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Voice over IP: technical presentation and economic analysis

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Abstract

The public switched network is today the way used by each one of us to make a phone call. However, data networks are spreading out more and more and replace the analogue copper wire-based networks. If there already exist techniques that have stood the test of time to carry voice over data networks, as ATM and frame-relay, the hegemony of IP as regards to private or public networks has lead to pose the problem of voice over IP, which continuously gains ground along the fact that the mastering of the quality of service problems are solved. Those are indeed crucial for an IP network. What is therefore the state of art in this field, and what are the situations that can be thought of when wanting to use a VoIP network?

We should not however imagine that, because we are facing a new promising technology, its success will be staggering. This will be the case only if economic reasons are found for its use. We have therefore to ask the question about knowing to what extent a VoIP network can be interesting for an organisation whose primary aim is not scientific research; thus, we will propose a method to help a decision-maker think about all the parameters he has to take into account in order to estimate the opportunity of deploying a VoIP network.

Résumé

Le réseau public commuté est la méthode aujourd'hui employée par tout un chacun pour effectuer un appel téléphonique. Cependant, de plus en plus, les réseaux de données se développent et remplacent les réseaux analogiques reposant sur la paire torsadée. S'il existe déjà des techniques éprouvées pour le transport de la voix sur un réseau de données, comme ATM et frame-relay, l'hégémonie d'IP en matière de réseau privé ou public a amené à poser le problème de la voix sur IP, qui gagne du terrain parallèlement à la maîtrise des problèmes de qualité de service. Celles-ci sont en effet cruciales dans le cadre d'un réseau IP. Quel est dès lors l'état de la recherche en ce domaine, et dans quelles situations peut-on envisager d'utiliser un réseau VoIP ?

Il ne faut cependant pas imaginer que parce qu'il s'agit d'une nouvelle technologie prometteuse son succès va être foudroyant. Il ne le sera que si des raisons économiques sont trouvées pour son emploi. Il faut donc se poser la question de savoir dans quelle mesure un réseau VoIP peut être intéressant pour une organisation dont le but n'est pas la recherche scientifique ; il s'agira dès lors de proposer une méthode d'aide à la décision économique dans le cadre d'un réseau de téléphonie privé que l'on voudrait basé sur IP.

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Introduction

A century ago, Alexander Graham Bell patented the telephone – it was back in 1876. His invention immediately met a huge success. The phones had to be sold in pairs, and be joined by a single copper wire. This was the first way to carry voice from one point to another. A drawback was that each phone had to be linked with its remote counterpart to establish a communication.

To avoid an uncontrollable mess of coppers wires all around the world, it became clear that some concentration points where needed. It is the reason why, the year 1878 saw the first switching office. This was the very basis of the public switched network, or PSTN.

The twentieth century came, and with him an era of telecommunications. Especially after the second World War, and the growth of the computer industry, more and more networks were created, either local area network, wide area network or metropolitan area network. One of the first long-distance dial-up connections lead to the Arpanet, and from then on to the world-wide Internet.

As the internet became popular with the launch of the world-wide-web, the so well known WWW today, the internet protocol became to spread itself all over the world. For the cost of a local call, everyone is now able to reach a computer distant from thousands of kilometres. For the cost of an international call, everyone is able to reach a person distant from thousands of kilometres. The price we pay for an hour of local call may be less than the one paid for one single minutes of an international call.

Seeing that difference, some people began to thought of the Internet as a cheap means to call someone distant from several thousands of kilometres: now that IP goes everywhere, why not to use it to speak to everyone? Several freeware, shareware or commercial software were proposed to take advantage of the Internet. Most of them were reserved to a few people, happy enough to enjoy a new technology to forget about the bad quality of the call.

If voice over IP was a sweet dream a few years ago, the idea did not seem so stupid to researchers who began to design efficient ways to communicate thanks to IP, and today VoIP is a popular subject of conferences.

The purpose of this thesis is to study what can be done with VoIP, in what situations it can be used, and how it can be used. What are the requirements of the VoIP, either in protocols or in network capacity. This will be the topic of the first part of this work: present an overview of what can be done with IP, and how this can be done.

If VoIP, as a new technology, may appear attractive to technicians and computer gurus, its real future will be decided by economic reasons. It is the reason why an economic study of the VoIP is important. This will be the topic of the second part of this work: see how the VoIP can be economically interesting and what are the elements to take into account when evaluating to cost of an IP-based telephony network when it needs to be compared to the price of the traditional PBX-based solution.

In order to illustrate the propositions made in the second part, this work will be brought to an end with a case study of a VoIP-network deployment.

Part I. IP Telephony

Introduction

In this first part, we will focus on both the operational and technical aspects of the so-called IP telephony. Before going on to the next part and its economic model, it is actually important to bear in mind what IP telephony means and how that technology works.

At first, we will examine the position of the VoIP in the telecommunication universe, and the specific aspects of IP as the basis for telephony applications when compared to other VoN solutions, namely frame-relay and ATM.

Having shown the way IP can be useful to support regular telephony applications, we will then take our focus to the operational side of LAN's and WAN's and try to cover the varied cases where IP telephony can be applied. It is, at this stage, important to notice that under the generic "IP Telephony" term, several meanings are hidden. If those meanings are always related to communication, this one can take several aspects such as video conference or data transfer (file transfer or application sharing, such as white board applications). The topic of this work is essentially related to basic telephony and as a consequence we won't have any interest in video or data transfer, and so restrict our vision of IP Telephony to one single meaning.

This first part will then be brought to an end with a presentation of the VoIP protocols in use and development today and see how they fit to the operational requirements and cases.

Chapter 1. Voice over IP in the telecommunication universe

1.1 The traditional telephony system

The traditional telephony system has evolved from a pure analogue system to a mix of digital and analogue links, as underlined in the introduction of this work. This introductory chapter has for purpose to examine more precisely the current telephone environment and its main characteristics. We will here describe the fundamentals of the public switched network, referred to as PSTN. The POTS acronym, standing for plain old telephony system is also used to refer to today's telephony system, both in private en public environments, usually meaning an analogue phone, and by extension an ISDN phone as well.

What is the way a phone call is processed today? From the user point of view, when you pick up your phone, the latter goes "off-hook", signalling to the carrier's switch that it wants to make a call. The switch provides a dial tone and from that moment on you can dial the number you want to reach. It's the PSTN job to find the dialled phone and send a ring tone to it, or give you a busy tone. As soon as the called party answers, you can speak.

All the job is accomplished by the network and its complex equipment. The phone you use in everyday life is quite stupid: all it does is sending and receiving electric signals to and from the network, signalling its off or on-hook condition as well as DTMF tones used for numbers. Those tones have taken the place of yesterday's pulses. It's the only clue that tells there is now a digital handling of signals, as compared to the old pure analogue system (at least when using a standard phone). But let's go back to the fact the complexity of the PSTN resides in the network, and think about what is needed to really give a phone call. We first need a way to tell the network we want to place a call or that the phone line is already in use. This is done simply by the on-hook and off-hook conditions of the phone. The circuit is closed or open, and that's the way the phone requires a dial tone from its local switch. Quite simple, but there are actually several kinds of signalling used to perform that task, though it is out of our focus.

Now, the problem is inside the network: how does the switch know how to handle the DTMF tones it receives from the phone? It needs to find out where to route the call throughout the network to the final destination, from switch to switch, establishing a route from hop to hop. The E.164 numbering scheme helps to have a standard way of structuring phone numbers, so that there's a hierarchical structure in the number, from the local link to the international one.

It is important to notice that a circuit is established throughout the network, circuit which will remain tied up in the network for the whole call duration, whatever its real use. The core network uses a 64 Kbps PCM digital scheme to carry the voice[1]. This means that huge bandwidth capacities are required from the carriers to cover all potential needs. What it worrying in this is that an average conversation means 60% of speechless time, which could be seen as a pure loss of means in the network. This is due to the scheme used which relies on circuit switching.

We've told that there is still an analogue part in the PSTN: this is merely the so-called local loop, i.e. the copper twisted pair that goes from the phone to the central office, which is the point of entry in the carrier's digital network.

In order to route the call and convey important information about call processing and the way to handle some special numbers (0800, 0900, etc), there is a need of exchanging information between call components inside the network. This exchange is called signalling, and is described by a special protocol for the PSTN, the Signalling System 7 (SS7).

SS7 provides what is called "out-of-band" signalling, which means that dedicated links are used to convey signalling messages. That way, signalling messages can be transmitted over relatively high-speed links, and signalling can occur during the entire call, and no just at the beginning or the end of the call. It also permits signalling to network elements to which there is no direct trunk connection.

If SS7 is widely in use, the way it has been implemented in Europe and North America differs. While European networks use associated signalling where the signalling link follows the same path as voice links between nodes, North American networks define a completely separate signalling network and is known as the North

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[1] At least in countries where the digital network has taken the place of the old analogue network, which can still be found today.

American SS7 architecture, made to be more efficient and make it possible to route signalling messages easily between each node in the network, directly connected or not.

Describing the SS7 in details is out of the scope of this work, but the North American SS7 has an interesting structure. That structure is shown on figure 1. As we can see, the signalling network is completely independent from the media network. STPs and SCPs are part of the signalling devices, and stand for, respectively, Signalling Transfer Point and Service Control Point. Class 5 switches are those equipping central offices, i.e. the point of concentration of local loops for a delimited area. Class 4 switches are those equipping the core network – there are five levels of hierarchy defined within the PSTN (in its US version).

To provide redundancy and reliability, STPs are often "mated" in the SS7 network so that a failure doesn't affect the availability of the telephony service. This is not shown on the figure. STP's are used to tell the switches how to handle and route the call, on basis of numbering information. Databases and SCPs are used for enhanced services and what is called "intelligent networks". Their purpose is for example to provide additional information on calls to 0800, 0900... numbers which need to be matched with a real customer number.

The Time Division Multiplexing network is the network carrying voice, usually in a digital form, as told here above.

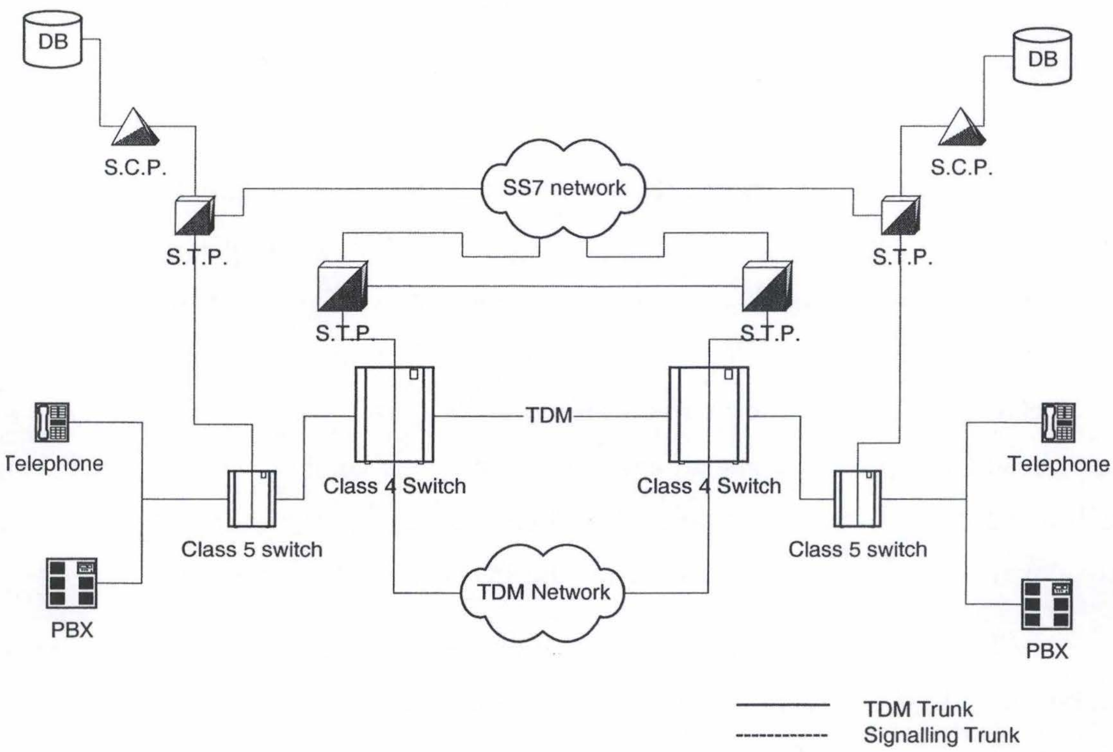

Figure 1: Overview of the PSTN with North American SS7 architecture

This was only a broad overview of the PSTN, but It was important for further chapters to have it in mind and to underline the difference between voice and signalling.

1.2 Merging of two worlds: data networks applied to voice transport

We have seen that today's PSTN relies on digital TDM networks. Once voice is digitised, it merely becomes a series of bits, as data of every kind would be. The PCM encoding is a toll quality coding and matches the analogue bandwidth available on a copper wire[2]. Such coding may be subject to compression, without

[2] The analogue available bandwidth is 4000 Hz. According to the Nyquist theorem, digitising that bandwidth requires a sampling frequency of 4000*2=8000 Hz. Using an 8-bit encoding, we get 8*8000 bits for a second of voice, i.e. 64000 bps.

degrading the quality of voice, or with such a slight degradation it is hardly noticeable.

There are today three major kinds of network which may be used to carry voice. The first of these, which is sometimes used in core telephony networks, is the ATM network. We also find Frame-Relay networks and more recently IP networks to carry voice communications. In this section, we will give an overview on how and why ATM and FR may be used to carry voice. VoDSL is also envisaged.

Voice over ATM

Asynchronous Transfer Mode networks are designed as high-speed cell switching networks: a common speed for ATM is 155 Mbps/sec. ATM was designed to get a single high-capacity, flexible network which would replace all different network we can find in the telecom universe, such as PSTN networks (signalling and voice), cable TV networks, ... It was supposed to be used for delivering video-on-demand, hence the speed of 155 Mbps which is required for digital television. Using fibre optics links, ATM networks get the reliability of those and do not need a high-fidelity mechanism to ensure that reliability. It will be the job of the application using the network. It has though an interesting feature in enforcing the sequence respect: a cell sent just after another one is sure to arrive just after it, but not necessarily with the same time sequence.

ATM is based on cell switching which may provide either constant bit rate or variable bit rate communications. It is also a connection-oriented network using virtual circuits, which means that a circuit has to be established throughout the network to connect two endpoints. Once the circuit is established, it will remain the same for the whole connection duration. We immediately think to the circuits established in the PSTN, either by mechanical copper wire switches or on the TDM digital network.

ATM uses several so-called ATM Adaptation Layer in order to provide transport services. Those AAL have been designed for several purposes, and with different characteristics (CBR, VBR, real-time or not, connection-oriented or connectionless). Four kinds of AAL have been designed by the ITU, while the AAL/5 was designed by the computer industry.

The AAL1 of ATM as defined by the ITU is intended to carry real-time data as voice or video. ATM is born as a high-speed network running on highly reliable fibre optics as a future replacement for the telephone system and means of distributing digital television. Using AAL1, uncompressed voice can be carried along a virtual circuit, in the way of the old copper wire circuit.

So, with AAL1 we get a high performance network adequately designed to carry digitised voice. ATM is so used as core network by some telephone carriers. AAL1 uses the ATM constant bit rate (CBR) circuits, so that the problem of using AAL1 is the one of the PSTN: the circuit is established for the whole call duration, leading to a loss of potentially usable bandwidth.

AAL2 is also intended for real-time traffic but uses VBR and is more commonly used to perform VoATM tasks where compressed voice is used.

The main problem of ATM for small or medium organisations is its high cost. Besides, its high speed is not required for a private use, and a more flexible and affordable way to carry voice has to be found. Frame-relay networks, which share some characteristics with ATM networks are more adequate in this context.

Voice over Frame-Relay

One of the shared characteristics of FR and ATM networks is the connection-oriented circuit switching feature. As physical networks became digital and more reliable, frame-relay was designed with the purpose of carrying data without any guarantee, excepted the sequence respect, exactly as ATM.

PCM-coded voice uses 64 Kbps bandwidth and a circuit must be established within the network for the whole call duration, which is what is done over the public TDM network (maybe using ATM). FR networks are designed to deliver bandwidth on demand: if the network is not congested and the user asks for more bandwidth, he gets it. The advantage of FR relies on that feature: you pay for a guaranteed bandwidth that you are sure to get on every moment. On top of that, you may get more bandwidth if asked and available. A FR network is flexible enough to do that.

Using compressed voice, small and medium organisations can take advantage of FR networks to carry their voice communications. According to the number of lines they want between their sites, they ask and pay for the corresponding bandwidth, or even less if they only may need so much lines: as we have already told it, FR services are paid on a guaranteed available bandwidth basis while more capacity can be asked for (within agreed limits of course) if the network is able to deliver it.

Within a big network involving several switches, the sequence respect guarantee may turn to a drawback; as frames are correctly put in their original sequence at each node before being sent further, the delay of the frame propagation may be increased to a point that it becomes hardly acceptable to enjoy a good quality conversation. However, this should not occur when there is only one FR link on the path of the voice.

Voice over (A)DSL

With the emergence of that new technology, it has been thought that voice could be carried over (A)DSL networks. (A)DSL specifies the physical way to transmit bits, and a standard network protocol has to be used on top of it, so that VoDSL becomes VoATM, VoFR or VoIP according to the case. VoIP is developed in the next section.

Voice over IP

As a third solution to carry voice over data networks, we get IP networks. Here, we don't have anymore a circuit switching network, but a connectionless packet switching network. There is no virtual circuit established before the packets flow between the endpoints, and IP doesn't require heavy hardware to be implemented. We will now bring our focus to the VoIP.

1.3 IP telephony in particular

Why is IP so popular for voice applications?

Since the growth of the internet and the expansion of the Internet Protocol, IP has imposed itself as the most popular network protocol for businesses and home users. As more and more people deploy IP-based intranets, the demand for voice over IP has increased, leading some major vendors and engineering groups to find an interest in VoIP and means to provide the customer with high quality voice transport over IP networks. The success of IP as the leading network protocol is also due to the low cost of IP routers and other hardware devices when compared to ATM or Frame-Relay equipment.

In the latter section we did not explicitly tell where the VoN links were used, but it was somehow supposed that the links used were long-distance links used to connect several scattered sites on leased lines (or leased capacity in a public network such as the public FR network). ATM and FR are not technologies often found in local area networks where they would seem rather odd (though it is possible to perform LAN emulation over ATM). On the opposite, we have just told that IP is a popular protocol for intranets or LAN's, while it can also be used on long-distance links.

This is also why IP can be so interesting for voice applications: used on the LAN, it can also be used on the WAN without translation headaches: VoIP has now become a profit-generating market and the place for a heavy competitions among majors vendors, coming from both the world of data transport and the world of telephony. Routers become voice enabled, and PBXs become IP-enabled.

The need for QoS

If IP is a flexible and democratic protocol, its best effort packet switching strategy can be an important drawback on the way to toll quality voice. The best effort strategy means that no guarantee of any kind is given on delivery of packets: interarrival jitter, sequence respect, packet loss are all unknown parameters. On public network, this can be very annoying, but on private IP networks, it is possible to design the network to handle a predictable load without problem.

Within the public Internet, or even on heavily used private networks, IP can be a nightmare for voice. In order to make it possible to place a phone call on such networks, we need to establish quality of service (QoS) policies, such as IP prioritisation or bandwidth reservation to give voice traffic a special handling, thus assuring that voice calls are effective.

There are several ways to achieve the required quality of service. The most simple of these is merely to increase the bandwidth available in the network to be sure it is over all needs. If this is simple, it is also costly, and not very bright. The other ways to achieve a good quality of service on the network is to activate some policies. Those policies deal with the incoming traffic in a differentiated manner. It is possible to define a priority for each kind of traffic, either based on the packet content (for example, all RTP packets are given a high priority), or on its source/destination (for example, all traffic coming from computer X is given a high priority).

Several mechanisms ensure the enforcement of those policies. They can be based on resource reservation with the use of the Resource reSerVation Protocol (RSVP), or on queuing mechanisms implemented in the router software. Several logical queues are established, and the bandwidth usage allocated to each of these queues is regulated by allocation (scheduling) algorithms. A third way is to use the Type of Service field in the IP header. The positioning of the IP precedence bit allows to define 8 priority levels[3] for the IP packet. Normal IP traffic is given priority 0, as VoIP packets may be given priority 3 to 5. Higher priorities (levels 6 and 7) are reserved for network control use. The IP precedence bits will be used as a basis for queuing algorithms.

The problem of QoS mechanisms is that they have to be implemented at each hop in the network. On a private network, either LAN or WAN, the user can choose its equipment manufacturer and activate useful QoS policies – all possibilities are not always good to use. He can also establish a manual routing table so that its preferred traffic is handled by QoS-capable equipment. On the Internet, we can not

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[3] Those priority levels are defined as follows :

routine :	0
priority :	1
immediate :	2
flash :	3
flash-override :	4
critical :	5
internet :	6

be sure that such QoS policies are activated, and even if we decide to use the RSVP[4] protocol, some routers on the path between source and destination may not be able to handle the RSVP messages, leading to a possible bottleneck.

The need for appropriate signalling protocols

We have seen that within the PSTN, a very important role is devoted to signalling. On VoIP enabled network, this signalling is as much important. We need to describe call properties, to know how to handle a call, etc... IP in itself does not perform those tasks, and it is not its role. Neither it is the one of other commonly involved protocols when speaking about the Internet. What we face here is a new need in the world of the Internet, the need for solving telephony-specific problems. In a sense, we have to translate the existing usage of the PSTN to the Internet, or invent a brand new solution.

Two majors organisations are concerned by VoIP: the International Telecommunication Union (ITU), which deals with all telephony problems, and the Internet Engineering Task Force (IETF), which deals with all problems related to the Internet. Keeping the specific problems of VoIP in mind, both the ITU-T and IETF have designed specific protocols to perform those signalling tasks on IP networks, given their specific requirements. We will later in this work (see Chapter 3) describe the different protocols associated to VoIP.

The way to unified messaging systems

As we have already underlined it, IP telephony covers voice telephony, facsimile, videoconference, and data exchange. As we tend to integrate data and voice networks, both relying on the same IP protocol, it will become possible to integrate all communications needs within a single equipment: it is what the computer telephony integration (CTI) promises, the unified messaging services. This concept means that with a single application you should be able to handle you phone calls, facsimile messages, voice mails as well as regular e-mail. Imagine that you

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network : 7

[4] drawbacks of RSVP will be shown when presenting the protocol in section 3.2.3

receive an e-mail from a person you may want to speak to. A simple double-click on his phone number in its e-mail signature could trigger the voice application, making his phone ring from your computer. Your faxes and e-mails could be read to you on your mobile phone as mere voice mail messages when you are away.

The generalisation of IP, the development of speech synthesis and use of a common single network protocol can make this a near reality. This is maybe the most interesting advantage in deploying VoIP networks, able to handle data and voice.

Chapter 2. VoIP application cases

In the last chapter, we have tried to give an overview of which can be the place of VoIP in the telecommunication universe. In this second chapter, we will examine the different cases where IP can be used to carry voice, from a controlled private LAN to the case of a telephone service provider offering its services to the home user. We will distinguish six different cases and show how the use of VoIP can apply to them.

In the following sections, we won't talk about software solutions imagined to effectively enable VoIP on the networks. This will be the scope of Chapter 3.

2.1 VoIP on a local area network (LAN)

2.1.1 IP phone to IP phone on a pure IP network

In this context, VoIP is the single means of making a phone call on the local network. That case is illustrated on figure 1:

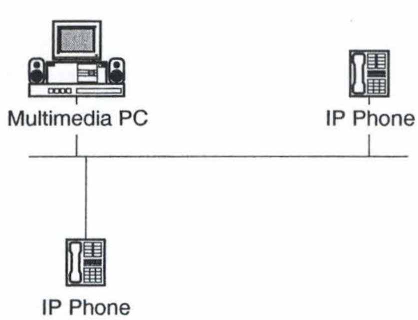

Figure 2 : the basic case for VoIP

Before going any further, it may be useful to define more precisely what we call an IP phone. Such a telephone resembles to any classical telephone, the only observable difference relying on the connector type. Instead of finding of regular RJ-11 connector (standard[5] for copper wire connector), we find an ethernet RJ-45 connector (notice that such an RJ-45 plug is also used for ISDN connections) and,

[5] The RJ-11 is actually the US standard, but such a connector is found on nearly any phone today. In the Belgian case, the RJ-11 connector is merely plugged into the Belgacom-type plug.

depending on its make, an external power supply. Such a phone must also integrate a software part to handle the different protocol stacks, for network protocols and signalling protocols. Let's keep in mind that an IP phone is a telephone with an integrated ethernet card, in order to be connected to a LAN. Figure 3 presents an example of IP phones, in this case the products of Cisco Systems. As we can see, those phones are not so special and look like any other business phone. The difference is found at the back with the Ethernet hub and the external power chord.

