the wholesale termination charges. Each operator bills its own customers for the inbound and outbound traffic. In this case, it would not be necessary to calculate the interconnection fees.
5. Many countries in Europe have adopted the LRIC cost model for the calculation of termination rates.

6. VoIP open standards such as the Inter-Asterisk eXchange protocol version 2 (IAX2) (Internet Engineering Task Force, 2009b), which works with the Asterisk open source Private Branch eXchange (PBX) server, are not described in this section. The reason is that IAX2 is suited to private VoIP networks with low-budget limitations, and it is not expected to be used by major VoIP service providers.
7. The usage factor (or the routing factor) measures the intensity of use of a network element by a specific service.
8. For example, a voice codec could generate a VoIP payload of 6 Kbps, but with the corresponding IP/UDP/RTP headers, the packet could require in practice a bandwidth of 50 Kbps or more, even if compression mechanisms are used.
9. Frederiksen (2006) studies the case of VoIP suppliers with and without QoS and concludes that ''it is an unanswered question how important it is for the customers to have a guarantee for QoS''. Later, Constantiou and Kautz (2008) in an analysis of IP telephony in the Danish market find evidence that price would be more important than quality of service. However, both studies do not neglect the importance of QoS for the improvement of VoIP service provisioning.
10. Even though there are proposals that help alleviate the scalability problem in IntServ architectures, they have not been widely deployed.
11. A VoIP service provider can also negotiate with network operators a service level agreement for the provisioning of voice services with specific levels of quality.
12. In this case, it is assumed that the interconnection is done through circuit-switched SS7 interfaces. Section 4.1 sheds light on the technologies used in the interconnection interfaces.
13. The calls to emergency services will probably be treated as a higher priority and, as a consequence, they could belong to the highest QoS class.
14. The storage systems that keep information about customers' communications are mostly databases.
15. In many countries it is mandatory to use SS7 signalling for the interconnection between voice operators.
16. Small VoIP operators could probably prefer to interconnect by means of IP-based protocols and hence avoid the investment on circuit-switched interfaces.

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