Figure 3 : Cisco IP phones

A PC with the necessary software and hardware may also be used as an IP phone. The network stack is present in nearly every operating system today, and it is not rare to find multimedia hardware in the PC: sound card, speakers and microphone. Many software applications can be found in the range going from freeware to commercial software, including shareware. Some of those applications handle video calls as well, and may or may not be compliant with IETF or ITU-T standards. The most common of these applications is without any doubt Microsoft NetMeeting which is given for free with Microsoft Windows. This application can handle voice, video or data communications and is compliant with the ITU-T H.323 standard. Figures 4 and 5 show the MS NetMeeting main windows, and advances parameter window, respectively. (French version)

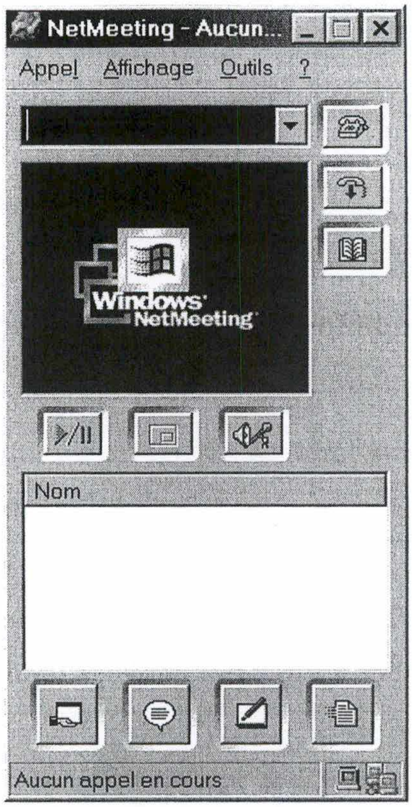

Figure 4 : MS NetMeeting 3.01 main window

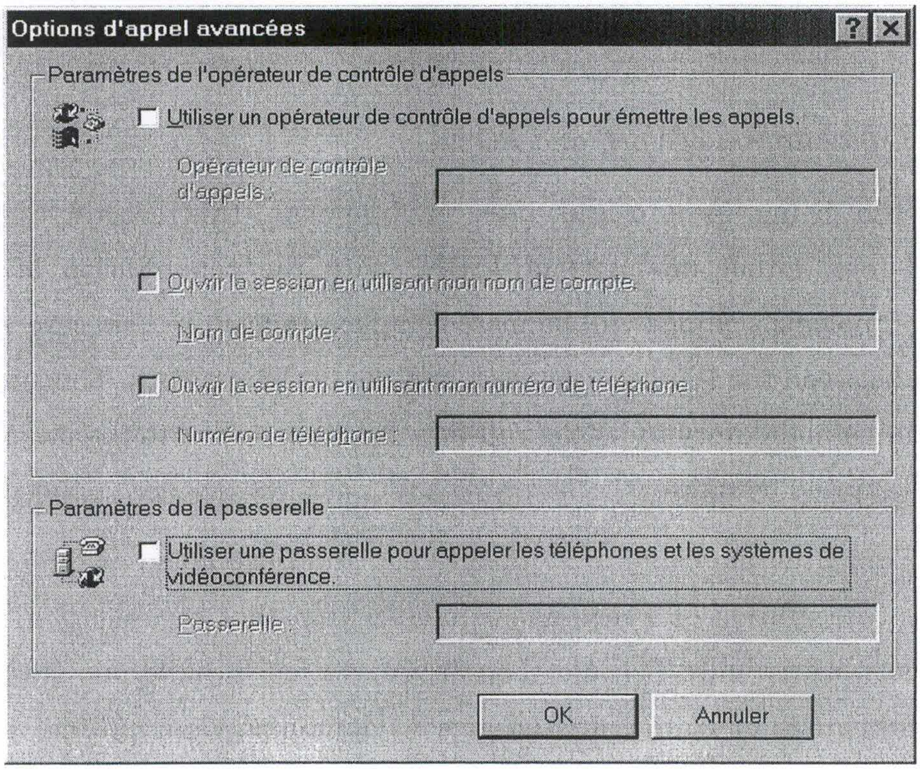

Figure 5 : MS NetMeeting advanced parameter configuration window

With such a simple, closed environment, the only persons you can reach are those whore are connected to the LAN. On a small LAN, there is no need for further equipment and phones may be manually configured, as well as the PC's. We face here a case similar to the management of a PC-only LAN: according to the size of the network, a system administrator is required or not, a network server is required or not, switches are required or not...

If the LAN becomes large, some management is needed. As VoIP may be heavily bandwidth-consuming, some QoS features may be useful, as well as dedicated servers. A DHCP server may be useful to automatically configure the phones, and it could be useful to have a centralised administration interface. There could also be a kind of dial plan within the network, and a VoIP router could be used to manage QoS and dialling routes.

As we see, even a simple LAN can somehow require specific equipment and management. This would probably be merged with ordinary LAN-management as seen for any large data network.

2.1.2 Cohabitation of VoIP and POTS

Still being in a closed LAN environment, this second case involves "classic" phones, either analogue or ISDN. With the cohabitation of those two different technologies, things will be more complex. Indeed, in this case we need an interaction between the IP phone network and the POTS network. This means that a gateway will be mandatory to let the call flow from a network to the other one. Such a case is illustrated by figure 6.

This situation could arise in the case of an extension of the organisation building, or the creation of a new department within that organisation, provided a new data network is built. It could then be interesting to build one single network carrying both kinds of traffic, and so give a later opportunity to get a single IP network throughout the buildings, when the existing equipment is obsolete. The reason to build a VoIP network could also be a maximum capacity reached for the present PBX. The reasons to deploy that IP-based telephone network are to be carefully thought, and part two of this work is dedicated to that problem.

The management problems of the first case are of course the same here.

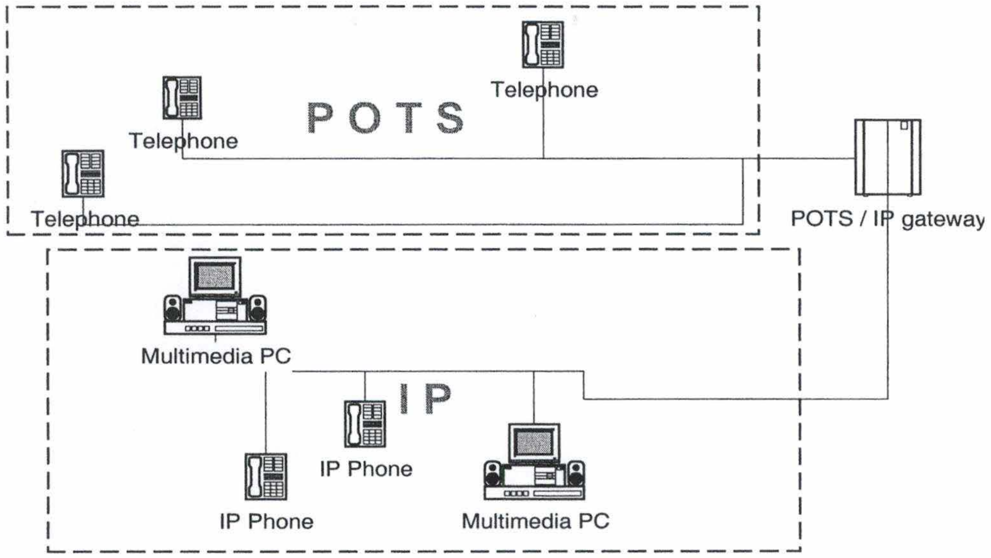

Figure 6 : POTS/VoIP cohabitation

POTS and IP network can also cohabit on an open network. Closed networks like the two first cases are unlikely to be met in real life. Telephony is made to communicate with the economic environment, and not only within a single organisation without external contacts. So, we will have to now come to a more realistic situation where the private network has a connection to the outside world. Based on the fig. 6 case, we need to put an interface to the PSTN somewhere on the network. That interface can be located on the POTS/IP gateway, if that gateway has direct POTS interfaces and plays the role of a PBX. This is illustrated by figure 7.

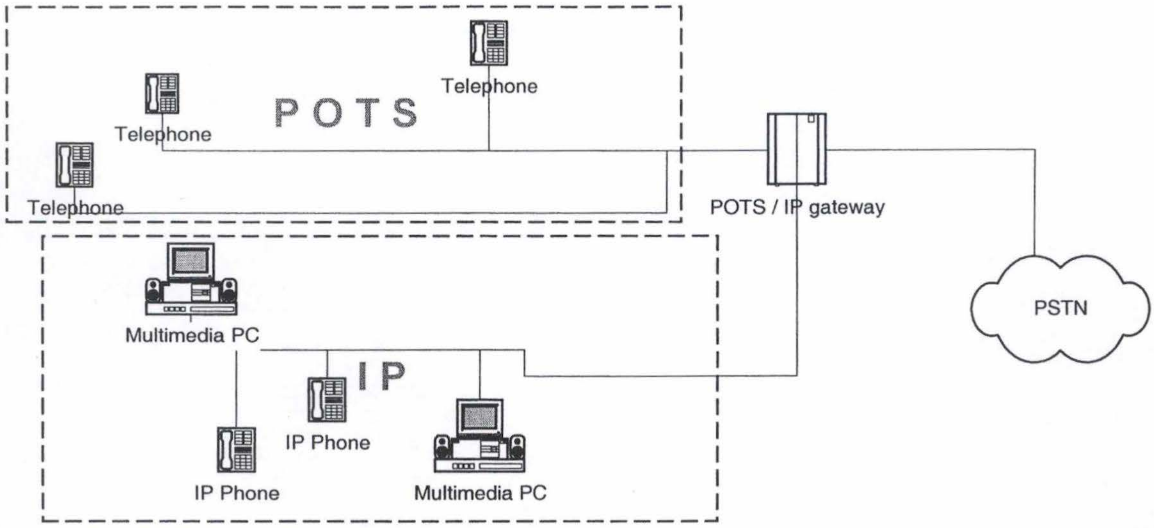

Figure 7 : gateway with PSTN interface

On figure 6 we did not consider the PBX, but in the case where the VoIP network is an add-on to the existing PBX-controlled voice network, such a device should be present, and connected via a Q.SIG (digital) or E&M (analogue) interface to the POTS/IP gateway. Both devices, the PBX and the gateway, may handle calls to the PSTN. It is then up to the network designer to choose where to put that interface, according to features offered by the device. Notice that the existing PBX will certainly already have a PSTN connection and that the most simple solution would be to keep that connection active. That case is illustrated by figure 8.

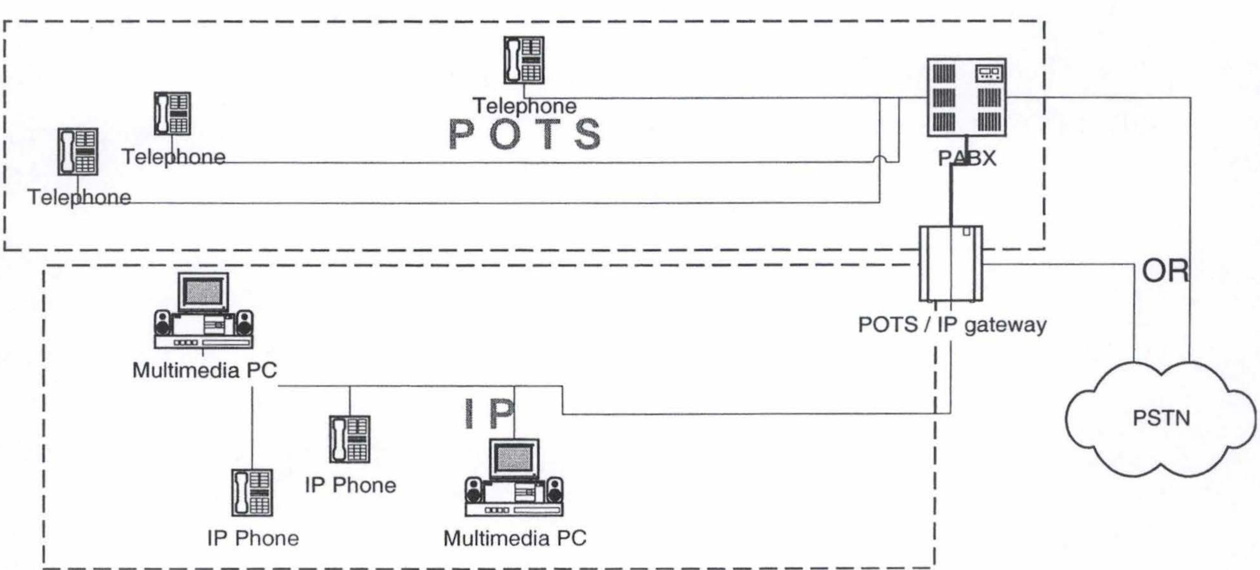

Figure 8 : gateway or PBX with PSTN interface

In this case, there are two different approaches, depending on the background of the manufacturer. We could have an IP-enabled PBX, by adding an IP interface card to a traditional PBX, or a voice-enabled IP router, by adding a voice card to a router. If the existing PBX is an IP-enabled PBX, it will also act as the POTS/IP gateway, and a single device will be used.

We won't argue about knowing which approach is the best, but the reader has to be aware of that distinction which can lead to different available features and ease of use. In both cases, the handling of any call on the private network should be completely transparent for the end-user. A problem about available features can however occur in such a case where a PBX and an IP gateway cohabit, due to the current development of VoIP technology and software which may lead to the lack of some feature on the VoIP network when compared to the its traditional counterpart. Moreover, the gateway and PBX have to be compatible: they need to understand each other and correctly translate the signalling language used (at either side a translation of signalling must occur, most likely at the gateway side).

2.2 VoIP as a means of interconnecting scattered sites

In this third scenario, illustrated on figure 9, voice over IP links are used to connect several scattered sites of the same organisation, without any interest for the local voice network inside each site. We have already told that there are potential savings to be made in using VoIP because of the possibility to use the same links for both voice and data traffics. The interconnection of sites is a perfect example of possible scale savings on communications.

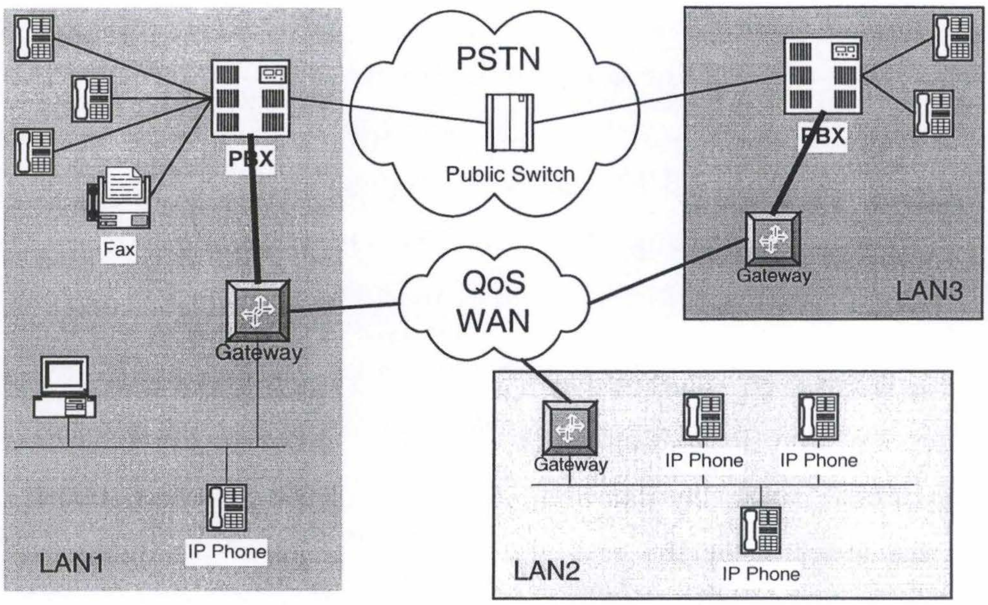

Figure 9 :VoIP to interconnect scattered sites

We could face two situations in this scenario: on the one hand, there could already be a data line between the sites, and voice calls could be placed via the PSTN leading to a heavy phone bill. On the other hand, there could be two parallel leased lines, one for data and the other for voice. The latter case will be met where the cost of the leased line is less than the cost of PSTN calls.

The growth in data traffic could require an update of the dedicated leased line, thus resulting in a cost increase. It could then be interesting to study the voice capacity requirements, and compare the price of a voice line and an IP line to determine if it would not be economically advantageous to lease a single IP line (or more accurately lease an IP-bandwidth capacity) for both kinds of traffic, instead of a costly voice line and a data line.

In order to assure an increased reliability to the voice links, a redundant connection through the PSTN may be appropriate. This connection should be used only when the VoIP WAN access is either out of order (congested network or hardware failure) or has reached its maximum capacity (a high priority call may be routed to the PSTN).

In the case of the same line used for both traffics, the equipment at the end of the line has to be a voice-enabled router, capable of handling any kind of IP traffic and to route it correctly within the local area network to which it is attached, whether voice is directed to a traditional PBX or on a VoIP LAN. This observation leads us to then next case.

2.3 VoIP on a wide area network (WAN), from IP to IP.

A merging of two of the above cases, namely cases 2 and 3, this case occurs when the whole voice network of the organisation is IP-based. Figure 9 is also used for this case, which could also be met in the context of an industrial complex or area, with a single data and voice network offered as attractive facilities to several companies that could choose this location.

Within a WAN, network design is not a negligible factor. In voice telephony, delay is critical for a enjoyable quality. VoIP devices each add a delay, when processing packets, or when translating codecs or signalling methods. In a well designed and privately-owned network, it is possible to carefully choose the VoIP equipment so that each device is directly compatible with its neighbour, thus reducing the delays due to codec or signalling translation. The number of hops in the network is also a delay-introducing factor, if for example the device has an interarrival jitter buffer and wants to reorder the packet sequence before sending the whole sequence further. This is a problem a VoFR networks, where each switch does that. Using IP, this should be done only by the device located at the receiving end of the link.

2.4 VoIP used to provide voice services to the home user

Carriers and service providers could also be interested in VoIP technologies when addressing the home market. VoIP can be an efficient way to use an existing IP backbone and provide both Internet access and voice telephony on the same link. This supposes of course a engagement regarding QoS management. We can suppose that the service provider knows its own network, and has enough capacity to offer a good quality.

A problem for home-targeted VoIP offers is the high cost of IP phones. We need here a basic phone, as cheap as any POTS phone. These phones are cheap because they're stupid: so we need stupid IP phones. Using specific protocols, as we will see them in chapter 3, this is possible. As with the PSTN, the intelligence of the network would be centralised in some devices within the network. VoIP to the home user is illustrated by figure 10.

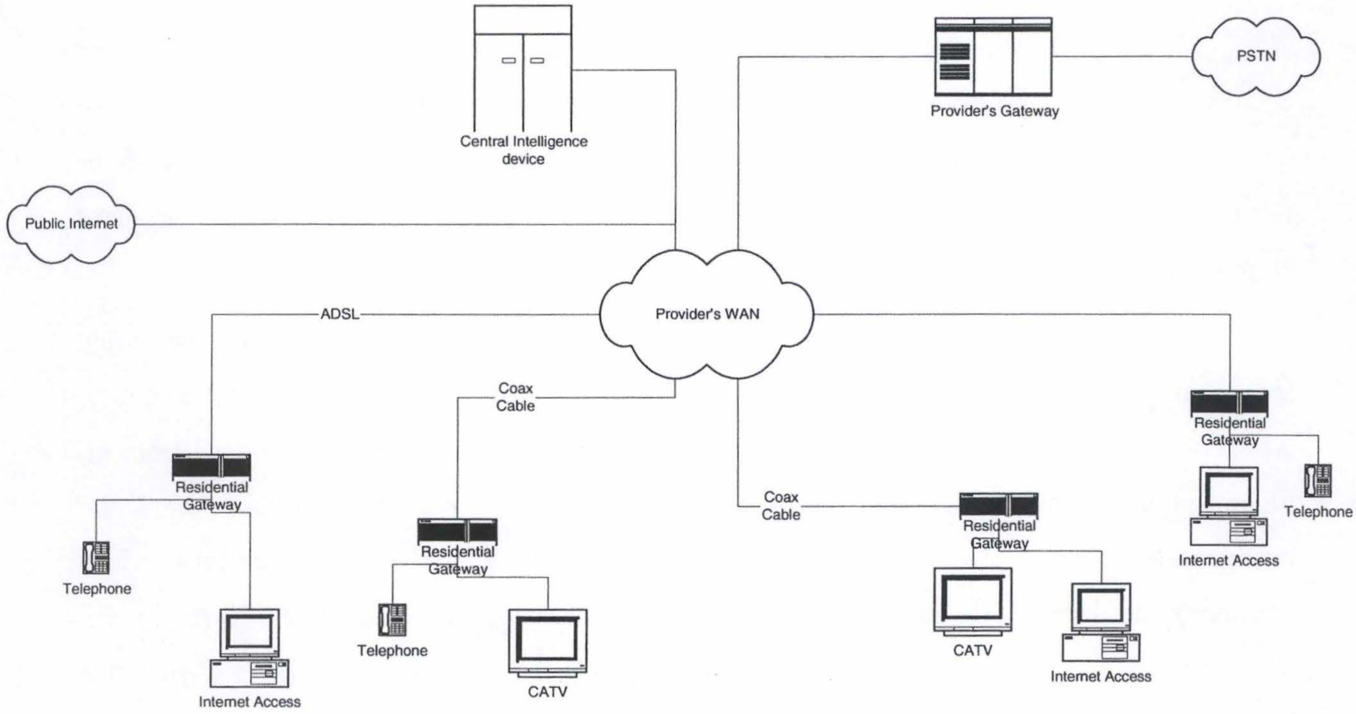

Figure 10 : VoIP going to the Home User

A service provider such as the Walloon WIN, relying on a high-capacity fibre optics network, could provide small businesses or home users with a telephony service based on a protocol allowing the use of cheap phones. The WIN S.A. acts as an Internet service and access provider, even offering ADSL solutions to the home users. Those high-speed lines are suitable for VoIP, and with compression the line could even be used for more than one communication, provided that the device is able to handle such features. Furthermore, ADSL lines are permanent lines, and their cost is an all-in price for a determined amount of data transferred. The pricing policy should be adapted to voice traffic.

Considering that the cost of an additional megabyte of data is 5 or 10 BEF (depending on the provider), and that one minute of communication with the standard G.729 codec at 8 kbits/sec yields a traffic of 193.3 KB/min[6], a minute of communication would cost 1.88 BEF (for 10 BEF per MB). The toll-quality G.711 codec (PCM) would yield a cost of 5.89 BEF/min[4]. Of course, those costs are based only on current ADSL. The problem of ADSL is that upload as well as download traffics are taken into account for billing. This can be a major drawback in the present situation where the calling party as well as the callee have to pay for the conversation. This shows that thinking of cheap VoIP calls because of the ADSL offer is not so straightforward.

It is also possible to deliver VoIP services along with cable TV. Using a special device acting as a private (residential) gateway, a service provider may be able to deliver cable television, internet access and telephony on the same cable. The use of the residential gateway not only does the splitting between IP and TV signal, but allows the home user to use its POTS phone, though removing the drawback of high-cost IP phones. This is the solution a Canadian company (Videotron) has chosen to deliver all communication services to its customer.

For small businesses or liberal professions, we could imagine a service integrating all communication needs over the same IP link, and even an interaction between the IP phone and the PC. Within a single package, voice and data communications could be offered at a minimised cost, thus leading to unified messaging for all.

2.5 VoIP on the public Internet

Within a privately-owned and managed WAN, there are few if any QoS problems. However, the use of the public Internet as a cheap means to carry voice telephony yields such problems. It is nearly impossible to tell what path the IP

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[6] If using PPP and RTP/UDP/IP without header compression, and if an RTP packet is sent every 20 milliseconds. The size of the RTP packet is 160 bytes with G.729 and 20 bytes with G.711. The header total size is 46 bytes. For one minute of half-duplex communication we need 3000 packets.

packets will take, and they could even take several different paths, leading to delay and interarrival jitter problems. A playback buffer can be, and shall be, on use at the receiving end, either a IP/POTS gateway or an IP-phone.

Voice quality highly relies on delay matters: for the human brain, a lost packet is less important that a delayed packet. It is commonly agreed that a delay of 150 msec should not be exceeded to keep an acceptable quality, and that this delay shall in no case exceed 400 msec. The problem of VoIP over the public Internet is the propagation delay. Coding delays are present in every VoIP network and are well-known.

If using the Internet as a cheap means to carry voice communications is known since almost the start of its existence, experiences are often disappointing. Such applications require a high quality of service level, as we have already underlined. The public internet cannot vouch for this. The diversity of devices, of bandwidth availability, of users etc makes it almost impossible to know in advance if a good conversation will be possible.

The fact is that toll quality can not be associated with cheap internet access. To deliver such a service, the provider must have the necessary bandwidth capacity, as well as some means to assure QoS features. Using the IP precedence classes can be a good idea. But this has to be done at each hop, and all devices along the path must handle that precedence bit correctly. The same is true for an RSVP bandwidth reservation. This means that investments are to be done to ensure the required quality level.

Besides, voice is often seen as a means to finance data networks. As data traffic grows, voice will probably be carried on data networks, and no more data on voice networks. The problem is not so relevant for our purpose. The relevant fact is that if no one is willing to pay the old price for voice, arguing that voice is now mere binary data, no means will be provided to ensure today's toll quality. That quality could be the premium service of tomorrow's data networks, and everyone knows that premium service has always its cost.

Chapter 3 : The protocols of VoIP

3.0 Introduction

Voice over IP is transport of voice over IP networks. Before going further, it could be useful to tell a word about computer networks. Such networks are of different kinds, in their physical or logical specifications. Both electrical and software solutions made different networks. Some of those solutions perform the same task: ethernet, token ring or token bus are all used on a local area network and all perform the same task, which is to carry information from computer to a linked computer. Those solutions are referred to as "data link layer" solutions.

The word "layer" is used to refer to a specific level of specialisation, whatever the implementation is. In a layered model, each layer uses the services of the lower rank layer, and provides services to the higher rank layer. The ISO has defined a seven-layer model for computer networks, from the physical layer (i.e. how to electrically transmit bits on several media, from copper wire to wireless transmission), to the application layer. That ISO model is named "Open Systems Interconnection", or short OSI.

The seven ISO layers are the following:

1. The physical layer, which describes how bits are to be coded on the used medium, from copper wire to wireless transmission. The main problem here is to insure that correct bits are received at the other end of the medium.
2. The data link layer, which describes how two local entities will communicate together, and here again it is simply a matter of bit transmission, the problem being to detect transmissions errors or collisions (when two entities send bits at the same time, those bits collide, and become irrelevant electrical signals).
3. The network layer, which has to control operations on a sub-network, and mainly the path that the transmitted messages will follow within that sub-network.

4. The transport layer, which has to take data, split it in smaller packets if necessary and ensure that they arrive to the recipient as efficiently as possible.

5. The session layer, which has to provide supplementary transport services, such as file transfer synchronisation when those transfer can be interrupted. This layer permits for instance to insert checkpoints in data flows, thus avoiding to transmit the whole file again when interrupted.

6. The presentation layer has to cope with bit semantics. It would e.g. translate a Unicode coding into an ASCII coding in use in lower rank layers, in a user transparent way.

7. The application layer to which high-level protocols belong. which have not to deal with transmission matters. Protocols such as POP3, HTTP, FTP etc belong to this layer.

This OSI model is a bit intricate and has never been actually used, but it is the reference model to design something else. The model in use on the Internet is called after its two main protocols, IP and TCP. The TCP/IP model has only five layers, i.e. the seven OSI layers without layers 5 and 6, as shown on figure X. On the internet, there are actually two transport protocols, TCP and UDP. UDP is a simple transport protocol which does not provide the reliability of TCP. We will see how this lack of reliability mechanisms can be useful, especially for voice transport.

We can now see that speaking of VoIP means 'voice over an internet-like network', whatever is the means used to carry the bits. Voice packets are actually not directly put in IP packets, but first in TCP or UDP packets. Though, that technology is called VoIP because the network characteristics are the most important, and those are the ones of IP.

3.1 Signalling protocols

As already several times above-mentioned, there could not be any communication without signalling, as simple this one can be. The signalling of the PSTN which is briefly described in section 1.1 has to be transposed to the IP world. To perform those signalling tasks, new protocols have been specified, by the ITU-T as well as the IETF: both worlds are involved in voice telephony, the traditional telecommunication world and the internet world. Signalling protocols are high-level protocols of the application layer. We will describe three signalling protocols in use for VoIP applications: H.323 from the ITU-T, SIP from the IETF and H.248/megaco from both the ITU-T and IETF.

3.1.1 The ITU-T H.323 recommendation[7]

a. Presentation of the H.323 recommendation

The H.323 recommendation has been defined by the International Telecommunication Union, Telecommunication standardisation sector committee (ITU-T). The H series of recommendation deals with audio-visual and multimedia systems, numbers from 300 to 399 are specific to systems and endpoint devices for audio-visual systems.

The H.323 recommendation has been approved in its first form in 1996, a second version in February 1998, and a third one in 1999. The recommendation is still a work in progress and the most recent document is the Draft v4 of May 2000. Version 4 is expected to be approved in November 2000. The original title of version 1, 'Visual telephone systems and equipment for local area networks which provide a non-guaranteed quality of service" was changed in version 2 to 'Packet-based multimedia communications systems' to reflect the extended scope of the recommendation.

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[7] Most of this presentation refers to the text of the 1998 version, which is the version currently implemented in commercial devices.

H.323 is not a protocol definition by itself, but more accurately a generic umbrella for several other ITU-T recommendations. In addition to H.323, we will find the following recommendations used by H.323:

- H.225.0 for basic messages definitions,
- H.245 for media canal control,
- H.246 for gateway operation mode,
- H.450.x for supplementary services,
- H.235 for security and encryption procedures.

The scope of H.323 is presented by figure 11.

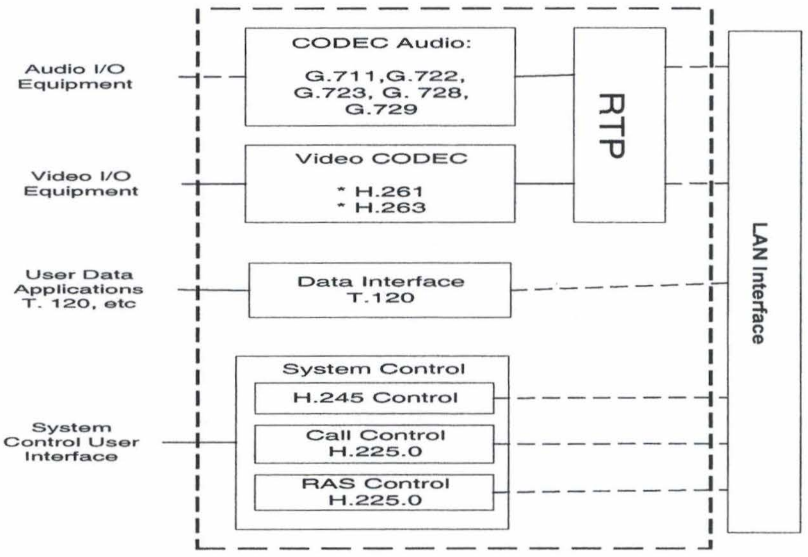

Figure 11: Scope of recommendation H.323

The signalling protocols which are used to set up the call are locate under the "System Control User Interface" heading, which is the main point of H.323. The other parts of H.323 deal with audio/video and data coding methods to use, thus making of H.323 a true multimedia recommendation.

As mentioned in the foreword of this part, we won't study video and data communication methods. The same specific transport protocol is used for video transport. We will focus on RTP later in this chapter.

Recommendation H.225 belongs to the transmission multiplexing and synchronisation recommendation series, and H.245 to the communication procedures series. Recommendation H.225.0 has the same grounds as the Q.931 signalling method used for ISDN networks. H.323 also describes interaction with the public switched network, as well as other signalling methods such as MGCP (in v4).

At least, and this is somehow important for H.323 implementers, all messages within the scope of the recommendation are to be coded following the ASN.1 binary encoding, which makes the reading of messages complex (it is impossible to read the messages in their actual form), and a specific parser has also to be developed. This can have an impact on development costs.

b. Call handling with the H.323 recommendation

H.323 is designed to operate on networks which may not provide a guaranteed quality of service. Though, it appears that the organisation of H.323 devices is best suited for LAN's or at least managed networks. The presence of gateways allows to connect several networks, and to use networks which do not comply to H.323, either with tunnelling or translation.

H.323 entities and their role

A H.323 communication may involve several entities, the exact number of which depending on the actual utilisation. Furthermore, some entities are mandatory and others are optional under the terms of the recommendation.

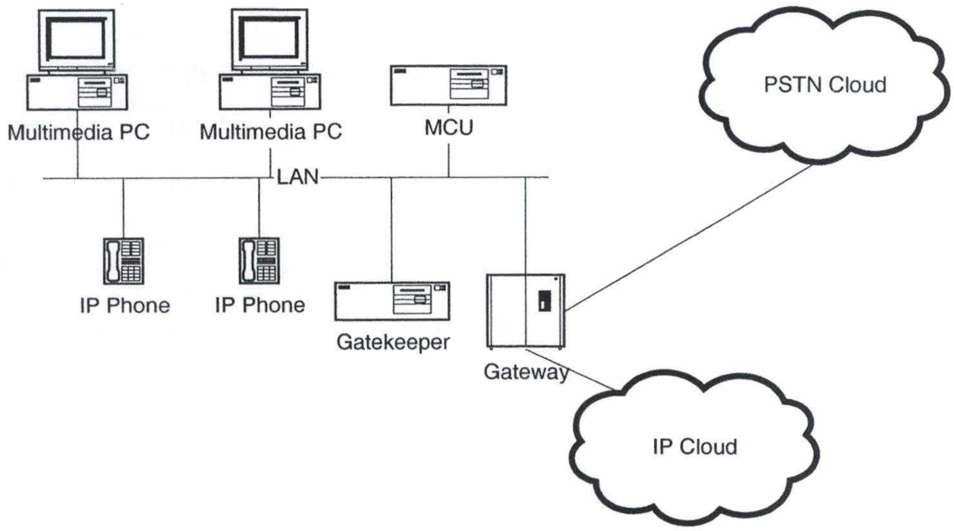

Figure 12: H.323 entities

Figure 12 shows the H.323 entities:

- **The endpoint**: a multimedia PC running, e.g., Microsoft NetMeeting or an IP phone directly connected to a LAN.
- **The gatekeeper**: an optional entity, which plays a major role in H.323 calls and their management. If the gatekeeper is present, has to offer several functionalities. Some of those are once again mandatory while some are optional. The gatekeeper brings up the concept of "zone", i.e. the whole group of entities under its control, whether on the same LAN or not.

  If present, the gatekeeper shall perform the following services[8]:

  — Address translation. The gatekeeper shall translate a call alias to a transport address, by using a translation table established with registration messages, or according to any alternative method.

[8] H.323 Draft v4 Recommendation, p43

— Admission control. The gatekeeper will authorise or not the call according to some predefined criteria, such as available bandwidth, caller address or any other choice of the manufacturer. It can also be a null function (i.e. all calls are accepted).

— Bandwidth control. The gatekeeper shall support bandwidth reservations made by endpoints. This can be a null function.

— Zone management. The gatekeeper shall provide the above functions for terminals, MCU's and Gateways which have registered with it.

- The gatekeeper may also perform other optional functions such as:

— Call Control Signalling (this being for further study)

— Call Authorisation. Through the use of H.225.0 signalling, calls can be rejected or accepted according to criteria outside the scope of H.323.

— Bandwidth Management: control of the actual available bandwidth, leading to accept or reject a call. This function should be used together with bandwidth control.

— Alias Address Modification. The gatekeeper may return a modified Alias Address. If returned in an ACF[9] message, that alias becomes mandatory for the endpoint.

— Dialled Digit Translation. The gatekeeper may translate dialled digits into an E.164 number or a Private Network number.

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[9] Authorisation confirmation. See lower.

— Gatekeeper management information data structure. (this being for further study).

— Bandwidth reservation for terminals not capable if this function. (this being for further study). *This would suppose the gatekeeper to anticipate the required bandwidth. It could for example be done on basis of the used codecs.*

— Directory services. (this being for further study). *This function can currently be performed by a LDAP server.*

The functions specified as being for further study show how signalling protocols are still under development, even if the case of this relatively old recommendation.

- **The Gateway**: the function of a gateway is to provide translation between transmission formats and between communications procedures. Translation between video, audio and data formats may also be performed by the gateway. Thus, the gateway is the door to any other network. It acts as an endpoint on both the networks it interconnects, endpoint meaning here terminal or MCU (see next point).

- **The MCU:** Multipoint Control Unit: the role of this entity is to provide support for multipoint (3 or more) conferences. Such an MCU is composed of one mandatory part, the Multipoint Controller (MC) and optional parts (i.e. zero or more), the Multipoint Processor. The MCU may be included into a gatekeeper or gateway, though it is only a matter of sharing the same hardware device.

  The MC controls the conference, and can connect two terminals for further extension of the conference. It provides parameters negotiation capabilities and ensures the same quality to all parties. It does not provide any audio or video processing capabilities. The MC is not an endpoint by itself, the only callable device is the MCU which it is part of.

The MP provides audio and video processing capabilities. The audio/video flows from all parties can be mixed into one single flow which is redistributed to all parties (video can be sent as an array of the different sources, or sent if there is an associated audio signal). The MP is not an endpoint by itself, the only callable device is the MCU which it is part of.

The steps of a H.323 call

We will now examine which are the different steps of a H.323 call. We won't here develop all existing possibilities, as their number is quite important. We will so examine the major steps of a H.323 call which are must always take place.

A first partition is given by the presence or the absence of a gatekeeper. Remember that this entity is optional, though its use is mandatory if present.

A second partition takes place when the gatekeeper is present. In this case, there exists two models to handle the call: whether the gatekeeper routes the call, or by using direct call signalling. This choice is made by the gatekeeper.

Along with those partitions, several hypothesis bring up an increased number of cases to consider: from the simple call involving only two parties to a multipoint conference. A conference can be managed by a simple MC, but this one can be combined with an MP, and there are several possibilities regarding how to combine those two devices, which can be centralised on decentralised. Furthermore, the call can take place on a single network, or within a single H.323 zone, which can be spread on several networks, or involve two or more zones, which means that the number of gatekeeper involved in the call may varies accordingly.

The gateway role will at first be concealed, knowing that its role is mainly to translate signalling protocols. That translation being transparent to end-users, it doesn't directly intervene it the call steps.

As it has just been told, the combinations of hypothesis make the number of cases to examine explode if we want to be exhaustive. The point here is to give an outlook of the H.323 operations, and we will restrict ourselves to the main cases from which it is possible to easily understand more complex cases.

Before going further, let's draw our attention to the fact that a H.323 call is always unidirectional. Therefore, each party has to establish its own communication canal to have a two-way conversation.

The progress of a H.323 call always involves the same steps, which are:

- call setup, which involves H.225[10] / Q.931 procedures using TCP,
- initial communication and capability exchange, which involves H.245 procedures using TCP,
- establishment of audio-visual communication, which involves RTP/RTCP using UDP,
- call services, which involves H.245 procedures using TCP,
- call termination, which involves H.245/H.225 procedures using TCP.

Those steps may follow an initial negotiation with a gatekeeper.

A. From terminal to terminal, without gatekeeper

This is the basic case: a terminal is trying to get in touch with a distant counterpart. Without involving a gatekeeper, no address alias service is present and this means that the caller has to use the exact address of the callee, in this case the IP address. The H.323 recommendation defines a TCP port to use when establishing the call,

[10] H.225.0 procedures are actually a subset of Q.931 procedures which are used in ISDN networks.

so that the caller knows the TSAP[11] to use. When the callee receives the caller's messages, it can itself initiate the reverse call setup in order to get a bi-directional communication.

B. Call with the intervention of a gatekeeper

If a gatekeeper is present, any call setup attempt shall transit through it. Two models are then possible to consider, as above-mentioned: the direct call signalling model and the call signalling routing one. The latter model means that the gatekeeper is involved in every step of the call setup, leading the gatekeeper to act as a proxy server. This does not change the steps to setup the H.323 call but change the flow of messages. For details about those flows, refer to the annex.

In a schematic way, the different steps take place as follows:

- the calling terminal shall proceed with its registration to a gatekeeper. This initial step is done with the ARQ message, leading the gatekeeper to send back the ACF or ARJ message. If ARJ is received, the gatekeeper doesn't accept the registration.
- The calling terminal may then proceed with the opening of the TCP connection on which the H.225.0 messages will be exchanged. The H.225.0 step will decide which TCP port to use for further operations. (H.245 negotiation). The H.225.0 setup message describes the call type (point to point, multipoint,...), security capacities, as well as a possible notification of usage of the *fastConnect*[12] quick procedure.
- The terminal proceeds then with the opening of the H.245 connection, with a capabilities exchange, the logical channels

[11] TSAP : Transport Service Access Point, which identifies the address of a service running on a computer. In the case of TCP/IP, the TSAP is the computer's IP address combined with the TCP (or UDP) port which the service is listening to. The initial TCP port for H.323 calls (more accurately the H.225.0 procedures) has been defined as port number 1720.

[12] See annex for more details about this procedure

opening and negotiation about which UDP port to use for the RTP connection.

- Once the terminal has opened the RTP connection, and the actual call is proceeding, it is possible to use the H.245 connection to renegotiate some call parameters such as requests for more bandwidth .
- To tear down the call, the first step is to close the H.245 connection, then the H.225.0 connection, and finally the disconnection is asked to the gatekeeper.

The scheme submitted above shows the relative complexity of H.323 processes which involves several other recommendation. The latter scheme does not involve more than one gatekeeper. If the called party is not registered to a gatekeeper, it could be a good idea to do so at that moment, but is not required to.

The initial negotiation phase with the gatekeeper allows the terminal to answer three important questions:

1. *Which is my gatekeeper?*

   The answer to that question is brought by an exchange of the 'gatekeeper request/confirm/reject' (GRQ/GCF/GRF) messages. The GRQ message is sent with multicast capabilities of the underlying network.

2. *Which IP terminal do I have to call?*

   The answer to that question is brought by an exchange of the 'location request/confirm/reject' (LRQ/LCF/LRJ) messages which involve the gatekeeper for alias resolution, which may be compared to the DNS in use on the world wide web.

3. *May I call?*

   The answer to that question is brought by an exchange of the 'authorisation request / confirm / reject' (ARQ/ARJ/ACF) messages which ask the gatekeeper permission to place a call. It can be refused if the available bandwidth is not sufficient (if the GK is able to perform bandwidth management), or for any other reason out of the scope of the recommendation.

Any 'reject' message is accompanied by a description of why the request has been rejected.

So, the H.323 recommendation involves other ITU-T recommendation, and the use of a gateway may involve more ITU-T recommendation such as H.32x, Q.931 for ISDN interaction. In order to provide interaction with the PSTN network, the gateway has to process SS7 signalling.

If we go back to the scenarios of chapter two, we see that in the most complex scenario involving LAN's, H.323 can be used to provide VoIP services. IP phones and multimedia PC's are H.323 compliant endpoints, and the POTS/IP gateway a H.323 gateway. This H.323 gateway can also be used as a means to connect scattered VoIP LAN's, either using VoIP links or any other network. The traditional functions of a PBX are then performed by the gatekeeper, which doesn't appear on figures used to describe those scenarios. Within a single organisation, whether using a single site network or several scattered sites having each their own LAN, the interconnection of which forming an organisation-wide WAN, a single gatekeeper can be used to manage all VoIP devices within the organisation. This can be useful if the organisation wants to have a single call center, but makes the network design somehow more complex, especially regarding delay problems.

3.1.2 The IETF protocol: Session Initiation Protocol, SIP

a. Presentation of SIP

The Session Initiation Protocol is a work of the IETF and was initially developed by Henning Shulzrinne from Columbia University and Jonathan Rosenberg. It is designed as a light and simple open protocol performing signalling tasks for multimedia applications, and as such belongs to the application layer of the TCP/IP protocol stack.

The current version of SIP is version 2.0 and is specified by RFC 2543. As implementation work makes progress, some problems have been risen, such as correct interaction with the PSTN. The development of SIP is still a work in progress and a new draft has been recently published to address those problems.

SIP is only designed to initiate any kind of multimedia session, between two or more participants using multicast capabilities of the Internet Protocol. It is only a signalling protocol, and is used with other IETF protocols such as RTP and of course UDP and IP to provide a true telephony service. The specification of SIP is a complete specification of how to perform signalling tasks, and so does not require the implementers to read many other documents. It is though required to be familiar with the Session Description Protocol (SDP), its format being used within SIP messages.

Deriving strength from its experience, the IETF has chosen to design SIP as a text-based protocol as HTTP is. The coding to use for text messages is the standard ISO 10646 in Unicode UTF-8. This has a double advantage: the design of parsers is easy to be generic (i.e. a simple text parser is sufficient), and anyone can read the content of the packets without involving any processing, which makes debugging more easy. To allow generic parser usage, SIP is designed with fixed-length fields, as variable-length fields would make the design of a parser more complex.

SIP has been designed to run in an environment larger than a LAN, with a design allowing the user to be very mobile. Any voice codec may be used, as none is specified as mandatory or optional in the specification. The codec to use for a session will be negotiated at the establishment of the communication. The text encoding of SIP and its structure of well-delimited fields make the IETF protocol quite flexible regarding the content of some of its fields which may be easily adapted for specific needs or further development.

Unlike H.323 procedures which use both UDP and TCP, SIP is designed to run on top of UDP. The choice of this unreliable protocol is driven by the drawbacks induced by the reliability mechanisms of TCP. Those are relatively slow to start (one of these is called "slow start"), and require several round-trip delays to be useful. For voice applications where real-time transport is more important that packet lost, this a not a good thing. Even if SIP deals only with signalling, which is not so sensitive to delay, the quicker things are the better it is. It is why SIP asks for UDP and has the required mechanisms to ensure that messages are effectively delivered and acknowledged. It may of course be used on top of TCP if extra reliability is needed. Besides, support for both protocols is required for proxy, registrar and redirect servers (see lower). Users agents should implement both UDP and TCP.

b. Call handling with SIP

So, how does SIP work to initiate the telephony session? It is what we will examine in this section. As it was the case for H.323, we will first have a look on the entities specified by the protocol, and then see how they interact to provide telephony services to their user.

The SIP entities and their role

SIP defines five types of entities, as follows:

1. The **User Agent Client** (UAC). The role of this entity is the one of a client application which initiates the SIP request.
2. The **User Agent Server** (UAS). The role of this entity is the one of a server application which contacts the user on SIP request reception and gives back the answer received from the user. This answer is the acceptation, rejection or redirection of the request.

   In practice, it is obvious that the UAC and UAS will be the two sides of the same SIP application. A SIP connection is unidirectional, from UAC to UAS. A true conversation is bidirectional, and requires both the UAC and the UAS at each end of the line.

3. The **Proxy**. This one is a intermediate entity which acts as UAS as well as UAC in order to make requests on behalf of other clients. The requests are processed internally or by passing them to other servers, and this can occur after translation. A proxy interprets and, if necessary, rewrites a request before passing it further.

4. The **Redirect server**. The role of this entity is to accept SIP requests and match it with zero or several news addresses which are given back to the client

5. The **Registrar server**. The role of this entity is to handle 'REGISTER' requests. It will typically be combined with a proxy server or redirection server and may provide localisation services.

   The Registrar may also be used to authenticate users on the SIP network, by means of a challenge included in SIP messages.

   This entity will probably be located on the same hardware equipment as the proxy server, because they may perform some associated functions.

Those entities may or may not be on the same network, or at least the same LAN. It is to note that the role of the SIP entities is not defined in the same way as H.323 entities are, and so that direct comparison is hard to be done. It is though possible to compare functionalities offered by those entities and so find some similarities.

A gateway to the PSTN is out the scope of SIP v2.0 as defined in RFC 2543. Such as gateway should be able to act as both UAC and UAS on the SIP network, and translate the PSTN signalling to SIP, and back.

How SIP works – the different steps of a SIP call

As we have already mentioned it, SIP is a text-based protocol which has been designed in the HTTP way. Not only does it share with the latter the text encoding, but it also shares the class messages. Within those classes, we will point out the so-called final classes, which put an end to a SIP transaction.

The SIP classes are the following ones[13]:

- 1xx: information on the on-going request: request received, continuing to process the request;
- 2xx: success: the action was successfully received, understood, and accepted;
- 3xx: redirection: further action needs to be taken in order to complete the request;
- 4xx: client error: the request contains bad syntax or cannot be fulfilled at this server;
- 5xx: server error: the server failed to fulfil an apparently valid request;
- 6xx: global failure: the request cannot be fulfilled at any server.

Classes 2xx to 6xx are final classes.

In addition to those message classes, SIP relies on 6 main requests or "methods", which are in turn combined with some 72 headers. The methods are the following ones:

- INVITE. This is the initial message which initiates a parameter negotiation phase Those parameters are carried according to the

[13] See Annex for more details about the valid codes

rules defined by the Session Description Protocol (SDP, RFC 2327).

- BYE. This method brings the call to an end.
- CANCEL. This method allows to cancel a parallel search which may be initiated by a redirect server having given several possible addresses for the SIP number. Those parallel searches are to be stopped as soon as one of the called UAS has given a "200 OK" message.
- OPTIONS. This method allows to ask a server to announce its capabilities.
- REGISTER. This method is used to register a user on a registration server (registrar).
- ACK. This method is used to acknowledge a final response given to an INVITE method.

Among the 72 SIP headers, the 'VIA' is an important one in the SIP processing. Indeed, each proxy on the path of the call inserts this header in the SIP packets, so that the path followed by the packet is easily identifiable. This mechanism makes it possible to avoid loops in the path followed by the packets by examining the list of successive VIA headers: if the same VIA field appears two or more times, there is a loop somewhere in the SIP network.

We can now see how all these elements (entities, message classes and methods) play in SIP exchanges. As we did when examining the H.323 steps, we will focus here on basic cases which are the grounds to more complex situations. We will remind the reader that the topic of this part of the paper is to provide an overview of VoIP protocols and interest, and not to provide an extensive study of one protocol which would require a whole paper by itself.

In the same way that there exist three possibilities for a H.323 call, either direct from endpoint to endpoint, or through a gatekeeper and in this case either gatekeeper routed or direct signalling, SIP gives three possibilities to call someone.

The first one is a direct call, from User Agent to User Agent. The second possibility is to use a proxy server, and the last one to use a redirect server.

We will first the basic call model of SIP, which does not involve any particular server. This shows the basic steps in establishing a SIP session.

In direct calling, the initiating UAC has to know the address of the UAS to reach. The message exchange is as follows:

- The caller UAC sends the INVITE method to the UAS of the callee, describing which codecs it is willing to use and the RTP port to address.

  This INVITE method contains the following information (header fields):

  - From: identification of the caller (SIP address),
  - To: identification of the callee (SIP address),
  - Via: routing information, as already described here above. In the case of a direct call, this field will be empty, no proxy being involved in the message handling.
  - Call-ID: unique identification number for the call. This number will be included in all subsequent SIP message exchanges which are related to this call.
  - Cseq: sequence number inserted in some SIP messages which require an acknowledgement. This sequence number allows for example to handle duplicate messages.
  - SDP: session parameters description according to SDP rules.
- The callee may send a progress report with a "100 TRYING" message, though in direct call this is unlikely
- The callee sends a "180 RINGING" message to indicate that the phone is ringing (softphone or real phone)

- When the called user has accepted the call, his UAS sends a "200 OK" message to the caller party.
- The caller UAC answers with an ACK method which may carry final parameters for the call according to the SDP format. Those parameters may result from what the called party is able to handle (according to its own parameters). If none is specified, the parameters to use are the ones specified in the original INVITE message.

  There is no answer to an ACK method.

At this point, the RTP connection is established on ports given by the parameters exchanged in previous messages.

- The call is brought to an end by either party by the sending of a BYE method. The other party sends a BYE too to terminate the call in both directions.
- Those BYE messages are acknowledged by sending of a final "200 OK" message.

This is the basic call model, without any particularity. Things are a bit more intricate in the case where a proxy server is involved.

This case is similar to the H.323 gatekeeper-routed call model. Indeed, the proxy will acts as UAC on behalf of the "real" caller UAC. It will also proxy any other status messages which are needed by the caller party (classes 1xx, 2xx, 4xx, 5xx, 6xx). The implications of a DNS-like mechanism for address resolution is possible. As already mentioned here above, each proxy on the path between the caller and the callee will inserts its own 'VIA' header in SIP messages, with the described utility.

When a proxy server is used, the "100 TRYING" message is useful because the proxy does not know if its own request will be accepted, proxied some more times, or refused. Call-ID headers have also their utility as they will be used to correlate messages at each proxy involved in the call (a proxy is likely to handle more than one call at the same time). Other header fields such as Cseq will also be useful in this context.

As with H.323, once the media flow is established the proxy is no more involved in the call, except for further message exchanges. There is though a difference when compared to H.323: with SIP, the proxy does not need to keep TCP connections open for the whole call duration as it is the case in H.323.

The third call model of SIP involves the redirect server. On reception of an 'INVITE' method, the redirect server will answer with a list of possible alternative locations where the user may be reached. Those locations will be included in a "301" or "302" code message.

In the case where several locations are returned, the caller's client may proceed with parallel connection attempts. In this situation, on reception of the "200 OK" message from one of the called destinations, a 'CANCEL' method has to be sent to each other called destination to cancel those useless attempts.

A proxy may handle those "30x" messages without telling it to the UAC on behalf of which it acts, and return a single "100" message to it, thus making the redirection transparent for the caller.

During a call, SIP messages can still be exchanged to change some parameters of the call with an "INVITE" transaction. For instance, putting a party on hold will be done by sending and INVITE message whose destination address will be set to zeroes.

3.1.3 Media gateway control protocols

a. Presentation of the protocols

The two protocols we have looked at in points 3.1.1 and 3.1.2, namely H.323 and SIP, are protocols using a distributed intelligence strategy. This approach means that all the network and call handling intelligence is shared by the protocol entities, endpoint or server. This way of doing things may be interesting is some situations as LAN telephony, but it is not when endpoints have to be cheap and widely available. It is why protocols using another strategy have been developed.

Remember that is the PSTN, the intelligence resides within the network, in signalling devices. When using the North American SS7 protocol, where switches are controlled by STPs, this is clearly visible. The same architecture has been thought of for VoIP applications. A central intelligence device is used to control media gateways which may be compared to Class 4 switches (though this comparison is quite rough).

There are currently two protocols using that architecture in the VoIP world. One comes from IETF working groups and is named Media Gateway Control Protocol (MGCP). It is an evolution of two other protocols, SGCP and IPDC and is specified in the IETF draft. The second of those protocols is a common work of the IETF and the ITU and as such has two "names": the IETF one, which is megaco, and the ITU one where it is known as recommendation H.248 which has just been recognised as a standard. If megaco seems to be an evolution of MGCP, there are slight differences between the two protocols, and MGCP is the one currently found in commercial solution. It is likely that megaco will appear in the next few months. We won't go into the subtleties of each protocol. As for H.323 and SIP, the purpose here is to give a brief presentation of each protocol and not a detailed guide for an implementation, which would be a mere commented copy of the specifications.

Those protocols describe only the procedures to use between gateways and the gateway controller and do not tell anything on how those devices have to be implemented. A trunking gateway would be required to handle other signalling methods, such as H.323, SIP, Q.SIG, Q.931, ISUP, SS7 where applicable. This is out of the scope of MGCP – H.248/megaco.

Once again, the protocol used to carry voice packets on the IP network is RTP. Those protocols are also ASCII text-based and use the SDP structure to carry their signalling information. The H.248/megaco specification also allows binary encoding in ASN.1 format.

b. Call handling with this protocol family

Media Control Protocols Entities

The entities specified by media control protocols are actually not numerous. Their purpose is to define a central intelligence device which controls gateways which have in turn the task to establish voice communications, thus separating the signalling trunks from the voice trunks as it is done with the PSTN SS7 North American architecture. Basically, we get two kinds of entities:

- **Media Gateway Controller / Call Agent**[14]: The MGC or Call Agent -according to the protocol used- is the central intelligence entity of the network. Its role is to control the gateways under its responsibility. It orders them to establish or tear down calls and tells them how to interpret signals from the endpoint (the phone). It also monitors the gateways.
- **Gateway**: the gateway is the entity which provides connectivity to either a end-user device (we will then call it a residential gateway) or another network, such as an SS7 or ISUP signalling network. Media gateway is the term used to designate a voice gateway. Voice and signalling gateways will be part of the same device, though they have different functions.

Figure 13 illustrates the architecture of such a central intelligence VoIP network.

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[14] MGC : H.248/megaco denomination ; Call Agent : MGCP denomination

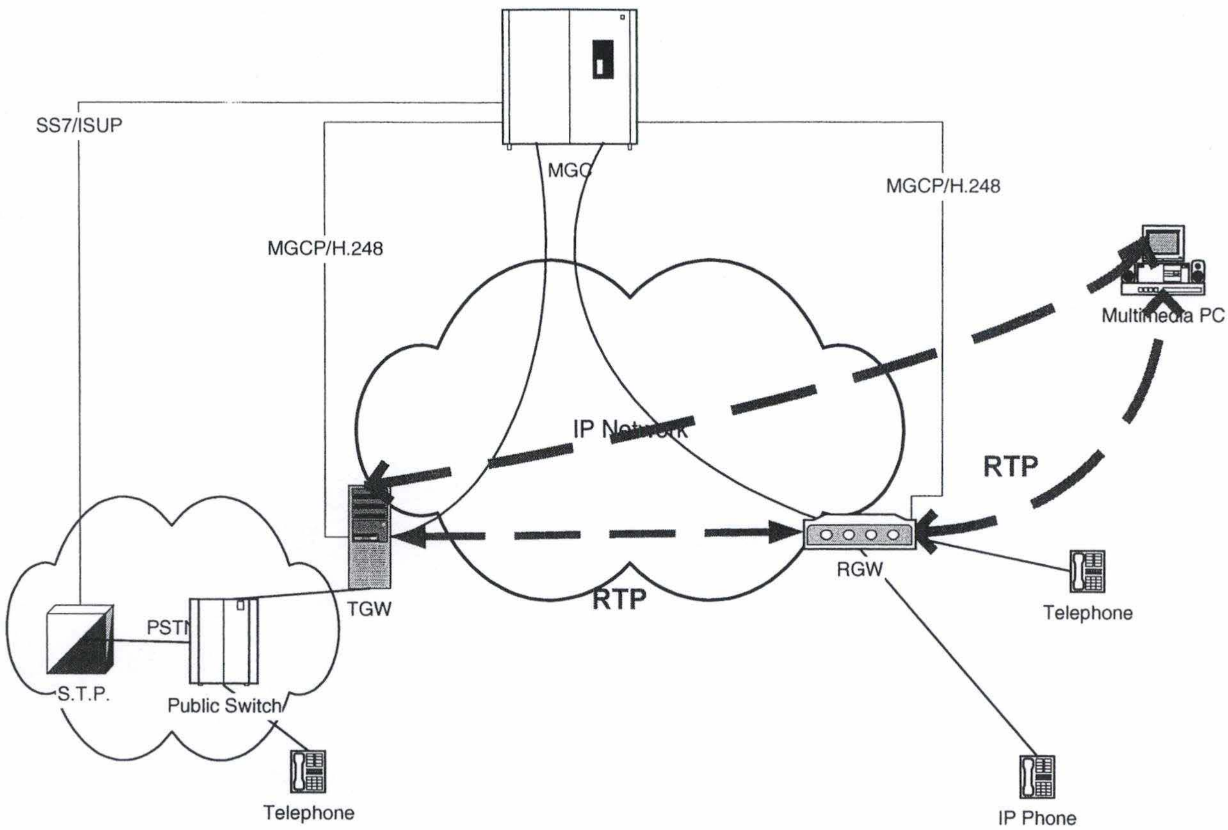

Figure 13: Media Control Protocols Architecture

On that figure, we see that a *Trunking Gateway* (TGW) is used to provide a connection to the PSTN, which would probably also involve a codec conversion to and from the PCM in use within the public network. The *Residential Gateway* is a gateway used to connect a home user to the VoIP network, either using an IP Phone or a POTS phone. As the broad arrows show it, the RTP packets flow directly from gateway to gateway, without going through the MGC or Call Agent. The latter is linked to the gateways it controls and not to the endpoint (as an H.323 gatekeeper would be)

How Media Gateway Control Protocols work

The protocols work with message exchanges between gateways and the gateway controller, as we may expect it. Schematically, the following procedure is performed to create a connection. Exact command names will be omitted in that description, as those differ between MGCP and H.248/megaco. The underlying idea is common.

1. The central intelligence device asks the first gateway to create a connection on the first endpoint. The gateway allocates resources and sends back a session descriptor (or error message) to the MGC. This session descriptor contains all relevant information about the connection, such as ports to use (UDP when using MGCP, either UDP or TCP when using megaco), IP address, codec).
2. Having determined on which terminating gateway the connection has to be made, the MGC orders that second gateway to create a connection to the called endpoint, with the session description provided by the first gateway. The second one returns its own description to the MGC, and allocates resources for that connection.
3. Once both gateways have agreed on connection parameters (i.e. once the MGC has sent parameters from one gateway to the other), the connection can be established between the two gateways and the RTP flow unleashed. This require of course that the called party has effectively answered so that this RTP connection is required.

During the course of a call, its parameters may be changed by the MGC, and this one can decide to request the gateways to send some information about the call and the current state of the gateways, by issuing a request for audit command.

The procedure described here above is actually more complex and each step requires several messages from and to the MGC. It is the MGC which analyses the dialled digits to tell the gateways where to route the call and to provide appropriate tones to the endpoint (ring back, ringing, busy…), while specific package definitions are required to add real-world functionalities to the core protocol. Such packages define for example how to deal with DTMF signals, how to deal with RTP, and how to supervise POTS lines.

If the originating and terminating gateways are not connected to the same MGC, there is need for signalling exchanges between the two MGC's involved in the call. This is similar to the hierarchical structure of the PSTN (If the two parties are connected the same CO (Class 5 switch on figure 1), this one is able to handle the call directly. If it is not the case, a higher level must be implicated (Class 4 switch on figure 1), and so on)

Using IPSec, some security may be added to the protocols exchanges by requiring authentication from gateways and possibly encrypting the voice packets. This would require a key exchange. That security scheme does not prevent from attacks directed at the gateways (either sending garbage voice to the listening port or a denial of service attack). To avoid playing garbage packets, the gateway could check where they come from and match the originating address with the one specified in the original session description received from the other gateway.

3.1.4 Comparison of the VoIP signalling protocols

The H.323 recommendation, SIP and MGCP-H.248/megaco protocols are all related to VoIP applications and signalling. It should now be clear that the first two solutions belong to the same family, while the third way is another one. Indeed, the two first ones have a distributed intelligence, while the third one relies on a central intelligence device.

Functionalities associated with the protocols are also different: H.323 and SIP provide many "intelligent network" functionalities by the use of the gatekeeper or SIP servers. MGCP and H.248/megaco are designed to perform signalling tasks between two gateways. Their purpose is clearly not the same.

When looking at their scalability, it appears that SIP and its light specification is more scalable than the heavy H.323 which would be recommended for use on a LAN. The mobility features offered by SIP are quite useful in that scalability, though the use of the redirect/registrar servers may induce a larger delay in the call establishment if many interrogations of servers are required (but is a native way of supporting call forwards). The advantage of SIP is that it supports multi-party conferences without any additional devices such as the MCU of H.323, and that it is able to try several locations at the same time, so that a follow-me function is natively supported by the protocol. Media control protocols are not concerned by such functionalities as their purpose is not the same. The H.323 draft v4 deals with H.248 interaction which may be useful in a generalised VoIP network where H.323 gateways are controlled by an H.248 MGC.

This shows that H.323 and SIP are more suited to organisations private network which would require PBX functionalities, while MGCP and H.248/megaco are more suited to companies wanting to offer VoIP services to the home user.

3.2 The annex protocols

If signalling protocols are an important part of VoIP protocol stacks and require a specific gateway when interconnected networks have made different choices regarding that problem, there are some constants for all VoIP networks. The first of them, and the most visible one is the common use of RTP for voice packet transport. We have also seen that both SIP and MGCP/Megaco-H.248 use SDP to carry the call parameters. At last, there is also the problem of the quality of service which may be solved by using RSVP.

A presentation of the VoIP protocols would not be complete without a word about those protocols. We will first take a look at RTP without which voice transport would be problematic. We will then take our interest to SDP and finally RSVP.

3.2.1 The specific transport protocol of the VoIP: the Real-Time Transport Protocol and its control protocol RTCP

Real-time transfer protocol

The Real-time Transport Protocol (RTP) has been developed within the IETF and is specified by RFC 1889 (version 2), but further work is in progress. It is indissociable from its associated control protocol, the Real-time Transport Control Protocol (RTCP) along which it is specified.

A characteristic of RTP to underline is that we could compare the RFC 1889 to an outline law: without decree providing for its enforcement, it is useless. Therefore utilisation profiles, as well as payload definition, need to be added to the core protocol to obtain something functional. The RFC 1890, entitled "RTP profile for A/V conference with minimal control" is used to define the payload used with VoIP applications. This shows that the RTP specification is open, thus allowing for further evolutions.

The RTP does not offer any kind of guarantee of service, and does not suppose the lower layer protocols to do so. It only provides a sequence number which is useful for the good ordering of packets, as well as a timestamp giving the generation time of the packet. It also gives a unique random identification number to the whole RTP session which identifies the packet sender, and if in a multi-party conference, all sources having contributed to the packet content. This last number is called the "synchronisation source" field. If other sources have contributed to the packet content, which means that the packet is the result of mixing several sources, their respective synchronisation number are included in the RTP packet header in a specific field.

RTP is used with UDP, on an even port number. The next port number (odd) will be used for the RTCP connection. The RTP may be used on a multicast network, using IP multicast capabilities.

Real-time transfer control protocol

RTP has its own control protocol, RTCP, which monitors the quality of the RTP connection. In order to perform this task, RTCP defines message exchange between the source and the destination endpoints:

- "sender report" packet: this packet, which is sent by the source, provides information about the amount of data really sent and allows a synchronisation between different packet types, such as audio and video packets. This is done by providing a synchronisation means between the RTP timestamp and a global clock.
- "receiver report" packet: this packet, which is sent by the destination, provides information about incurred losses and arrival delay of packets, as well as the timestamp of the last received packet.
- "source descriptor" packet: this packet makes it possible to control the RTP session, and especially to solve possible problems of identification number chosen by RTP (this random number is a 32-bit number, but conflicts are always possible). This conflict resolution will be made on basis of a canonical name, the

CNAME, which is carried by RTCP. This text name is a unique description of the packet sender.

RTCP is by nature bandwidth-consuming, bandwidth which is a scarce resource within an IP network, particularly when dealing with real-time transport constraints. It is the reason why RTCP includes a mechanism which allows to control the consumed bandwidth. The underlying principle is quite simple: the more RTP requires resources, the less should RTCP consume them. A proposal has been made in an IETF draft that RTCP should not be allocated more than 5% of the session bandwidth, of which 1.75% for the senders and 3.25% for the receivers.

Compressed RTP

The RTP header is 12-byte long. When combined with the 20-byte header of IP and 8-byte header of UDP, this yields a 40-byte long header. When we know that a voice packet is sent every 20 milliseconds and that the standard G.729 codec has a debit rate of 8 kbps, yielding a payload size of 20 bytes, we see how the RTP/UDP/IP header is a major overhead. There are several fields in that combined payload which are either constant or evolve following a constant scheme (we will call this the second-order difference).

Fields for which the first-order difference is zero may be omitted from time to time, as well as fields for which the second-order difference is also zero. Some other fields evolve in an unknown way, while other change all the time (such as the UDP CRC).

This establishment of facts lead to think to a way to reduce the overhead of the header, and is described by RFC 2508 *Compressing IP/UDP/RTP Headers for Low-Speed Serial Links.* The idea is to transmit only the differences between packet headers, and nothing when the first-order difference is nil. That way, the total header can be reduced to 4 bytes (and even 2 when not using the UDP CRC) most of the time.

A context header is regularly transmitted to insure that the compressed header is still relevant (for instance, a nil second-order difference may be false due to packet losses).

The use of CRTP is an important factor in the QoS battle: instead of carrying 60 bytes for 20 msec of voice, the network has only 24 bytes to carry, to which context packets must be added.

3.2.2 The SDP protocol

The Session Description Protocol has been designed for use on the M-bone in conjunction with the Session Announcement protocol and is specified in RFC 2327. It is a text-based protocol, as are the other IETF VoIP protocols. (ISO 10646 character sets in UTF-8 encoding – ISO 8859-1 allowed for compact representations).

It has been designed to provide relevant information on sessions, and such sessions could be VoIP sessions.

Standard SDP packets include the following information:

- Session name and purpose
- Time(s) the session is active
- The media comprising the session
- Information on those media (addresses, ports, formats and so on)
- Information about the bandwidth to be used by the conference
- Contact information for the person responsible for the session.

The last two parameters are not always required but maybe useful. The media information includes relevant parameters such as the type of media – video, audio,..–, the transport protocol and format of the media –the codec used–.

Each SDP descriptor is structured in the same way, i.e. <letter>:<description>, where <letter> is a case-significant character telling the parameter described by <description>[15].

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[15] SIP message example in the Annex shows the SDP payload structure

All this is relevant for VoIP sessions, and it is the reason why SDP which is already a standard has been chosen to carry SIP or MGCP/megaco parameters.

3.2.3 A QoS protocol: RSVP

As we have already told it, the best-effort service of IP is not satisfactory when carrying voice packets which belong to a real-time telephony session. If some mechanisms implemented in the network routers may be useful to address the QoS problem, there is also a protocol dedicated to this.

The resource reservation protocol has been designed to give internet users a means to ask for the bandwidth they need to carry their data without undergoing congestion problems. It is an IETF protocol specified in RFC 2205, but its usage is not generalised in the Internet and some routers are not able to understand its messages.

The principle of RSVP is to set up a temporary path in the network, on which the data will flow, on request of the receiver party (this to allow efficient multicast support). This path must be renewed every thirty seconds to ensure that the reservation is still needed, as the requester may have crashed.

We will now roughly see how RSVP works.

The source of the IP flow send a "PATH" message which indicates its traffic characteristics. The destination analyses this packet and decides how much bandwidth it will reserve in the network (it may ask for less than announced by the source), and sends back a "RESV" message which has to be handled at each hop on the path to the source: it would make no sense to reserve resources at some hops only.

The reservation has to be made by the receiver party in order to provide an efficient support for multicast transmissions, because it is this party which knows what are the different destinations of the multicast flow and thus can aggregate all bandwidth reservation demands.

This scheme also means that the reservation is unidirectional. For VoIP applications, it means that both source and destination endpoints have to make RSVP reservations, which can add delay to the call establishment. It is why it is appropriate to synchronise the RSVP and signalling messages. But this has also as a consequence that we can not be sure that the two reservations will succeed as they are fully independent: the reservation may be accepted or rejected by any hop in the network according to the policy they implement.

Another major drawback of RSVP is its lack of scalability (a core network router may have to handle several thousands of RSVP messages and require quite a lot of CPU power), which relegates RSVP usage to small networks only. Furthermore, RSVP works only with the IP packet and does not account for any compression scheme such as CRTP or further header overhead (such as the PPP header). This means that RSVP will ask for more bandwidth that really needed when using a compression scheme, and thus allow for less communications than possible.

In conclusion, we can say that RSVP is a tool in the QoS battle, but is not the almighty answer. It has to be used in some well thought situations and carefully configured. It may sometimes be more efficient to use only the ToS IP field.

Conclusion

Over the past twenty years, telephony has evolved from an analogue world using copper wire circuits to digital circuits over a digital TDM network. Today, with the increasing demand for cheap data networks, new solutions emerge to replace the costly data over voice network transport. Those solutions lead to think of carrying voice over the new data networks. But the replacement of the high-profit PSTN by large telephone companies will probably not occur soon.

Instead, private organisations may well think of data networks to provide for their communication needs over private or virtual private networks, as new companies may develop their own broadband data network to offer telephony services to the home user, thanks to the lost of the old telephony monopoly.

Today, IP seems to be the more flexible and democratic protocol to enable voice over data networks services. IP has its own drawbacks, but there exist several mechanisms to reduce them by managing the QoS levels of differentiated data flows. The demand for voice products taking advantage of the existing IP intranets and extranets leads major vendors to offer voice products to what has become a profit-making market.

VoIP solutions may use three different protocols, each having their advantages and drawbacks. H.323 and SIP are well suited for LAN and WAN intelligent communications, while MGCP-H.248/megaco is well suited to offer VoIP services to the home user.

Today, H.323 rules the VoIP market, and is the de facto standard. But we begin to see SIP products, as well as MGCP networks. It would require an entire study to tell what is currently on the market and what is soon there. Any major vendor like: Cisco, Lucent, Nortel Networks, Oki, Nokia, Siemens, Alcatel, ... has taken the way of IP. As we see, IP merges the world of data and voice transport. The first names in this short list are well known on the data market, while the last ones are well-known in the traditional PBX world. It is also quite hard to distinguish between what is still marketing department promises and real offer. But one thing is quite sure: nobody wants to miss the IP hype, and everybody seems convinced this is the future of telecommunications.

VoIP is not only a means to place a phone call, but also a means to go further with the Computer Telephony Integration and achieve a true unified messaging. With the emergence of IPv6 and the new addresses it will provide, coupled with the use of mobile IP, we could even imagine that mobile phones will become mobile IP phones. It would so be possible to receive one's e-mail directly on the phone, without even having to log on to the corporate network and use a unified messaging application. But this is not yet the reality.

PART II.

Introduction

In the first part of this work, we have envisaged several cases where VoIP technologies could be used. Those cases were developed in chapter two[16], from the LAN to the public Internet. All of these have different economic backgrounds and thus lead to as many reflections.

This work's second part will focus on LAN telephony which is described by figure X. Our purpose here is to propose an economic model of the deployment of a VoIP network within a LAN. It should help its user to determine if it is interesting for him to deploy such a network when compared to a traditional PBX-based solution.

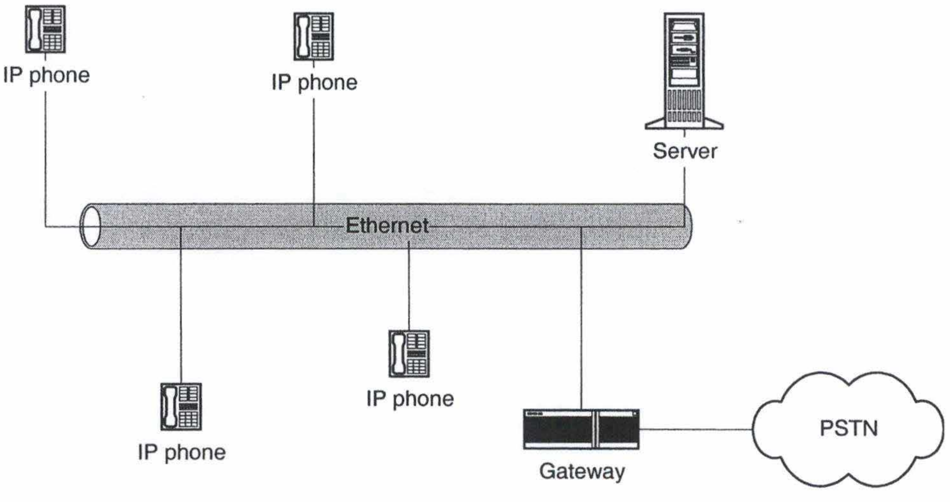


Figure 14: The scope of part II

We will first examine the elements to consider when thinking to a VoIP solution, elements which are not only bound to telephony functionalities but also to specific characteristics of data networks: the environmental parameters need to be

[16] see page 13

carefully studied as they determine the design of the network and the choice of the corresponding equipment. This in turn has a direct influence on costs.

Having shown the elements to take into account when evaluating the global cost of a VoIP solution, we will then propose a way to structure them in a model of which the final aim is an economic evaluation, but also to help the user to think to all necessary questions.

REM: In this part, the contracting authority of the telephony solution will be called "the user"

Chapter 5 : Elements of the model

The aim of this chapter is to determine the questions to ask when envisaging the way of IP for the telephony needs. Those questions are organised on two axis.

The first one is the technical axis which will lead us to examine which are the functionalities offered by the telephony solution. Those technical aspects include the awaited functionalities offered to the end user, as well as the VoIP-specific quality and performance problems.

The second axis is the one of environmental aspects which determine the equipment to choose when designing the network. Those parameters will be presented in two groups, at first the internal environment parameters (where the network will be deployed) and then the external ones (what is outside the network but need to co-operate with it).

Those so defined two axis will form a group of constraints which will have to guide the user towards the best possible choice, either IP-based or PBX-based.

With this aim in view, an evaluation variable will be associated to every parameters presented herein. This variable has for purpose to allow the user to compare the diverse solutions which are offered when facing the same problems. The next chapter will show the use of those variables when evaluating the cost of the solution.

5.1 Functionalities

The list of functionalities that can be offered by a telephony network is quite large. According to the expectations of the prospective user, some of these functionalities will be considered as essential or on the contrary useless.

Therefore we propose to attach a subjective mark to each of the functionalities, mark that will reflect its importance for the end user. The sum of these marks will be used to decide between several competing offers: it will be the evaluation variable associated to the functionalities parameter.

The VoIP technologies lead here to a problem which is related to their youth. Indeed, the functionalities that are offered by such a system are above all dependent on the used software. As for any new software application, successive versions are released on the market and offer more and more possibilities. In the same way, the standards which they are compliant with, either the relatively old H.323, SIP or MGCP – H.248/megaco, evolve together with the problems encountered during their implementation and the needs expressed by the end-user market.

Consequently, it appears important to draw the attention of the user to the fact that functionalities not present today may well be offered tomorrow and even by a simple update or upgrade of the software. A completely new version of that software may also turn out to be mandatory, and this will have a repercussion on the maintenance cost. In this context, it will be needed to take into consideration the commitments of the VoIP manufacturers.

Considering those remarks, we propose the following way to assign a mark to a functionality:

- a useless or not desired functionality will receive one point
- a desired but not mandatory functionality will receive three points
- a mandatory functionality will receive ten points.

When available within a short-range delay (e.g. six months), the functionality will be considered as present now (that short delay will probably be reached when actually deploying the network). If available within a medium-range delay (e.g. between six and twelve months), the functionality will only receive half of the possible mark. This way, we will have a means to favour a solution offering all functionalities now.

The functionalities that are to be considered when giving a mark to a solution may be found in the following list. This list can evolve according to the needs of the users and the response of the market:

- clustering of telephones
- several classes of service (according to the user's hierarchical level)
- wait music
- localisation of faults and related reports

- direct inward dial
- outward dial with mandatory implication of the operator for some users
- direct outward dial for the other users
- restricted outward dialling for some users
- authorisation / authentication code for some users: this code allows the use of another telephone to get some special functionalities normally not allowed with this phone
- common abbreviated numbering
- individual abbreviated numbering
- call transfer if busy
- call transfer if no answer
- call transfer to a standard common phone
- "follow-me"-type call transfer
- redial
- automatic redialling if busy tone or no answer
- call transfer bypass
- call pickup on a phone belonging to the same group
- call pickup on any phone of a common signalling group (all phones of the group ring until someone answers at any of these phones)
- second call signal and acceptation of this signal
- multi-party conference
- intervention of a director's phone
- phone clustering for inward dial routing
- on hold call
- automatic connection upon unhooking (e.g. emergency phone)
- after putting a call on hold, be able to take it again on an other phone.

5.2 Technical parameters

The technical parameters in such are parameters expressed in terms of reliability and performance of the telephony equipment. The performance criteria are voice processing quality criteria as well as technical performances criteria. The latter are the technical characteristics of the VoIP equipment which are not directly related to the telephony functionalities but which may influence the behaviour of the proposed solution in a particular environment.

Reliability: hardware and software

The reliability criteria are dependent on two factors. The first one of these is the reliability of the hardware equipment on which the VoIP solution application run. This reliability may be expressed with the mean time between failures (MTBF). These data will have to be provided by the manufacturer or validated by independent tests performed in a laboratory. The second reliability criterion is related to software applications. This one will probably be more difficult to estimate, as it would be the case for ant software. The user will therefore be invited to ask the manufacturer for a guarantee of reliability, associated with a guarantee about an intervention delay in case of the appearance of a bug in the software.

To make up for the lack of availability due to a hardware or software failure, a redundant architecture may be deployed. This will necessarily have an impact on costs, and an opportunity analysis will therefore have to be carried out: the cost of the redundancy will have to be compared to the possible loss of revenue due to the telephony system failure.

Another factor may influence the reliability of the telephony equipment: the power supply of the telephones. The telephony network provides for the necessary power supply of traditional phones, so that even when there is a failure of the electricity distribution network, the telephony service is still available. An IP phone is not connected to a traditional telephony network and does not get any electric current from its network (a standard ethernet network). It is therefore dependent on the availability of electric power. Some manufacturers propose an alternative solution which carries the electric power on unused wires in the UTP cable. It is then the job of the switch to provide that power.

The problem is thus transferred to the single switch which is of course dependent on the electric network. If this brings some easiness in the physical handling of the IP phones, the dependency on the electricity distribution network remains. As for critical computers, a supplementary equipment (batteries, generating unit) will have to be planned if the user wants to be able to make calls under any circumstance.

In this field of reliability, another important technical factor is the possibility of maintenance, preferably with use of a modem or another remote device, so that an action can be taken by a remote technician. It is especially important if the maintenance is outsourced because in that case the technician is likely to work at the maintenance company's premises.

Voice and technical performances

In addition to those reliability criteria, we have to interest ourselves to voice and technical performances criteria. We will first examine what factors influence the quality of the voice, and then the technical factors to take into account when thinking of a VoIP solution.

Voice quality factors

The performance of the voice quality is directly influenced by three factors, which are the following:

- **The choice of the coder-decoder (codec) used**. Each ITU normalised codec has indeed an intrinsic restitution quality. The G.711 codec is the one offering the best possible quality, and is used on digital trunks within the POTS. This codec makes use of the pulse code modulation (PCM) coding: it is an uncompressed coding that uses 64 kbits/sec of bandwidth, which is exactly the usual telephony quality. Other codecs use different coding algorithms having each their own bandwidth and quality characteristics. However, the use of a particular codec has also an impact on the CPU load and the processing delay of a voice packet.

  In a large size network, the processing delay may prove to be an important factor considering the maximum delay of 150 msec that should not be exceeded, as exposed in the first chapter. The different characteristics are recapitulated in table 1.

Furthermore, the choice of codecs available may also be important when an interaction with other codecs is necessary, for instance when going through a gateway. The change of codecs may actually have a harmful influence on the resulting quality, when using a compression with loss algorithm.

Evaluation: the intrinsic quality of a codec is given by its mean opinion score (MOS[17]). This score is given on a scale going from 1 to 5 where 4 is the toll quality (POTS quality) and 1 reflects an incomprehensible speech. However, the other characteristics of the codec must also intervene in the evaluation. The CPU load is not quite relevant because a dedicated DSP is supposed to be able to perform the task for which it has been designed. On the other hand, delay and consumed bandwidth are important.

Each of there three factors –MOS, delay and bandwidth–, may be favoured by the user who is aware of the consequences on the network design (in interaction with the internal environment parameters that are the number of simultaneous communications to handle, or the existing network capacity – see below).

The user will so be invited to evaluate the importance he gives to each factor, and so allowing to favour the choice of a codec or another. One user will for instance favour the bandwidth consumption to the

[17] MOS estimation method : this is a subjective method, in which several « testers » are invited to listen to the same voice sample which is coded with the different codecs. Each participant gives a mark to each sample according to the following scale:

1 ≡ unsatisfactory speech quality, very annoying level of distortion

2 ≡ poor quality speech quality, annoying level of distortion , but not objectionable

3 ≡ fair quality speech quality, perceptible level of distortion, slightly annoying

4 ≡ good quality speech quality, just perceptible level of distortion , not annoying

[TOLL QUALITY]

5 ≡ excellent quality speech quality, imperceptible level of distortion

detriment of the processing delay, or the voice quality whatever will be the need for bandwidth.

- The second factor having an influence on the voice quality is **the QoS factor**. If this one comes indirectly from the chosen codec, this factor has all its importance in a high-level traffic network or a network where the resources are limited. It is indeed of the utmost importance to be able to favour the voice traffic inside the network, still having in mind that delay of 150 msec. The packet loss has an influence on that delay (by the use of buffers), but has also an influence on the quality of the conversation: if losses are too important, playback buffers actually lose their interest. The QoS problem has already been mentioned here above, in section 1.3.

Codec	**Coding type**	**Bandwidth**	**Frame size**[18]	**Algorithmic delay**	**MOS**
G.711	PCM	64	0.125	0.125	4.1
G. 726	ADPCM	32	0.125	0.125	3.8
G.727	ADPCM	32	0.125	0.125	3.85
G.728	LD-CELP	16	0.625	0.625	3.6
G.729a	CS-ACELP	8	10	20	3.7
G.723.1	MP-MLQ	6.3	30	45	3.9

Table 1: Codec characteristics

Evaluation: QoS support is hardly estimable in some other way that by the observation of the presence or absence of appropriate mechanisms. If QoS support is not always relevant, some mechanisms may be more appropriate than other ones in a specific situation. On a switched LAN

[18] msec

for instance, the QoS mechanism that can be provided is the 802.1p/q (working with ethernet frames). However, if extending the VoIP to a WAN is scheduled, QoS support will become important. The user will so have to give its preferences for the following QoS mechanisms: WFQ, IP ToS, RSVP.

The compatibility of the equipment must also be carefully studied, as for codecs. (the chosen QoS mechanism has to be implemented by each hop of the network!).

- The third and last factor influencing the voice quality is the presence of **echo suppression mechanisms**. Echo is a very annoying thing when speaking to someone else and may be present even with VoIP applications as a physical characteristic of the VoIP devices. There exist echo suppression mechanisms that lead to a better comfort when speaking to another person on the phone.

Evaluation: this factor is merely evaluated by establishing the presence of such mechanisms, which will add to the global technical mark for the telephony solution.

Technical performance factors

The technical performance factors as such may be identified as the following ones:

- **The extensibility of the network**: to protect the investment, the user needs to think of scalability and upgradability problems. Those problems depend on the capacity of the chosen hardware equipment but also on the choice of the protocol used

If the hardware is chosen to match the exact current needs, it will maybe be unable to support further developments of the organisation. The increase in traffic needs may oblige to change that equipment.

If several protocols are suitable to the situation in which a VoIP network is deployed, a future extension of the organisation or interconnection with other organisations may be facilitated by a specific protocol. The reader is here invited to refer to the first part of this work, in section 3.1 for the description of the different VoIP protocols.

A protocol gateway may induce the loss of some functionalities if these ones are not supported in the same way by the interconnected networks. At this stage of things, it is quite difficult to tell what protocol is the best when addressing that problem, as VoIP technologies are not broadly spread and still undergoing developments.

- **Voice activity detection**: a voice activity detection (VAD) mechanism allows a better use of the available bandwidth by suppressing the packet flow when no speech is detected. Remember that a conversation needs a two-way connection, even if both parties are not speaking at the same time. If there is no VAD mechanism, "empty" packets are transmitted and lead to consume two times the necessary bandwidth.
- **CRTP support, and adaptation of the RTP payload size**: the support for compressed RTP (more accurately the RTP/UDP/IP header compression) and adaptation of the RTP payload size are bandwidth efficiency factors. The adaptation of the RTP payload size (RTP MTU) is a trade-off between bandwidth consumption optimisation and delay, and as such could also be a QoS factor.

[*VAD, CRTP and RTP MTU adaptation are all related to bandwidth consumption optimisation.*]

- **Fax over IP support:** the fax over IP (FoIP) support is also to be taken into account when evaluating a solution. FoIP requires different mechanisms and protocols than those used for VoIP and is not always available together with VoIP features, even if it is an interesting application which leads to a really integrated network (if no FoIP support is available, the use of faxes may involve a dedicated network, thus reducing the interest of VoIP).
- **Security functionalities**: the security of the telephony network may be expressed with several factors, some of which being directly related to the use of VoIP technologies:

- **Network access restriction**: a network access restriction may be desired and implemented on the basis of either a personal identification number (PIN) or a smart card. The use of a PIN code may also be used to define user groups, as found in the functionalities list here above.
- **Call security**: the security of the call may be provided by the use of encryption mechanisms such as IPSec or H.235 (depending on the signalling protocol used).
- **Physical attack protection**: the protection against physical attacks depends directly on the vulnerability of the network that is used to carry the voice (and data), and is similar to the vulnerability of any computer network.
- **Protection of the servers against attacks**: VoIP solutions rely on software applications which in turn run on standard operating systems (MS Windows NT/2000 or any "UNIX flavour"). DNS and DHCP servers are also used for the good functioning of the telephony network. All these servers should be protected against software attacks, e.g. by setting up a good firewall.

Evaluation: To evaluate the security provided by the network, the user will have to establish the presence or absence of those parameters, and give them a relative importance (encryption of calls for instance may not be considered as a critical function)

5.3 Internal environment parameters

The parameters of the internal environment are parameters that do not directly come under the technical or functional level, but that are bound to the environment in which the telephony network will be deployed. These parameters have in turn a direct implication in the cost evaluation.

They may be identified as being the following ones, and evaluated in terms of cost as developed in chapter 6 below.

- **The existence of a telephony or data network in the organisation.** A VoIP network may be used as a means of integrating data and voice transfer on the same network, so that even if there is already a telephony network, this one could be abandoned for a single integrated network. If there is already a data network but no telephony network, the question is to know if this network has the required capacity to support the voice over IP, either hardware (VoIP-enabled devices) or in terms of bandwidth.

  When considering the absence of either network, this does not necessarily mean that there is absolutely no network, but that a new network of the considered kind is needed (redesign of either network; obsolete equipment, etc...)

- **The skills to acquire to support the VoIP.** The administration of the network may require that new skills are acquired by the organisation, or a supplementary person be appointed. An existing system administrator could also have the task to monitor the VoIP network. The problem here is to examine if there is a need to appoint a new employee, or not. This is part of the maintenance cost, maintenance that can be outsourced, partly or entirely to the choice of the user.

The skills to acquire may also intervene in the installation cost as the cost for an initial training of either end users or system administrators (either newly hired or not). The use of a VoIP integrated network may lead to an economy when having to train systems administrators.

- **The size of the building to equip.** The size of the building will influence the kilometres of cable to be placed, and thus the cost of the installation. If a single network is installed, this means that there is also a single wiring to do, thus reducing the total cost.

- **The effective number of network users.** Within an organisation, not everybody uses its telephone at the same time, so that it is useless to design the network to support so many simultaneous communications, either internal or outward. For a VoIP LAN with a PSTN gateway, the internal capacity is function of the bandwidth available on the network as well as the capacity of the dedicated servers (H.323 gatekeeper or SIP proxy) while the outward capacity if function of the line capacity of the gateway.

- **The itinerant nature of the employees, or the amount of home workers**. If some employees of the organisation often travel for their work, such as in the case of sales representatives, or that other employees often work at home, it can be interesting to use an IP network for telephony. An IP connection is often less expensive that a voice connection and can be used to go anywhere IP can go on the organisation's network, which means that the VoIP features of that network can also be used by the tele-worker. For a phone call, the call can be initiated on the end-user's PC but handled by his regular phone (possibly with a call-back feature from the organisation's central VoIP server).
- **The benefits of a unified messaging solution**. As we have already underlined it in section 1.3 when introducing the voice over IP, the integration of data and telephony networks makes the idea of the unified messaging more real and is a further step after the CTI functionalities offered by PBX's today. A total integration of voice mail, fax, e-mail and regular telephony may lead to a increased productivity of the employees and so be translated in terms of benefits for the organisation. The interest of the unified messaging is also to relate to the last parameter here above, *the itinerant nature of the employees, or the amount of homeworkers*. The main problem when using a unified messaging device is of course that in case of failure, all communication means are unavailable.
- **The frequency of movings in the organisation**. It is generally considered that an employee moving from one office to another induces a configuration cost of USD 500 (≈ BEF 22400)[19]. A VoIP network permits to avoid a reconfiguration, due to the characteristics of an

[19] source : CISCO Systems

ethernet network (unless the IP address of the phone has to be changed due to a change of network segment – another switch).

5.4 External environment parameters

The parameters of the external environment are of the same kind that those of the internal environment, that is parameters which do not directly come under the technical level but have an influence on them. Once again, this influence has in turn a repercussion on costs.

The external environment parameters may identified as being the following ones:

- **The number of external lines to provide.** The number of lines for outward dialling to provide is itself dependent on the size of the organisation (i.e. the number of employees that could call the external world), but is also dependent on the nature of the organisation's work (e.g. when having many customers likely to call at the same time). For that reason, it is also a external environment parameter. This influences the choice of a gateway.
- **A possible extension of the VoIP LAN to a WAN**. Prospects about a further extension of the VoIP telephony is to be thought of. This could be envisaged in order to connect the VoIP LAN to another branch of the same organisation through a WAN or VPN (which is functionally the same), or to connect to business partners (if there is already a strong EDI WAN connection, we could imagine that voice and EDI links would be integrated)

  Such an extension leads to consider several parameters of the technical level. The protocol used on the LAN is important, as described in the technical parameter section, for compatibility and scalability reasons. The choice of the gateway and its QoS and codec capabilities is also important when using a WAN connection where several routers may be implicated. Those routers have to be compatible with the QoS features of the gateway (which in this case is no more a VoIP/PSTN GW), and there should be as few codec changes as possible. This has already been exposed in the technical parameter section here above.

Conclusion

In this chapter we have examined the parameters that should be taken into account when thinking of a voice over IP telephony solution and comparing it to a traditional PBX-based one. Among those parameters we can see that some of them are related to VoIP while the others are related to any type of telephony solution, such as the capacity of the central server / PBX.

Chapter 6 : The model: structuring the parameters

The fifth chapter of this work was to bring the parameters to examine in evidence. The subject of this chapter is to organise these elements in a structured evaluation model. For this purpose, the first step is to define a field of work hypothesis which will lead to take some specific factors into account, so that the choices in terms of hardware and human investments will be different. Those choices will in turn have a direct implication in terms of costs.

The question is now a matter of associating a particular situation with the cost induced by deploying a VoIP network or PBX-based telephony network.

6.1 Evaluating the cost of the telephony solution

The first thing to examine is the existence of networks within the organisation, and this will lead to specific technical questions. Having determined the situation and its specific parameters, we will then have to examine the costs they induce (i.e. the cost of the matching equipment and human investment), but also the benefits that can be brought in by the choice of the VoIP and that are not situation-dependent.

The cost of each possible solution will then be associated to the technical mark obtained by that solution according to the preferences of the user.

REM: *for the following situations, an obsolete network of any kind that is to be replaced should be considered as equivalent to the absence of this network.*

1. If there exists a telephony network only.

   In this first situation, there is no existing data network, the opportunity to deploy a VoIP network is dependent on two factors: the need for an IP network, or the need for a new telephony network. We suppose that the user knows the reason why he is thinking of deploying a VoIP solution.

If the telephony network is still useful, the question to ask is whether the amortisation for the existing network is completed or not. If a VoIP solution is scheduled to merely replace the existing network, it is likely to be the case and would match situation two. If there is a need for a data network (either new or as a new design), and that this is the opportunity of integrating both kinds of traffic, the residual amortisation is to be counted as a supplementary investment cost. The existing PBX could however be used as a VoIP/PSTN gateway if it is able to communicate with the LAN thanks to either a native IP interface or a standard signalling protocol (e.g. Q.SIG) interface that can be connected to a compatible router, as exposed on figure 8 in the first chapter.

Figure 8 in the first chapter shows a situation where POTS and VoIP cohabit. If the VoIP is an extension of an existing POTS network, then this should be considered as the situation four (see lower) where no network exists.

2. If there exists a data network only.

In the hypothesis where the only existing network is the data network, the question to ask is whether this network is able to support a telephony traffic, according to the needs of the organisation (i.e. the number of end users) and chosen codecs –see technical and internal environment parameters here above. It is here a question of bandwidth capacity and network technology (many end users demanding the best PCM quality on a 10-MB ethernet network would lead to congestion problems). On a switched network, the estimation of bandwidth needs is somehow made more complex (is there a difference in the voice load on each segment?) and the 802.1p/q QoS feature may be looked for.

The main problem here resides in the choice of the VoIP protocol and gateway which should be chosen according to the external environment parameters (number of lines to the PSTN, and WAN extension).

3. If there exist both kinds of network.

If both kinds of network exist, then we are facing a combination of the two situations here above. This situation, where both networks are still useful, is not likely to happen, except in a case where there would be a strong attraction of VoIP benefits, such as the unified messaging and the mobility features of the VoIP. If it is the reason to think of a VoIP telephony, then the questions of both situations here above have to be considered: the reusability of the PBX as the VoIP/PSTN gateway and the upgradability of the data network.

4. If there exists no network.

When facing a situation where there is no network, which would be the case when having to equip a new building or having two obsolete networks, there is no question of device reusability or upgradability but merely a question of choice of a technology.

From then on, the diverse costs have to be estimated according to the situation. Those costs may be organised in four groups on the basis of their nature. Those groups are the following ones:

- the equipment cost

  The equipment cost heavily depends on the situation the user is facing, and is influenced by technical and environment parameters (capacity, choice of codecs, etc).

  - When facing situation one, the VoIP equipment cost is the cost of the gateway, IP phones, central servers (either H.323 gatekeeper or SIP proxy / registrar / indirection – both hardware and software) and wiring. As noted, the gateway may be the existing PBX. The possible residual amortisation cost is to be added to this equipment cost (except maybe if the PBX is used as the gateway).

    There is also the cost of entirely new data network devices (switches, hubs) that is to be taken into account. If the VoIP way has been chosen because there were anyway a need for a data network, those devices should not be taken into account for the cost of the telephony, as they are primarily needed by the data network. Switches will be present according to an efficient data network design. For a small network, they may be irrelevant.

- When facing situation two, the VoIP equipment is roughly the same as the one of situation one, although an existing WAN access router may be used as gateway. If the network is able to support the VoIP traffic, there is no further cost of wiring or new devices.
- When facing situation three, the consideration of situations one and two are united.
- When facing situation four, the VoIP equipment cost is similar to the one of situation one, although in this case there is a need for a new gateway. Once again, the cost of wiring and data network devices may be allocated on the basis of its primary use which is the data network part of a global project – if the distinction is needed.

- the installation cost

The installation cost is the labour costs associated to the solution deployment. Those costs are always present but depends of course on the amount of work to be done. If a usable data network is already present, there is not labour cost due to setting up that network. The same remarks for the costs of data devices are relevant for the labour costs (the more data devices are needed, the more the labour costs will be high).

- the skill acquisition cost

The skill acquisition cost is the cost of an initial training of people. This training will be more or less important due to the skills already present in the organisation, which is dependent on the situation faced by the user.

The cost of wages for the employees performing system administration or support tasks is not a skill acquisition cost but a maintenance cost.

- the maintenance cost

  The maintenance cost is both related to the hardware and software maintenance (hardware to replace, software to upgrade or update), and to the all-day operation cost. The latter includes the wages of the employees performing administration and support tasks and the exceptional cost of a moving (which should be nearly nil if the VoIP is chosen).

  The integration of both networks could lead to lower operation costs, as a single skill type is needed: IP management. Administering IP phones is similar to administering any workstation on a network (e.g. same principle of DHCP).

Skill acquisition and maintenance may be outsourced by the user, so that there is no real skill acquisition cost but a heavy cost of maintenance. It is up to the user to make the choice of an internal skilled staff or of the outsourcing.

6.2 Evaluating the global technical mark of a telephony solution

Once having determined the cost of a telephony solution, either PBX or VoIP-based, it is time to evaluate the mark of this solution, according to the preferences that the user has expressed. As we have proposed it, there will be two marks, one for the telephony functionalities and one for the technical aspects.

6.2.1 Calculating the functionalities mark

As already mentioned in section 5.1 here above, each functionality will be given a mark that reflects its importance for the user.

The functionalities offered by the solution to evaluate will be matched to the user's preferences, so that the functionalities mark for that solution will be obtained by summing each mark. As the list of potential functionalities is an open list, there is no definition of a possible maximum score, although a downsizing to a maximum of 100 points would be possible.

6.2.2 Calculating the technical mark

The technical mark is somehow more complex to calculate than its functionalities counterpart. Some of the technical aspects actually influence design choices, or could on the contrary be guided by design constraints (such as the available bandwidth), but also depend on user's preferences. Security aspects are not related to design choices, but depends on the user's interest.

It is the reason why we do not want to give a fixed score to all technical functionalities, but prefer to let the user indicate his preferences within a fixed scale, for some of the technical parameter groups. Using fixed scales, we are able to have a fixed maximum score which is set to 100 points. The following maximum scores are used :

- voice quality factors: 40 pts,

  of which: 20 for QoS factors, including VAD, CRTP and RTP MTU for 10 points out of the 20.

  10 for codec choice

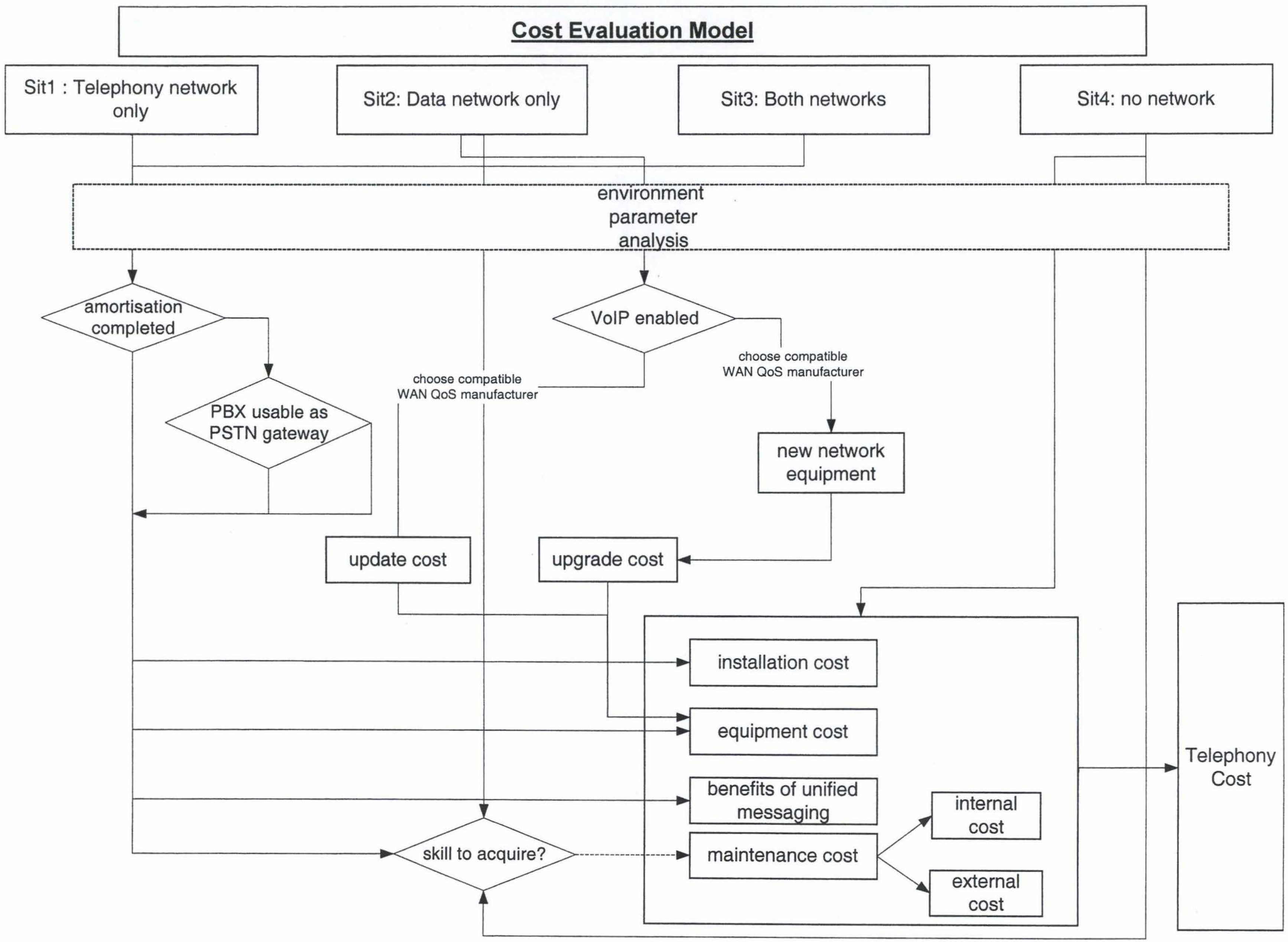
  10 for echo cancellation

- reliability factors: 30 pts
- security features: 10 pts
- scalability (extensibility potential): 10 pts
- FoIP support: 10 pts.

The user will have to make a choice for QoS factors, codec choice and security features, as he may have strong preferences for those parameters, preferences which may well not be those of another customer. It will then be up to the software to calculate the technical mark of the proposed solution according to the user's choice. A score of 100 would match exactly the preferences of the user, which means that all he asks for is present.

Those marks, and their relative importance, are justified by the fact that the voice quality is the whole point of a telephony system : it is why this is the most important item – and inside the voice quality factor, each item has the same significance. Reliability is also crucial for a telephony system: even if it is able to achieve the best possible voice quality ("toll quality"), if the equipment fails to run correctly, there is no more phone call possibility. Security, scalability and FoIP support may all have the same relative importance and it is why they all receive 10 points.

Reliability here is meant to include the hardware reliability, software reliability (included the possibilities of redundancy) as well as the easiness of performing maintenance tasks.


Cost Evaluation Model
Sit1 : Telephony network only
Sit2: Data network only
Sit3: Both networks
Sit4: no network
environment parameter analysis
amortisation completed
VoIP enabled
choose compatible WAN QoS manufacturer
choose compatible WAN QoS manufacturer
PBX usable as PSTN gateway
new network equipment
update cost
upgrade cost
installation cost
equipment cost
benefits of unified messaging
maintenance cost
internal cost
external cost
skill to acquire?
Telephony Cost

PART III.

Chapter 7 : Case study : The « CAMET » building of the Walloon Ministry of Equipment and Transports in Namur

This last chapter, and the single one of this work's third part, will be devoted to the case study of the CAMET [*Centre Administratif du MET*, Administrative Center of the MET) which is a building of the Walloon Ministry of Equipment and Transport (MET) located in Namur.

This building is an interesting case of VoIP deployment and a big reference for Cisco Systems, the manufacturer of the VoIP equipment. It is currently in the top five of major VoIP networks around the world with 900 IP phones, and probably the largest one in Belgium.

This chapter will have three sections, the two first ones introducing the context of, respectively, the MET and the CAMET building. The last section will apply the model proposition of part II to this case.

7.1 The context of the MET

The MET (*Ministère de l'Equipement et des Transports*), Walloon Ministry of Equipment and Transports, is a Ministry that was created in July 1989 as a consequence of the federalisation process of the Belgian State. Its mission is to manage, keep up and develop:

- the motorway and road network,
- the waterway network, dams and rivers port,
- the airports and airfields,

of the Region, as well as the buildings of the Walloon Region's services and ministries.

It is a Ministry that operates 160 buildings scattered throughout the Walloon Region of Belgium, and has its own telecom network which was primarily installed to monitor the motorways of the Region.

This network is an important asset of the ministry, as it is a high-speed fibre optics network using state-of-art technology. As there was a high excess of available bandwidth, it was decided in 1996 to use that capacity to offer high-performance data network services to public organisation as well as home users. That is how the WIN network (Walloon Intranet) was launched, and a commercial company was started to operate the network, the backbone of which being the fibre optic of the MET.

7.2 The particular case of the CAMET building in Namur

Until 1999, the administrative services of the MET were dispersed throughout the Walloon Region: this was obviously not an efficient situation. It is why it was decided to build a brand new building to group all those services in the capital of the Region, Namur.

The new building had to be big enough to accommodate a thousand or so civil servants, most of them needing a personal telephone and computer, as belonging to administrative services. It was so necessary to equip that building with a high-performance computer network and a telephony system.

In the first half of 1999, a call for tenders was made which has for object an advanced PBX-based telephony system. The value of the contract was then estimated to 26.6 MBEF.

At that moment, Cisco Systems proposed its VoIP solution as an effective telephony solution. As a guarantee of its seriousness, it offered to replace its solution by a traditional PBX in case of failure, and committed itself to collaborate with the prime contractor, the WIN S.A. Being already the supplier of network equipment for the WIN, no compatibility problems were to fear if the LAN telephony were extended to a WAN telephony network, which was actually the idea of the MET.

WIN S.A. had already been appointed as prime contractor for the installation of the LAN in the CAMET's building, so that it was really able to add the VoIP features to that network. The LAN is a switched 100 Mbps ethernet network relying on a gigabit ethernet redundant network. It was obviously able to support the VoIP traffic, even using the uncompressed PCM-based G.711 codec: 900 users using their phone at the same time would yield a traffic of 8.84 Mbps[20], and this without any bandwidth limitation scheme such as VAD or CRTP. Moreover, as the network is switched, those 8.84 Mbps would not be sent to every segment and the backbone running at 1 Gbps is evidently able to absorb that traffic. This already shows that no QoS feature is needed on the LAN. Figure X shows how the CAMET LAN is structured: the core 1 Gbps network is switched by 6509 Catalysts with redundant connections to the 4303 Catalysts that switch the user 100 Mbps ethernet network.

We have told that the idea of the MET was, and still is, to extend the VoIP use to its VPN and so have a cheap telephony network for its 160 buildings. On the WAN, QoS are more important than they are on the high-capacity LAN. Cisco's equipment is able to deal with QoS features such as the automatic choice of a low-bandwidth codec such as G.723.1 and G.729a, IP ToS (the IP precedence bit is automatically set by Cisco's IP phones), CRTP, RTP payload adaptation, WFQ and WRED to prioritise the VoIP (RTP) traffic. As all the routers of the VPN are manufactured by Cisco, this would mean that QoS respect would be ensured.

The specifications of the MET demanded 60 lines to the PSTN (ISDN network of Belgacom) as well as 30 lines to the mobile Proximus network (i.e. 2 +1 PRA accesses). To answer to that demand, the Cisco AS5300/Voice Gateway was chosen for its high duty performance and high reliability (>350000 hours of MTBF). Independent tests also showed that it was one of the best VoIP gateways available on the market.

Analogue lines were also demanded, for faxes and modems. A standard Alcatel A4400 PBX was chosen for that purpose. – the MET wanted to reuse its faxes, IP faxes being far too much expensive.

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[20] G.711 yields a packet of 206 bytes every 20 msec, which corresponds to a 82.4 kpbs bandwidth consumption.

Finally, it is also interesting to notice that the MET is not the single Walloon ministry that uses a WIN-based VPN. Another major administration of the Walloon Government, the so-called MRW (*Ministère de la Région Wallonne*, Ministry of the Walloon Region) has also a VPN solution from WIN S.A. It would so not be stupid to think that the MET telephony could be interconnected with the MRW's VPN in order to offer a cheap telephony between the two ministries. It is also quite interesting to notice that the RTT[21] on the WIN network is less than 60 msec: keep in mind that the maximum delay for a good voice quality is around 150 msec!

[21] Round Trip Time

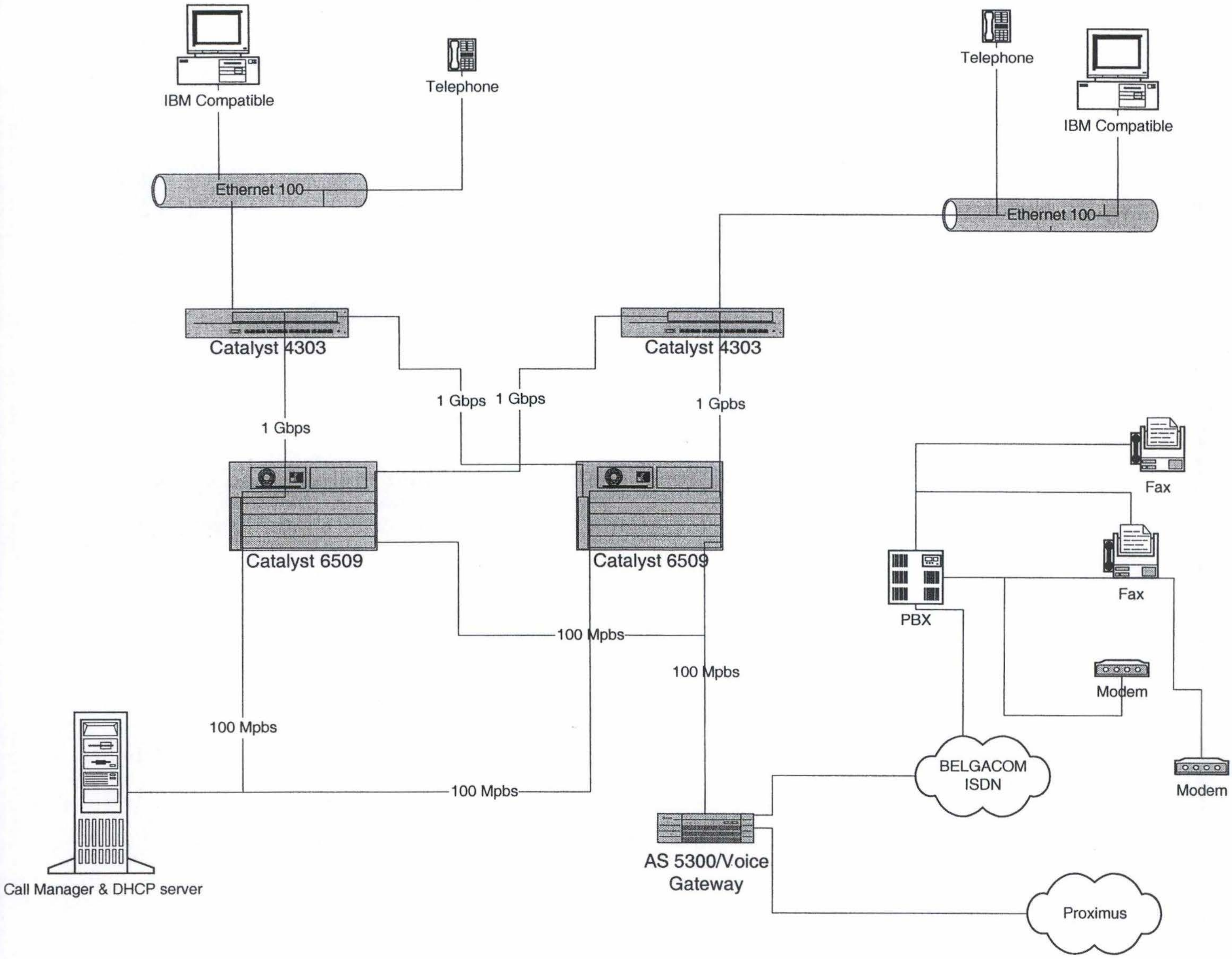

Figure 15: CAMET VoIP LAN Design

7.3 Application of the proposed model to the CAMET building context

Considering the elements presented here above about the CAMET building and its context, we can make out some of the internal and external environment parameters of the model proposition that was the subject of the second part of this work. The subject of this last section is to apply the model proposition to a real case, namely the CAMET case. We will so review the model's parameters one by one and apply them to the present case.

The first thing to do is to identify the situation to refer to. The CAMET is a new building, so that nothing was present beforehand. This would lead us to think that we are facing situation four of the model proposition. This is not really true, as the LAN was already on its way when it was thought of a VoIP telephony solution, which had to insert itself on this LAN. So, the situation to consider is the second one, where a network already exists. It is true that this was a brand new one, but nevertheless it was there, and explicitly quoted as a constraint.

7.3.1 Functionalities and technical parameters

With regard to the functionalities evaluation matter, the CAMET case was not subject to such an evaluation, as Cisco's offer was the only one that could be delivered immediately and that was a whole telephony system (voice gateway, IP phones and controller). However, some of the needed functionalities were not present but were promised by Cisco at the latest on the 1st of July 2000 – six months after the deadline for the VoIP network deployment, but more than one year after the initial proposition. Willing to chose the IP way, the MET agreed to wait for some of the functionalities that were though considered as mandatory. The resulting functionalities mark would have been lower than any of those obtained by a PBX-based solution, but the only one available for a IP-based solution.

The functionalities not present at that time were the following ones:

- abbreviated dialling (but some numbers can be programmed)
- automatic redialling if called line busy
- "follow me" call transfer and call transfer bypass
- intervention of a director's phone.

[notice that the use of the SIP protocol would have allowed a native "follow me" call transfer mechanism thanks to the use of the redirected call model. At that the time of this market however, there was no SIP product available]

We have now to evaluate the technical mark of the VoIP solution proposed by Cisco for the CAMET building. The voice quality factor for Cisco equipment is one of the best available on the market: it offers a broad choice of codecs (G.711 and G.729 on the Selsius phones – the new models offer G.711 and G.729a, G.711, G.729, G.723.1 on the AS5300), as well as all QoS factors (RSVP, CRTP, RTP payload adaptation, VAD, WFQ, IP ToS, WRED, and control admission at the CallManager) and echo cancellation.

The reliability of the solution offered by Cisco was high, at least when looking at the hardware reliability. For instance, the AS5300 has an MTBF of 343 581 hours, and the NT server used for the CallManager and DHCP server is highly redundant. A spare AS5300 was also scheduled, so that a possible problem could be solved very quickly. WIN S.A. had to ensure a quick service (2 hours) in case of a major failure, so that the global reliability is high.

Being compliant to the H.323 standard, both the CallManager and the IP phones have to provide to required security features. The CallManager is furthermore compatible with the MGCP standard, which gives it more scalability. It also provides FoIP support, as do the AS5300.

This would give the Cisco solution quite a good technical mark, but independent tests[22] have also showed that when in a WAN environment, the Cisco VoIP gateways have to work together with Cisco routers to achieve a good quality: if working with other routers, and especially if the QoS features are not compatible, the voice quality is so bad it is nearly impossible to speak to someone else. In the particular context of the CAMET, this is not to be taken into account as all network equipment is supplied by Cisco (even on the VPN).

7.3.2 Internal environment parameters

The situation of this case is identified as being Situation 2, "Data network only". The first question to ask is thus to know if that network is able to support VoIP traffic. The description of the CAMET LAN was done in section 7.2 here above: this is obviously a VoIP-enabled network that had been designed to provide high-performance services. In such a case, it is recommended to choose the VoIP equipment from the same manufacturer, here Cisco Systems, which happens to

[22] published in december 1999 in French dedicated magazines (*Décision Micro et Réseaux*, and *Réseaux*)

supply a complete VoIP solution (it was nearly the only one at the time of the market). Having a backbone at 1 Gbps, it was natural to connect the VoIP servers (CallManager, DHCP) and gateway to that backbone. The use of the data network for telephony needs avoids the need for a second separate network.

The size of the building to equip: it is here a building that required some 114 kilometres of UTP cable and as much of gigabit cable for the core network. The kilometres of cable needed for a second network, dedicated to telephony services was not estimated but would also have been a quite large number, so that a single wiring lead to economies (wiring for the 100 Mpbs network cost 8 642 667 BEF, VAT included).

The effective number of network users is defined by the amount of IP lines required, in this case 1100. Even if using the G.711 codec, this is not a problem for the capacity of the LAN. (see section 7.2).

The benefits of a unified messaging has not been established, but is an underlying idea when thinking of a VoIP solution. This is a scheduled step when listening to the responsible at WIN S.A., and was quoted as an advantage of using the Unity system for Voice Mail – system which had to be abandoned due to the too fast developments made by Cisco.

The frequency of movings in the organisation was also a factor taken into account for VoIP, as such movings are costly and occur with a not negligible frequency.

The skills to acquire were estimated as follows:

- 1.5 full-time equivalent for the administration tasks, either performed by employees of the MET or of WIN S.A.

- the maintenance cost was invoiced 280 755 BEF per month (after the first year – 219 162 BEF during that year of guarantee)
- 4 operators were to be trained.
- a training session for end users had to be scheduled.

The analysis of the internal environment parameters show that, on the one hand, the LAN was able to support VoIP traffic without problem, and that a voice gateway and CallManager could easily be added and take advantage of the gigabit ethernet core network. On the other hand, it also shows that VoIP can be useful for such an administration where internal moving occur and where a unified messaging system can be interesting.

The VoIP in that context required also the acquisition of skills, needed to use the telephony system in all-day operation, as well as for the maintenance.

7.3.3 External environment parameters

The number of lines to provide was fixed to 2 PRA accesses to the ISDN network of Belgacom (with VPN support), and 1 PRA access to the mobile Proximus network.

The possible extension of the VoIP LAN to a WAN was scheduled and guarantees of feasibility demanded.

The future prospects of the WAN extension, promising a cheap telephony for the about 160 buildings of the MET could be a major factor to take into account. Having a end-to-end VoIP connection is more efficient that using the single VoIP WAN connection, because then all signalling is done by the IP network without any needed translation, and the codec can be chosen at the phone to match the available bandwidth of the WAN, so that no further harmful codec translation is needed.

7.3.4 Cost analysis

The above elements have their impact on the total cost of the VoIP solution, which was 35 062 442 BEF, VAT included. One year of maintenance was also included in that price. Without that maintenance, the total cost is 31 880 208 BEF. The PBX market has been estimated to 26.6 MBEF.

This means that the initial investment was higher for a VoIP-based solution, but we have to remember that it brings some advantages such as a zero-cost moving, way to unified messaging and WAN telephony. A drawback was also the high cost of the IP phones, which is expected to decrease as demand grows.

How is that total cost divided among the four cost entries?

The **installation** (900 IP phones, configuration, analogue links, ...) **cost** amounts to 2 109 740 BEF – VAT excluded.

The **equipment cost** amounts to 23 612 539 BEF (VAT excl.), of which more than 13 MBEF (16.5 MBEF VAT included) for the phones only: this clearly shows that the high cost of those phones has a negative impact on the total cost. The cost of the standard phones (with their installation) for a PBX-based solution was estimated to 4.4 MBEF. The remainder of that amount is taken by the voice gateway, NT server and analogue PBX. This PBX was needed to support the analogue faxes of the MET, as explained in section 7.2. The AS5300/Voice Gateway was chosen due to its high reliability and capacity to handle the required 3 PRAs. A Compaq server was chosen to run the CallManager and DHCP server, due to its high redundancy level and induced reliability. In situation 2 where the network is fully VoIP-enabled, no other device is required. The servers cost was so far lower than the one of a standard PBX, estimated to 18 MBEF.

The **benefits of unified messaging** were not estimated.

The **maintenance and skill acquisition cost**.

The training cost for the four operators was invoiced 375 000 BEF (VAT excluded), at the rate of 25 000 BEF per man day. The end-user training was invoiced 250 000 BEF, at the same rate. Such a training was not estimated for the PBX market, but is likely to have been similar (as there would have been a new PBX which would have required a training. Most of the PBXs use proprietary technologies).

The cost of administrative tasks, estimated to require 1.5 FTE per year, was 550 000 BEF/month if executed by employees of WIN S.A.

The maintenance in itself was invoiced 280 755 BEF (VAT excluded) per month, after the first year of guarantee.

Conclusion

This case study leads to some interesting conclusions, even if some elements lack to match perfectly the model proposed in part II, and that some other political elements may have influenced the choice of a VoIP solution: the MET wants to be a example of usage of new technologies and to promote them in Wallonia, and the choice of the VoIP was perfectly corresponding to the telecommunication development politics of the Ministry.

The decomposition of the equipment cost clearly shows that a major drawback for the VoIP is the high cost of the telephones, when the cost of the servers and gateways may be qualified as cheap when compared to the one of a equivalent PBX.

It has been told that the cost of installation and maintenance would have been higher if the problems encountered by this network had been foreseen. At the time of deployment, neither Cisco or WIN S.A. had a strong experience of a large VoIP network, and the technology was a new one. There have been a lot of problems with that VoIP solution, mostly related to software instabilities, which needed to set up a new CallManager to avoid breaks in the availability of the telephony service. Some of the promised functionalities are still not available, and Cisco seems to have problems to supply them. The voice mail, for instance, has not been installed yet. But, there seems to be a very positive point for the VoIP: its voice quality sounds better than the usual quality achieved by a PBX, and users seem satisfied by their telephony network. When it does not fail.

Lacking of experience with VoIP, it is difficult to correctly estimate its cost, except of course for equipment cost. The CAMET case also shows that even if there is a high reliability for hardware devices, the software may well fail, even if it is supposed to handle the required amount of end users (there were obvious programming errors in Cisco's software, among others).

If things are taking a good path, and that more and more manufacturers offer VoIP solutions, including some of the telephony world major players, this experience shows that the user who wants a truly reliable network and can not afford to continuously look after bugs may have to wait a little bit more. When the cost of IP phones will be comparable to the one of standard phones, or even a little higher, then VoIP could really be an interesting solution, especially is there is already a high-performance network in the organisation.

Conclusion

In this work, we have first studied how the Internet Protocol can be used as an effective means to carry voice telephony signals, and how a good result can be achieved. We have then tried to determine if voice over IP applications are interesting to think of when being in the context of the internal telephony network of a private organisation, before presenting the case study of a large-sized VoIP network in Belgium, namely the network that can be found in the CAMET building of the Walloon Ministry of Equipment and Transports in Namur.

The first part showed us that the basic principles of the public switched network an inefficient solution with today's technologies which have replaced the copper wire with high-capacity fibre optic. There is indeed a huge loss of means within the PSTN where a circuit is dedicated to a single conversation, even if it is unused. With the emergence of voice over network technologies, such as VoATM and VoFR it became possible to use packet switching networks to carry the voice, thus avoiding to reserve needless capacity in the network.

With the mastering of quality of service factors that are required by guaranteeless IP networks, VoIP is now also a reality. As IP has become a popular and democratic protocol, the growth of demand for VoIP networks has lead all major manufacturers to offer VoIP devices, and many studies, such as ones of the GartnerGroup, anticipate the fact that IP will become a generalised means to carry voice and replace the traditional PBXs in the organisations.

As for today however, the case study showed us that VoIP technologies are not so reliable yet, at least when used for LAN telephony which involves more than a simple QoS transport of RTP packets, but also major signalling and management tasks by a central server. The hardware devices may be reliable, but as long as the software will not be so reliable, there will be a risk of service unavailability.

The last part also showed that VoIP is still expensive today, and this mainly because of the high price of IP phones. If economies can be expected with the use of a VoIP network, it is above all because of the easiness of administration and increase in productivity that be due to the use of a unified messaging system. This is the major promise of IP when used for LAN telephony.

Maybe will we see a global generalisation of the VoIP in the next years, even for public services. This could lead to a mobile communicating world, with a single protocol for all communication needs. But for now, there are still legal limitations to the use of IP in some countries, which have for purpose to protect the (old) monopolies of the telephony network. This must also be a sign that IP could very well replace the PSTN we are used to,...

Glossary

🕮 ATM : Asynchronous Transfer Mode

Designates a high-performance digital network, using virtual circuits. ATM is a data-link layer protocol with a strong intrinsic quality of service.

🕮 BRI : Basic Rate Interface

Designates an RNIS interface working at the basic rate of 64 kpbs.

🕮 Codec : encoder-decoder

A codec allows to convert an analogue signal, as voice or video, into a digital signal, and back. There exists a lot of codec types, based on the compression algorithm used. The term of "vocoder" is also used for such codecs. Commonly used algorithms are PCM (Pulse Code Modulation, uncompressed), and ADPCM (Adaptative Differential PCM, compressed) or ACELP (Code Excited Linear Predictor, compressed)

🕮 CRC : Cyclical Redundancy Check

Polynomial allowing to detect transmission errors thanks to the use of arithmetic methods.

🕮 DSP : Digital Signal Processor

A processor dedicated to process a specific type of signal and convert it from analogue to digital form. DSPs are notably found on sound cards and sometimes called "Digital Sound Processor".

🕮 EDI : Electronic Data Interchange

Refers to quite standardised procedures to exchange information by means of data network. EDI is often used for business-to-business links.

🕮 E&M : Ear & Mouth *or* rEceive & transMit

Interface that typically connects a PBX to another one, using analogue signalling. E&M is a signalling method using 2 or 4 copper wire.

🕮 FXO : Foreign Exchange Office

POTS interface having the same characteristics than a standard telephone. It allows to connect a VoIP device to a PBX or a telephony switch..

🕮 FXS : Foreign Exchange Station

POTS interface allowing to directly connect a standard telephone or fax to a VoIP device. It simulates a telephony switch.

🕮 IETF : Internet Engineering Task Force

Working group whose aim is to produce standards to use on the Internet, and so make its use spread out.

🕮 IP : Internet Protocol

The network layer protocol used on the Internet. It is a connectionless protocol. He gave its name, together with TCP, to the four-layer TCP/IP model.

🕮 ISDN : Integrated Services Digital Network

In the field of telephony, the ISDN is a digital network offering the same voice quality as the analogue network. This is achieved by using a 64 Kbps channel, called the B channel, along with a out-of-band signalling channel, called the D channel. Two B channels are typically associated to a D channel.

🕮 ISO : International Standardization Organisation

International standardisation organism which publishes standards for nearly all fields.

🕮 ISUP : ISDN User Part

Part of the ISDN signalling that provides the connection with the end user. This signalling operates on a higher layer that SS7 signalling.

🕮 ITU : International Telecommunication Union

Having its home office in Geneva, the ITU is composed of three sectors, among which the ITU-T is the telecommunication standardisation sector, formerly known as CCITT (Comité Consultatif International Télégraphique et Téléphonique). The ITU manages the communication standards on an international level, as for instance the world-wide numbering plan (E.164 numbering scheme). The two other sectors of the ITU are the ITU-R for radio-communications and ITU-D for development.

🕮 MTU : Maximum Transmission Unit

Refers to the maximum allowable size of the payload of a network protocol. For instance, the MTU of ethernet is 1500 bytes, and 64 Kbytes for IP.

🕮 LAN : Local Area Network

A local area network is a small size network, generally corresponding to the size of a building or a single room. An often met LAN technology is the ethernet, which works at the rate of 10 Mbps (on either coaxial cable or UTP cable), 100 Mbps (UTP cable only) or even now 1Gbps (called the gigabit ethernet). Token ring and token bus technologies are also LAN technologies.

🕮 P(A)BX : Private (Automatic) Branch Exchange

Telephony exchange that operates at the scale of a private telephony network. A P(A)BX manages an internal numbering plan and the outward connection to the PSTN.

🕮 POTS : Plain Old Telephony System

Term referring to the "old good" telephony system, mostly based on analogue technologies.

🕮 PRA : Primary Rate Access

Refers to an RNIS interface offering 30 (in Europe) B channels (see BRA), or 2 Mbps.

🕮 PSTN : Public Switched Telephony Network

The standard public telephone network. The term "switched" refers to the fact that a circuit is established in the network for the duration of each connection, whatever its real use is.

🕮 QoS : Quality of Service

Generic term referring to the quality of service offered by a network, in terms of reliability (packet losses) and delay.

🕮 RSVP : Ressource reSerVation Protocol

Protocol that specifies a method to achieve QoS in the Internet, on basis of a resource reservation mechanism that is processes at each router on the path between two endpoints. RSVP is specified by the RFC 2205.

🕮 ToS : Type of Service

One of the field in the IP header. Its purpose is to allow for QoS support directly at the IP level. It is however not really used, except for the 3 "IP Precedence" bits that define the priority of the IP packet.

🕮 SS7 : Signalling System 7

The signalling method used for the PSTN. The SS7 signalling runs on its own dedicated network which is independent from the telephony network, at least in North America. In Europe, SS7 trunks are often coupled with the voice trunks.

🕮 TCP : Transport Control Protocol

The transport layer protocol in use on the Internet. It is a connection-oriented protocol which needs to establish a communication channel before further steps can be performed. He gave its name, together with IP, to the four-layer TCP/IP model.

🕮 UDP : User Datagram Protocol

Connectionless transport-layer protocol also is use on the Internet. Packets are sent without knowing if they will even arrive somewhere. UDP provides no guarantee of delivery.

🕮 WAN : Wide Area Network

Network that covers a wide area and is most of the time composed of point-to-point between several nodes, that can themselves be LAN's.

🕮 (W)RED: (Weighted) Random Early Detection

A mechanism that randomly discards packets that arrive in a router buffer when this one is getting full. It may be weighted, so that a kind of traffic may be favoured.

🕮 WFQ : Weighted Fair Queuing

A scheduling mechanism used with network routers and that allows to define a different weight for several queues, but in such in such a way that every queue receives a fair share of bandwidth.

Annex

1. VoIP protocols message and packet formats.

1.1 H.323 message exchange diagrams

This section presents diagrams of message exchanges for H.323 operations. There is no packet format shown here, as they are binary-encoded (ASN.1).

Figure 17 presents the Gatekeeper routed call signalling model, where the H.245 procedures may be routed through the Gatekeeper or handled directly by endpoints (this method is still under development).

Figure 18 presents the Direct call signalling model.

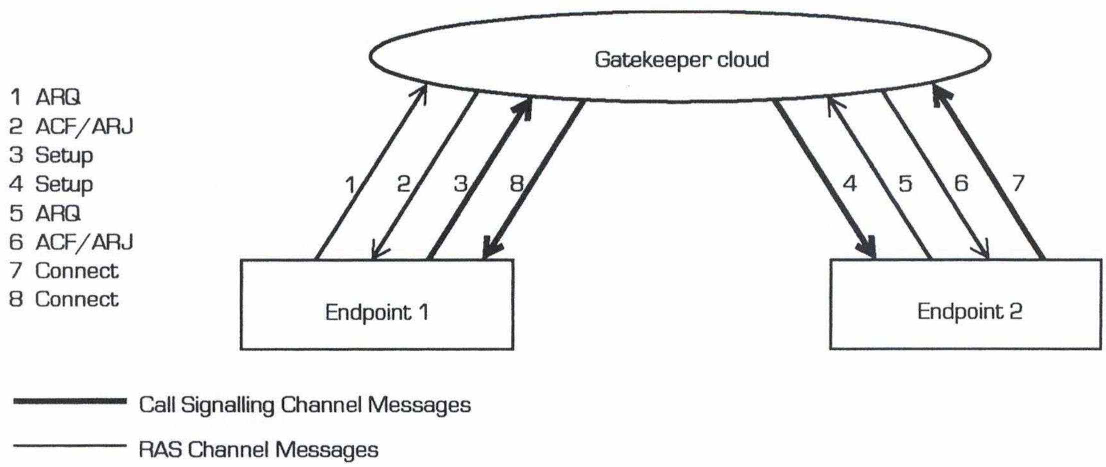

Figure 16 : Gatekeeper routed call model

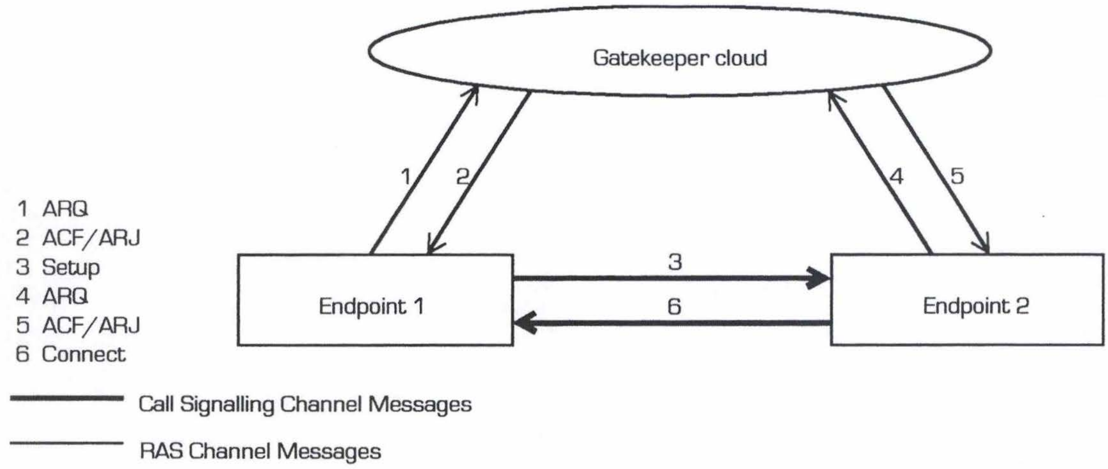

Figure 17: Direct endpoint call signalling

The following figures present the message exchange between endpoints to set up a call, either in a basic direct call model, with the presence of a single gatekeeper or with two gatekeepers. See legend of each figure for the exact case representation. When using a gatekeeper, it may happen that only one party is registered : that party only will have an ARQ – ACF/ARJ message exchange with its GK, but in the case where the GK routed call signalling method is chosen by the GK, the unregistered party has to use its services, i.e. use it as a proxy for all H.323 messages. All of these cases are not presented, as it is not very useful. The admission request exchange is always done just before placing a call, so that the "setup – call proceeding" messages may be received before an admission request is sent to the gatekeeper, as shown on some of the figures.

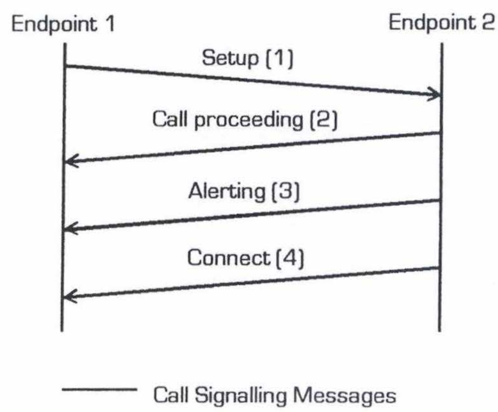

Figure 18: Basic direct call

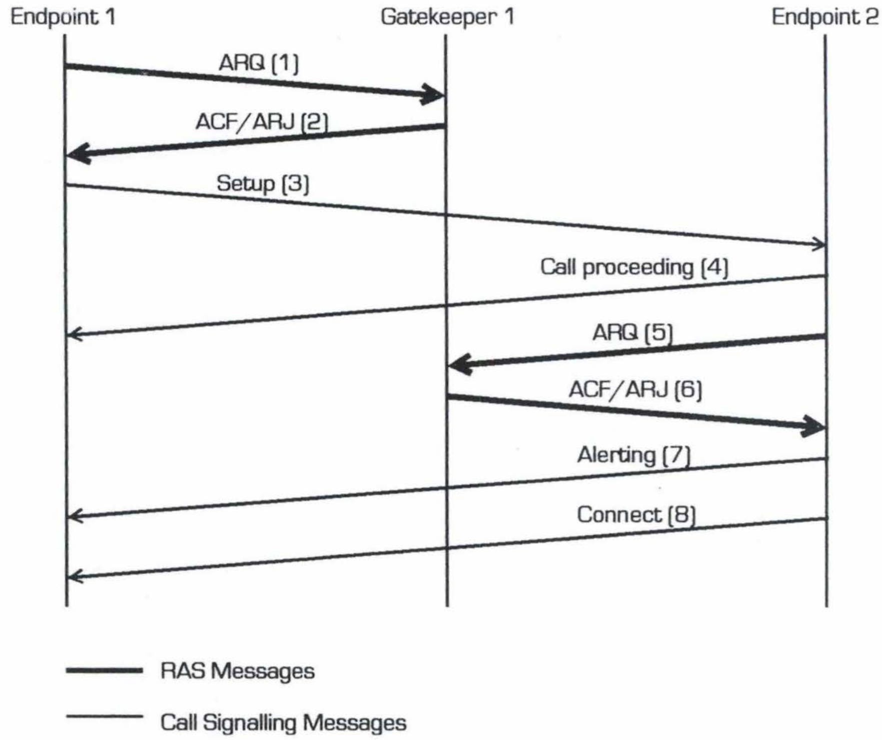

Figure 19: Direct call signalling, both endpoints registered

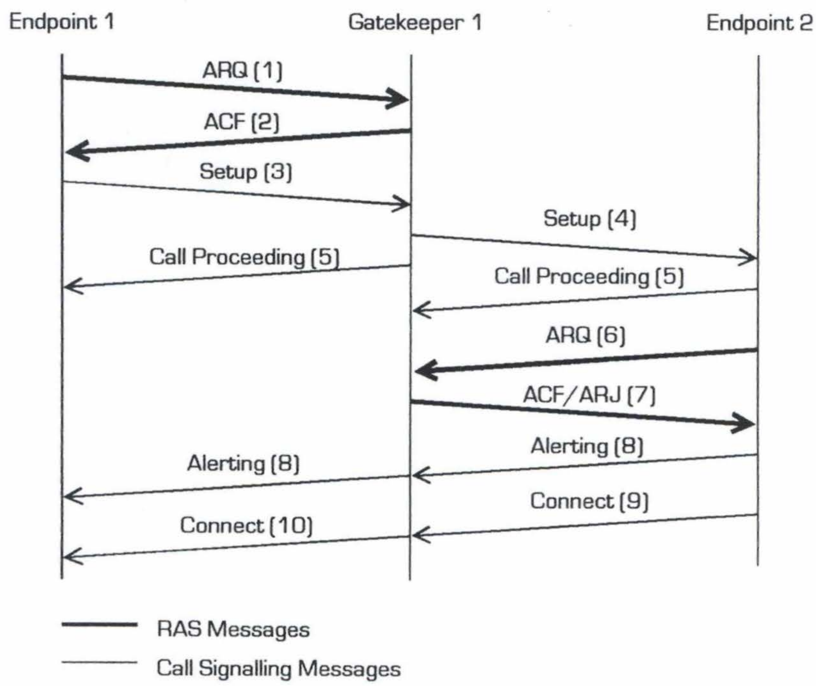

Figure 20: Both endpoints registered to the same GK, GK routed call signalling model

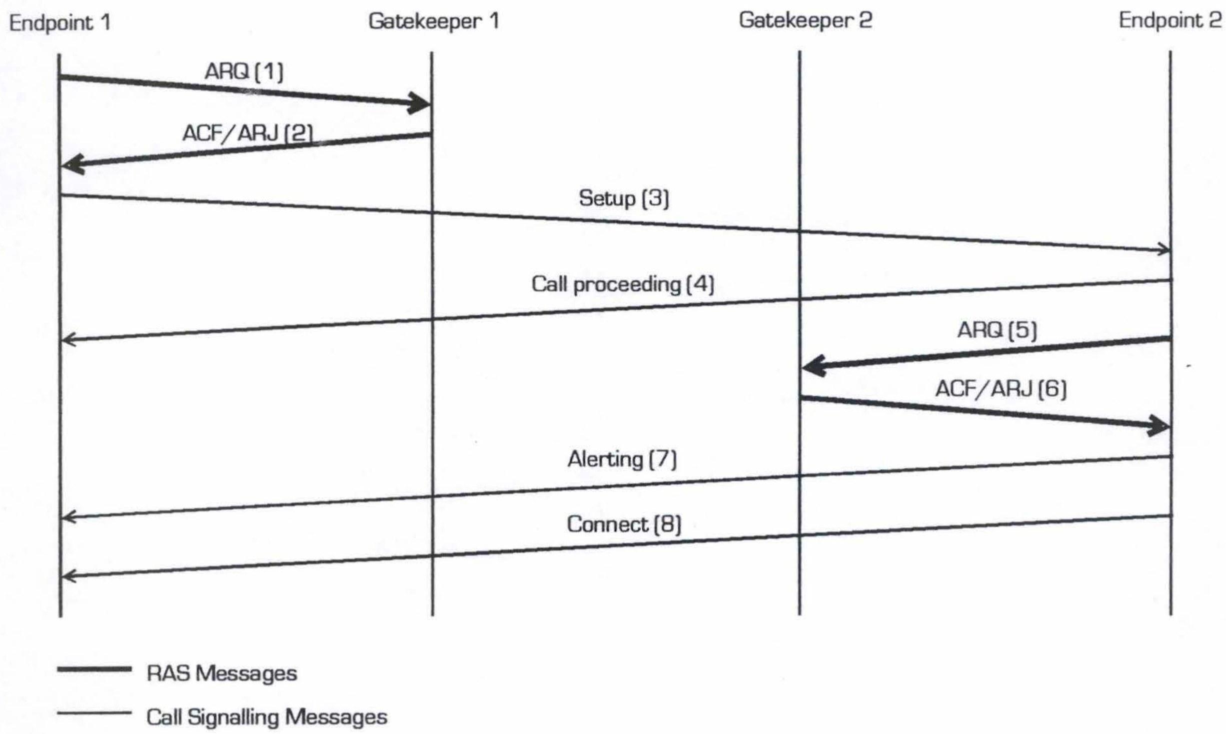

Figure 21: Both endpoints are registered, and both GK use the direct call signalling model

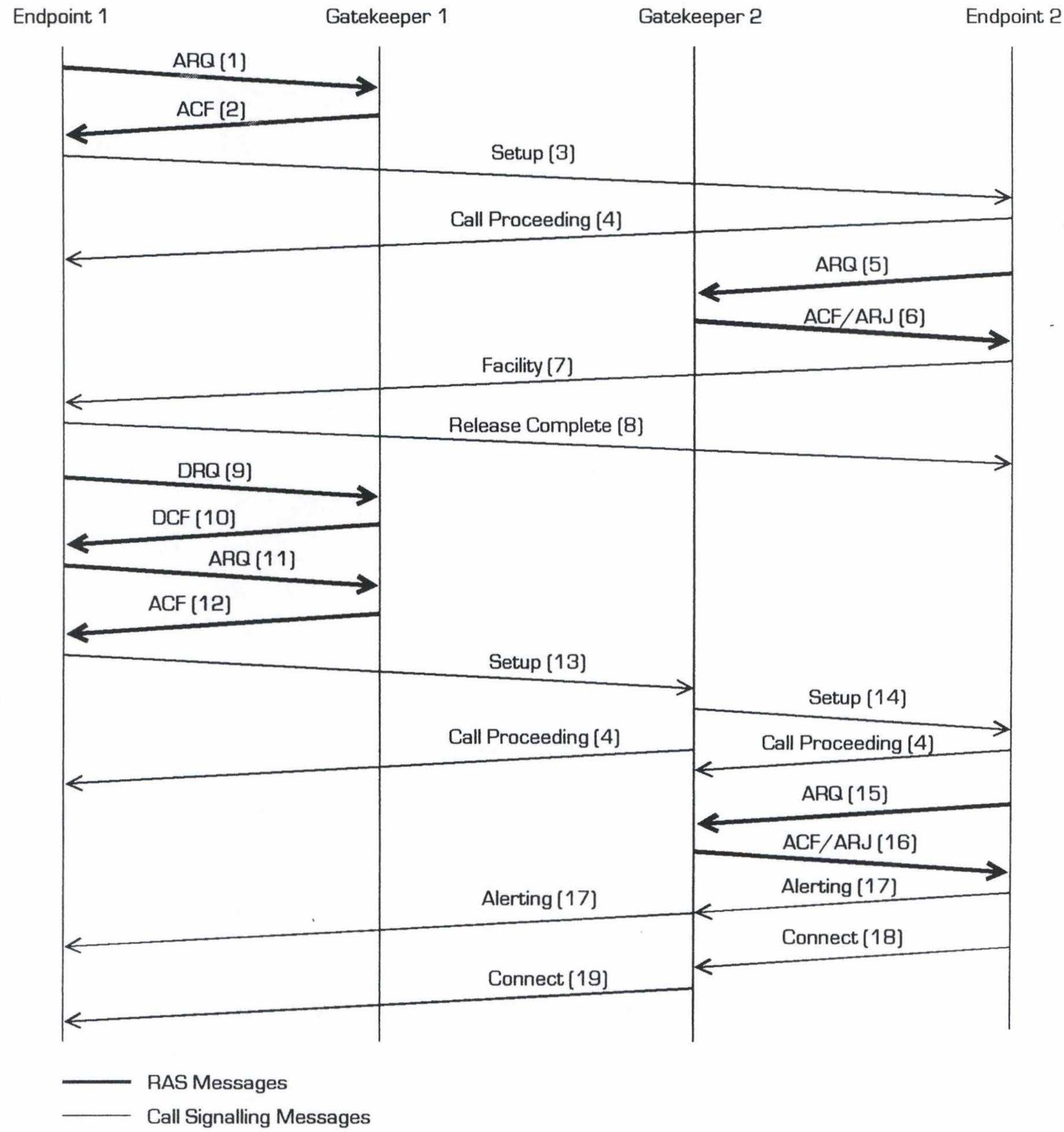

Figure 22: both endpoints are registered, but GK 1 uses direct call signalling and GK 2 uses gatekeeper routed call signalling

The FastConnect procedure:

The purpose of the fast start, or FastConnect procedure, is to allow endpoints to set up a call with a single round-trip message. It is up to the calling endpoint to initiate that accelerated procedure by sending a "setup" message containing the "fastStart" element. That structure contains a sequence of OpenLogicalChannel structures which describe all parameters to immediately open a media connection.

This procedure may be rejected by the called endpoint, and in this case the regular H.245 procedures are used. If it is accepted, the called party has to return Q.931 messages containing a fastStart element with the accepted parameters chosen among the propositions done by the caller. The called party may then begin its media transmission at once, even before the Q.931 message is received by the calling party: this one has so to listen on any of its initial proposition, and may stop to listen once it receives the Q.931 describing which was the accepted proposition.

1.2 SIP packet format and message exchange diagrams

Packet format :

SIP is a text-based protocol using SDP to carry session description parameters. A typical SIP message begins with the request or answer (error) code followed by relevant information: SIP headers that can be followed by SDP parameters.

As an example of SIP messages, we present here an INVITE request for a two-party call to illustrate the way a SIP message is organised. If the content-length is equal to 0, this means that no SDP information is carried.

```
INVITE sip:bob@c.domain.com SIP/2.0
Via: SIP/2.0/UDP b.domain.com
From: Alice <sip:alice@domain.com>
To: Bob <sip:bob@domain.com>
Call-ID: 3298420296@b.domain.com
CSeq: 1 INVITE
Subject: Bob, what about a meeting?
Content-Type: application/sdp
Content-Length: ...

v=0
o=dom 53655765 2353687637 IN IP4 128.3.4.5
s=Bob, what about a meeting?
c=IN IP4 b.domain.com
```

```
m=audio 3456 RTP/AVP 0 3 4 5
```

The INVITE request contains information proper to SIP call handling, and the SDP parameters about the call, such as the audio capabilities of the caller: in this case it is able to receive the following audio codings using RTP: 0 (PCMU), 3 (GSM), 4 (G.723) and 5 (DIV4).

SIP error codes:

SIP error codes are similar to those used for HTTP and are listed below:

100 Trying	Analogous to the Q.931 Call Proceeding
180 Ringing	Analogous to the Q.931 Alerting : the ucalled user has been located and its phone is ringing
181 Call forwarding	
182 Queued for service	The call has been put in a queue and waiting to be serviced
200 OK	The request has been successfully executed
300	Several possibilities exist for the same called name
301 Moved permanently	Contact field in the response tells the new location to try
302 Moved temporarily	New location is also returned. May be used as manual call forwarding
305 Use proxy	The call has to be handled through a proxy
380 alternative service	
400 bad request	There is a syntax errori the request
401 Unauthorised	The request requires user authentication
402 Payment required	
403 Forbidden	This is not a problem of authentication, but may mean that the callee does not accept the caller. The request should not be repeated
404 Not Found	
405 Method not allowed	The response must include an Allow header that tells the valid methods for the

	indicated address
406 Not acceptable	The called endpoint will generate reponses that will not be understood by the caller
407 Proxy atuthentication required	Similar to 401, but means that the client must first authenticate itself with the proxy before going further
408 Request timeout	
409 Conflict	Sent whenever a REGISTER request conflicts with existing registrations
410 Gone	Supposed to reflect a permanent situation. If not permanent, 404 should be used instead
411 Length required	The server needs a defined Content-length
413 Request Entity too large	
414 Request-URI too long	
415 Unsupported media type	
420 Bad extension	The server does not understand the SIP protocol extension
480 Temporarily unavailabe	The callee is temporarily not reachable (either because not logged or away)
481 Call leg/transaction does not exist	A BYE or CANCEL request has been received, that does not match any existing transaction
482 Loop detected	VIA path indicates that the same proxy has already been used
483 Too many hops	
484 Address Incomplete	
485 Ambiguous	The callee address provided in the requets is ambiguous and the respoense may contain a list of possible choices
486 Busy here	Called endpoint has been reached, but is not able to take additional calls. The call may be redirected to a voice mail service
500 Server internal error	
501 Not implemented	
502 Bad gateway	The server received a bad response from a downstream server or gateway
503 Service unavailable	
504 Gateway time-out	

505 Version not supported	
600 Busy everywhere	Returned if no endpoint (such as a voice mail service) is able to answer the call – see 486
603 Decline	The callee does not want to answer the call
604 Does not exist anywhere	The server knows very well that there is no such user, and that further searches are useless
606 Not acceptable	The callee would like to accept the call but is unable to do so for technical incompatibilities

Message exchange diagrams :

SIP can operate according to two different operation modes, either with use of a proxy or of a redirection server – which can be compared to the gatekeeper-routed and direct call signalling models of H.323 . Figure h presents a broad overview of the SIP proxy call model, while figure i presents the SIP redirected call model.

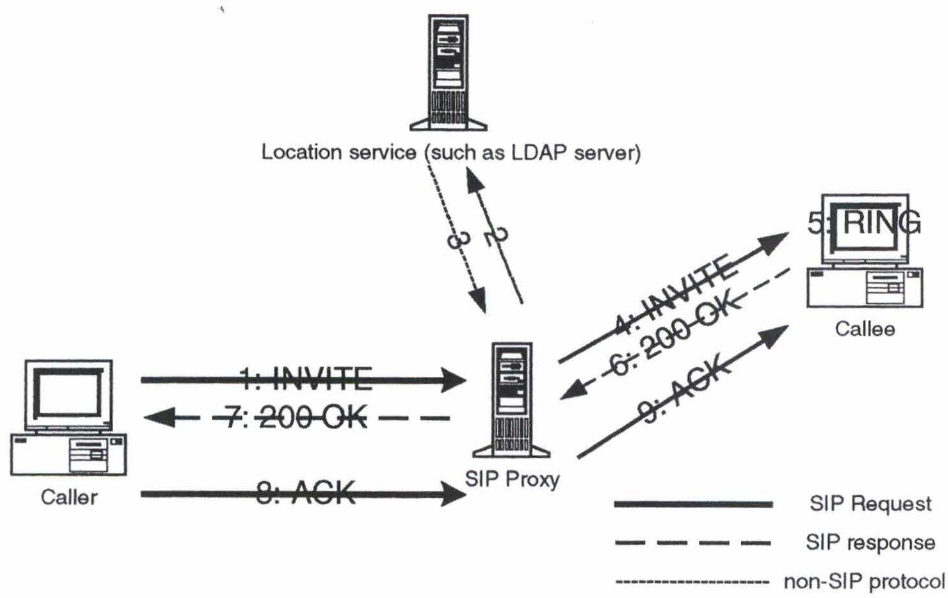

Figure 23: SIP proxy call model

452	No such signal in this package
453	No such statistic in this package
454	No such parameter value in this package
455	Parameter illegal in this Descriptor
456	Parameter or Property appears twice in this Descriptor
461	TransactionIDs in Reply do not match Request
462	Commands in Transaction Reply do not match commands in request
463	TerminationID of Transaction Reply does not match request
464	Missing reply in Transaction Reply
465	TransactionID in Transaction Pending does not match any open request
466	Illegal Duplicate Transaction Request
467	Illegal Duplicate Transaction Reply
471	Implied Add for Multiplex failure
500	Internal Gateway Error
501	Not Implemented
502	Not ready.
503	Service Unavailable
504	Command Received from unauthorized entity
505	Command Received before Restart Response
510	Insufficient resources
512	Media Gateway unequipped to detect requested Event
513	Media Gateway unequipped to generate requested Signals
514	Media Gateway cannot send the specified announcement
515	Unsupported Media Type
517	Unsupported or invalid mode
518	Event buffer full
519	Out of space to store digit map
520	Media Gateway does not have a digit map
521	Termination is "ServiceChangeing"
526	Insufficient bandwidth
529	Internal hardware failure
530	Temporary Network failure
531	Permanent Network failure
581	Does Not Exist

1.4 RTP/RTCP packet format

This section presents the format of RTP and RTCP packets. Figure 26 presents the RTP header format, while figures 27 & 28 presents

505 Version not supported	
600 Busy everywhere	Returned if no endpoint (such as a voice mail service) is able to answer the call – see 486
603 Decline	The callee does not want to answer the call
604 Does not exist anywhere	The server knows very well that there is no such user, and that further searches are useless
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Message exchange diagrams :

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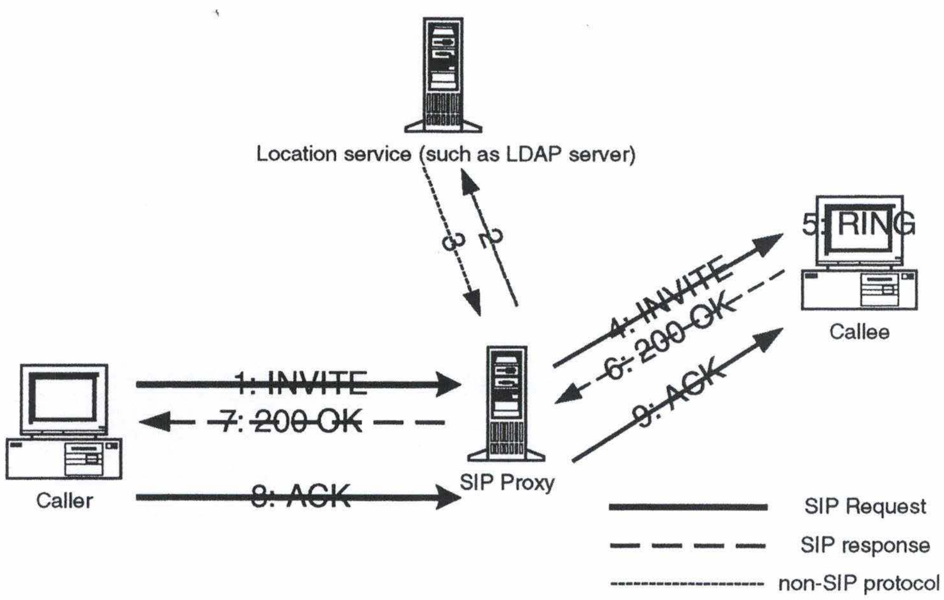

Figure 23: SIP proxy call model

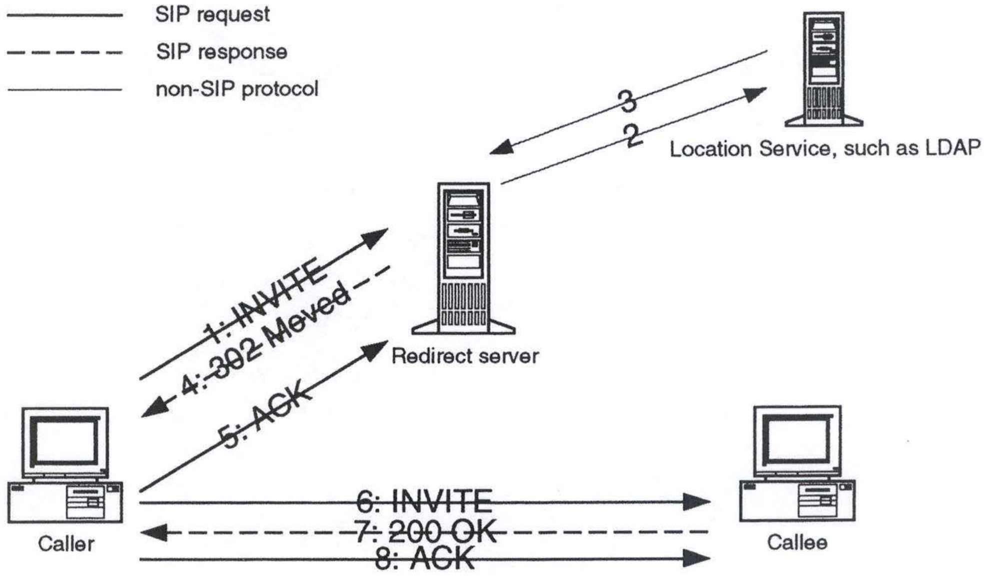

Figure 24: SIP redirected call

1.3 MGCP – H.248/megaco error codes

As we have told it when presenting the media gateway control protocol family, MGCP and H.248/megaco rely on the same principle but have differences in their implementation. This is notably the case for the protocol error codes which are listed below.

MGCP error codes:

200	The requested transaction was executed normally
250	Connections were deleted
400	Transient error; transaction not executed
401	Phone is already off-hook
402	Phone is already on-hook
500	Endpoint unknown
501	Endpoint not ready
510	Protocol error
511	Command contained unrecognised extension
512	Gateway not equipped to detect one of the requested signals
513	Gateway not equipped to generate one of the requested

	signals
514	Gateway could not send the specified announcement
515	Incorrect connection ID
516	Unknown call ID
517	Unsupported or invalid mode
518	Unsupported or unknown package
519	Endpoint does not have a digit map
520	Endpoint is restarting
521	Endpoint redirected to another MGC
522	No such event or signal
523	Unknown action or illegal combination of actions
524	Internal inconsistency in LocalConnectionOptions
525	Unknown extension in LocalConnectionOptions
526	Insufficient bandwidth
527	Missing RemoteConnectionDescription
528	Incompatible protocol version

H.248/megaco error codes:

400	Bad Request
401	Protocol Error
402	Unauthorized
403	Syntax Error in Transaction
404	Syntax Error in TransactionReply
405	Syntax Error in TransactionPending
406	Version Not Supported
410	Incorrect identifier
411	The transaction refers to an unknown ContextId
412	No ContextIDs available
421	Unknown action or illegal combination of actions
422	Syntax Error in Action
430	Unknown TerminationID
431	No TerminationID matched a wildcard
432	Out of TerminationIDs or No TerminationID available
433	TerminationID is already in a Context
440	Unsupported or unknown Package
441	Missing RemoteDescriptor
442	Syntax Error in Command
443	Unsupported or Unknown Command
444	Unsupported or Unknown Descriptor
445	Unsupported or Unknown Property
446	Unsupported or Unknown Parameter
447	Descriptor not legal in this command
448	Descriptor appears twice in a command
450	No such property in this package
451	No such event in this package

452	No such signal in this package
453	No such statistic in this package
454	No such parameter value in this package
455	Parameter illegal in this Descriptor
456	Parameter or Property appears twice in this Descriptor
461	TransactionIDs in Reply do not match Request
462	Commands in Transaction Reply do not match commands in request
463	TerminationID of Transaction Reply does not match request
464	Missing reply in Transaction Reply
465	TransactionID in Transaction Pending does not match any open request
466	Illegal Duplicate Transaction Request
467	Illegal Duplicate Transaction Reply
471	Implied Add for Multiplex failure
500	Internal Gateway Error
501	Not Implemented
502	Not ready.
503	Service Unavailable
504	Command Received from unauthorized entity
505	Command Received before Restart Response
510	Insufficient resources
512	Media Gateway unequipped to detect requested Event
513	Media Gateway unequipped to generate requested Signals
514	Media Gateway cannot send the specified announcement
515	Unsupported Media Type
517	Unsupported or invalid mode
518	Event buffer full
519	Out of space to store digit map
520	Media Gateway does not have a digit map
521	Termination is "ServiceChangeing"
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529	Internal hardware failure
530	Temporary Network failure
531	Permanent Network failure
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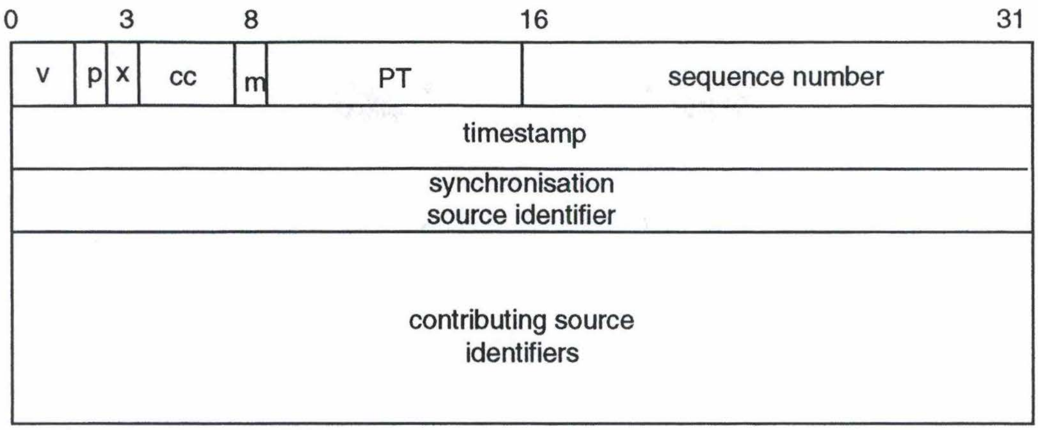
1.4 RTP/RTCP packet format

This section presents the format of RTP and RTCP packets. Figure 26 presents the RTP header format, while figures 27 & 28 presents



v: version, currently = 2
p: padding bit
x: extension
cc: CSRC count
m:marker
PT: payload type

Figure 25: RTP header format

V	P	RC	PT=SR	Length
SSRC of sender				
NTP timestamp, MSW				
NTP timestamp, LSW				
RTP timestamp				
Sender's packet count				
Sender's octet count				
SSRC_1 (SSRC of 1st source), block 1				
Sender's packet count				
Fraction lost			Cumulative number of packet lost	
Extended highest sequence number received				
Interarrival jitter				
Last SR (LSR)				
Delay since last SR (DLSR)				
SSRC_2 (SSRC of second source), begin block 2				
...				
Profile-specific extensions				

Figure 26: RTCP Sender Report Packet Format

V	P	RC	PT=RR	Length
SSRC of sender				
SSRC of first source, begin block 1				
Fraction lost			Cumulative number of packet lost	
xtended highest sequence number received				
Interarrival jitter				
Last SR (LSR)				
Delay since last SR (DLSR)				
SSRC_2 (SSRC of second source), begin block 2				
...				
Profile-specific extensions				

Figure 27: RTCP Receiver Report Packet Format

On those last figures, V means « version », P means « padding », PT means « payload type » and is either equal to SR (Sender Report, 200) or RR (Receiver Report, 201).

2. Software design for the model proposition of part II.

This annex presents the characteristics of a possible software implementation of the model proposition exposed in part II: software design, database schema and function specifications.

Due to a lack of time, this implementation could not be realised.

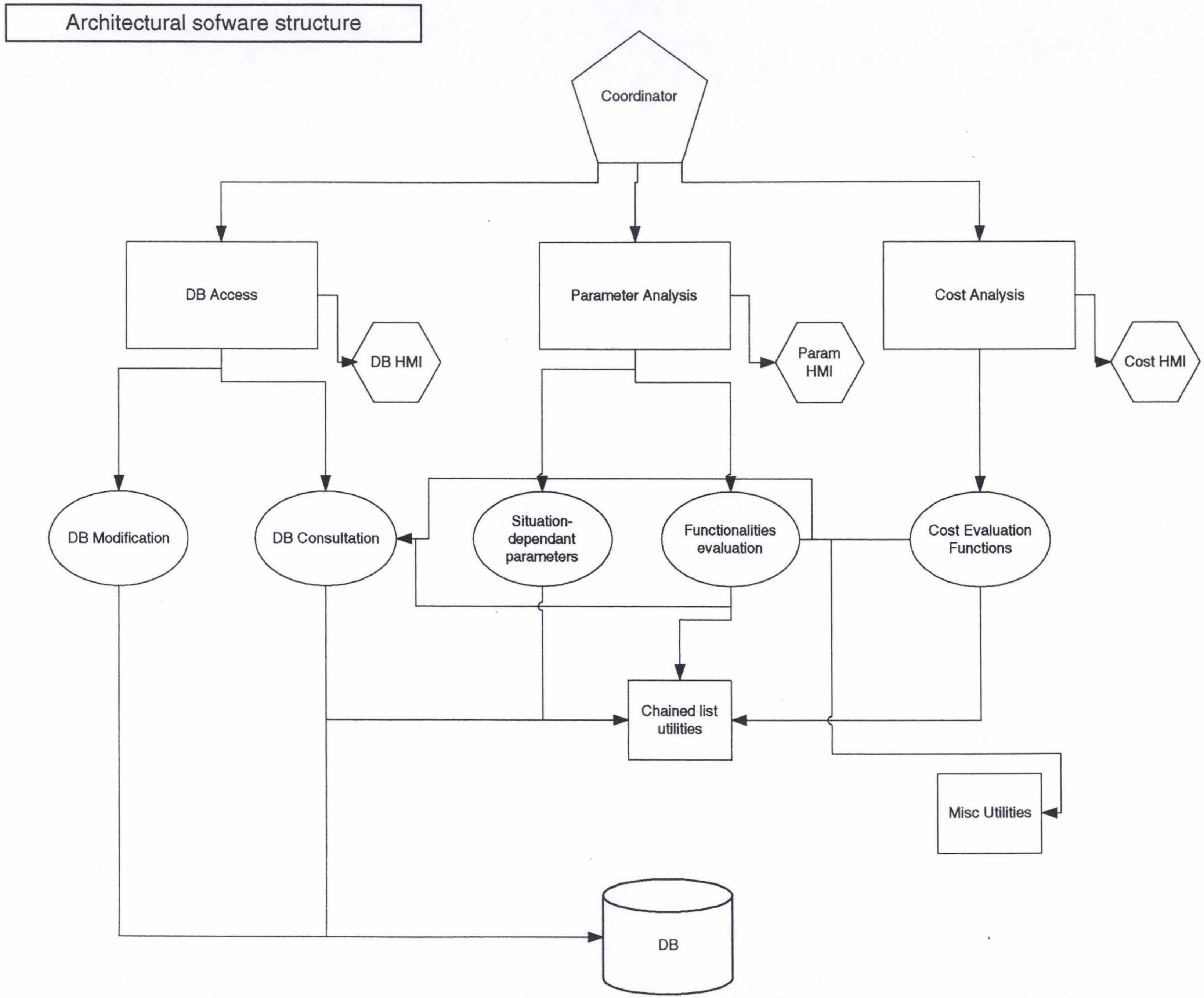


Figure 28: Architecural design

2.2 Database schema

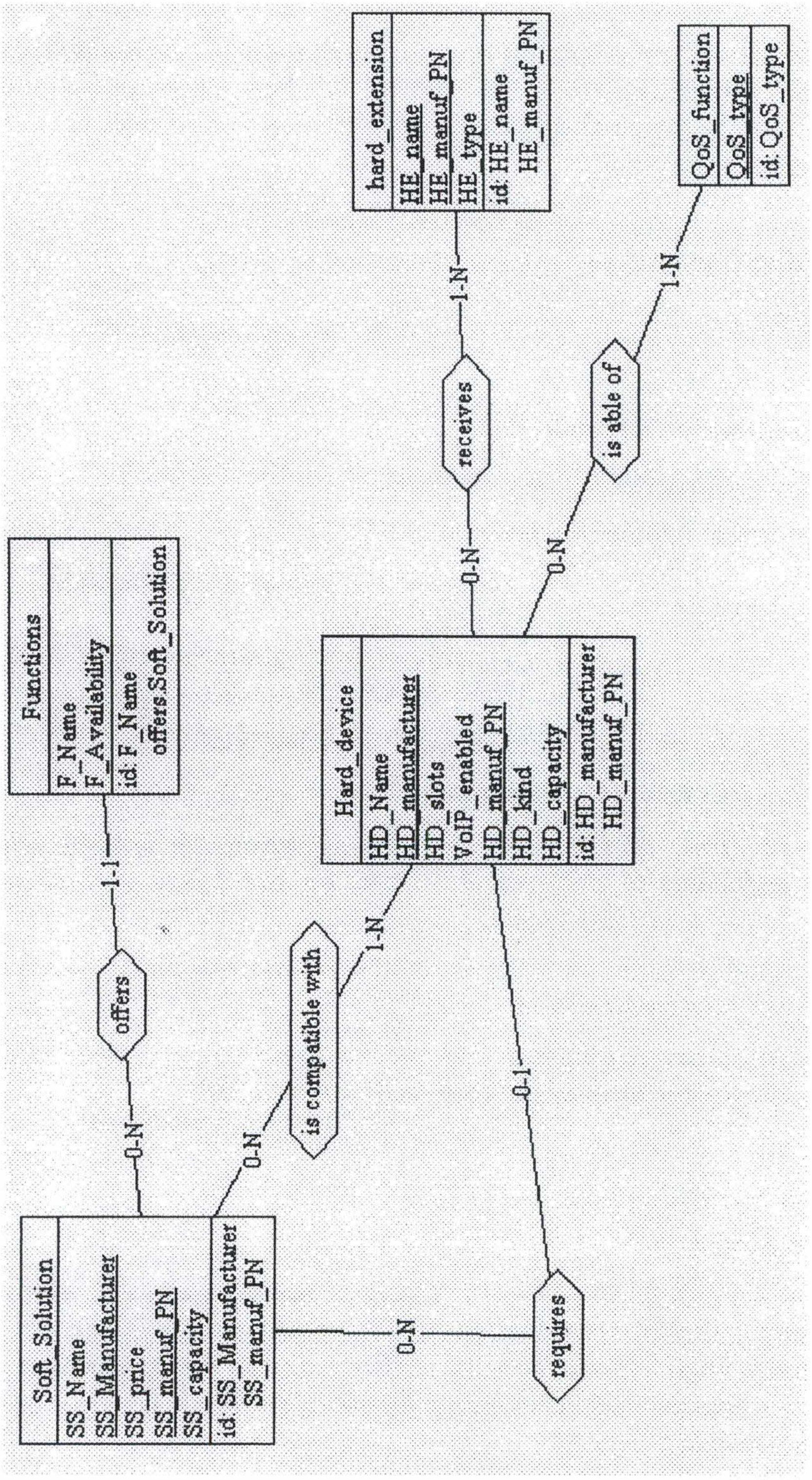


Figure 29: DB schema

Some constraints are associated to this schema, primarily constraints on the possible values for specific attributes, such as HD_KIND which should be either *router, PBX, server* or *switch* according to the kind of hardware device that may met in VoIP networks. QoS_Type has a constraint of the kind, here it should be one of the following; *ToS, WRED, RED, WFQ, RSVP, 802.1p/q, CRTP, RTP_MTU.*

The "receives" relationship means "the hardware extension is compatible with this hardware device and may be used on it", and does not refer to the hardware extensions actually put in the device.

2.3 Specifications of the software functions

Purposes of the modules shown on the architecture diagram.

- Parameters evaluation modules
  - Functionalities evaluation module

    This module has for purpose to:

    - *collect the user's preferences on telephony functionalities as well as VoIP-specific functionalities (protocol supported, QoS features, interoperability needs)*
  - Situation-dependant parameters evaluation module

    This module has for purpose to:

    - *at first, identify the current situation. A simple single function is sufficient to perform this task, on the basis of a closed choice (radio buttons in HMI terms)*
    - *at this stage, we need to collect the environmental parameters related to the chosen situation*
    - *those environmental parameters yield some questions according to the graphical view of the cost evaluation model*

- Database access modules
  - DB Modification

    This is a module giving write access to the database. It will provide functions to insert new software solutions or hardware devices in the database, as well as to modify their characteristics, either intrinsic or dependant devices (such as giving a new extension card to a hardware device). Deletion of DB entities will also be possible.
  - DB Consultation

    This module will be used to get information from the database while evaluating a solution. It will provide functions to retrieve the database contents and put it in chained lists on which to work on. Functionalities of a solution will be put in alphabetic order in the list (function "put" for chained lists will do that automatically).

- Cost Analysis module

  This module will get information from the parameters evaluation modules in order to give the estimated cost of the whole telephony solution related to its scores (functional & technical).

  Budgets for installation, maintenance, skills to acquire will be asked to the user. The application will provide him with some indications about those costs, such as the points to estimate and possible values.

Some implementation details

The choice made by the user about the importance of each parameter, either functional or technical, will be temporarily stored in memory. This will be done with records, each of these matching a parameter name and its value. That value is stored as an integer.

For functionalities, this integer will reflect the importance of the parameter to the user, according to the following table:

useless ≡ 1
not essential ≡ 3
essential ≡ 10.

The availability attribute will be used to give its mark to the functionality. If it is available within 6 months, the given mark will be the one shown here above. The mean-term availability (between 6 and 12 months) will made the given mark being the one shown here above INT(<mark>/2). The long-term availability (more than one year) will made the given mark being 0.

For technical parameters, the value is given by the user according to a choice he will be invited to do. Each choice will have its own procedure in the software. Choices are to be made for the following parameters groups;

Quality of Service: need for QoS and if yes relative importance of RSVP, RED, WFQ, WRED, IP ToS, RTP_MTU, CRTP, 801.2p/q.

Security features: authentication modes (PIN code, card), encryption means (IPSec, S/MIME, H.235).

Reliability: minimum of MTBF expected, maximum desired, in hours. The score of the reliability will be computed on that basis. A score of zero will be given to devices having a MTBF equals or smaller than the minimum expected, and a score of 15 will be given to devices having a MTBF equals or greater to the maximum expected. The remaining 15 points are given to the maintenance easiness factor.

Description of the functions implemented by the modules

1) Situation-dependent parameters module

   a) HMI-related functions

   Functions specified here are specific get-like functions for each form associated to a particular form. Each of these forms (windows) will of course have initialisation and destruction functions.

   Forms are:

   - *SIT_MENU, which asks the user for its situation*
   - *SIT1, which asks the user for situation1-specific parameters*
   - *SIT2, which asks the user for situation2-specific parameters*
   - *SKILLS, which asks the user for skills to acquire and their related cost. Skill has here a broad sense, meaning formation and wages costs. They are related to the maintenance cost.*

   There is not SIT3 or SIT4 form: situation three will cause the call of both SIT1 and SIT2 while situation four is not relevant for specific questions.

   i) Function **id_situation** [associated to SIT_MENU form]

   _ → integer

   pre : window initialised

   post: the integer value is either
   1 for "telephony network only"
   2 for "data network only"
   3 for "both networks"
   4 for "no network"

   ii) Function **get_sit1_param** [associated to SIT1 form]

   _ → boolean x boolean

   pre : window initialised

   post: first boolean value is for the amortisation completion and the second boolean is for PBX usability as GSTN gateway.

   iii) Function **get_sit2_param** [associated to SIT2 form]

   _ → boolean x boolean

   pre : window initialised

   post: first boolean value is for the VoIP-enabled character of the network, and the second one for the network upgradability.

b) Processing functions

i) Function **analysis** (code:integer): sit1 x sit2

Integer → ?

Pre: code ∈ {1,2,3,4}

Post: the output contains all relevant information for cost estimation, which is detailed below.

This function shall perform the following tasks:

- *if code = 1:* ° *is the amortisation of the PBX completed?*

° if this amortisation is not completed, is the PBX usable as PSTN gateway?

- *If code = 2:* ° *is the current network VoIP enabled?*

° if the network is not VoIP enabled, is it possible to upgrade it? With same manufacturer equipment? With compatible equipment?

- *If code = 3: same questions as for code 1 and 2.*
- *If code = 4: no specific questions*
- *In either case, ask for skills to acquire and their estimated cost, both for VoIP and PBX-only case, whatever the situation is. The final aim is to compare the costs of VoIP and PBX, so we need that information in both cases.*

Code 1 and 2 have each one their own sub-function. Code 3 will call those two functions. Code 4 means nothing to do. Situation-specific functions will be followed by the skill to acquire function.

ii) Function **tel_net_only** () : sit1

_ → sit1

Pre: /

Post: the output record S1 has two boolean fields: Amort & PBX_GW. If this amortisation of the PBX is completed, then S1.Amort = 1. If the PBX is usable as a GSTN gateway, then S1.PBX_GW = 1.

Type:
```
sit1 = record of Amort: boolean;
                     PBX_GW: boolean
       end;
```

Calls: SIT1 form, get_sit1_param

iii) Function **data_net_only** () : sit2

_ → sit2

Pre: /

Post: the output record has two boolean fields: VoIP_en & upgrad. If the network is VoIP enabled, VoIP_en = 1, and upgrad = 0. If the network is not VoIP-enabled but upgradable, VoIP_en = 0 and upgrad = 1.

Type:
```
sit2 = record of VoIP_en: boolean;
                          Upgrad : boolean
       end;
```

Calls: SIT2 form, get_sit2_param

iv) Function **both_nets** () : sit1, sit2

_ → sit1 x sit2

Pre: /

Post: the output is the two records given by tel_net_only and data_net_only.

Calls: SIT1 and SIT2 forms, get_sit1_param and get_sit2_param

2) Functionalities module

a) HMI-related functions

i) Function **get_choice** (): func list

b) Processing functions

i) Function **func_choice** (): func list

This will uses HMI_func_choice to ask the user for functionalities evaluation.

Pre: /

Post: the output is a list of functionalities names and importance

Type:
```
func = record of name: string;
                 imp : integer
       end;
```

3) Database access module

a) Function **get_solution_func**(name, manuf : string; PN:integer): func list

Type:
```
func = record of name: string;
                 avail: integer
       End;
```

Gives a list of functionalities of a software solution

b) Function **get_extensions** (name, manuf: string; PN: integer): ext list

```
Type: func = record of HE_name: string;
                       HE_PN: integer;
                       HE_type: string;
           End;
```

Gives a list of available extensions for a specific hardware device

c) Function **get_harddev_qos** (name, manuf: string; PN: integer): string list

Gives a list of QoS implemented on a hardware device

4) Database modification module

a) Function **new_solution**

Allows to enter a new software solution in the database, with its offered functions. According to the database structure, the user needs to enter compatible and required hardware devices. If the specified device does not exist in the database, the user will be invited to enter the device's characteristics (call to new_harddev and exists_harddev – internal function to this module.)

Sub-function: exists_harddev (name, manuf: string; PN: integer): boolean

b) Function **new_harddev**

Allows to enter a new hardware device in the database, along with its QoS features and possible extensions. This will be done by adding the extension afterwards (new_hard_ext). If the extension already exists, a new matching entry will be created (new "receives" relationship).

c) Function **new_hard_ext**

Allows to enter a new hardware extension for a hardware device. If there is no such hardware device in the database, it will first call new_harddev.

d) Function **modif_harddev** (name,manuf: string; PN; integer)

Allows to modify an existing hardware device, by either modifying its QoS features, hardware extensions or direct characteritics.

e) Function **modif_soft_func** (name, manuf: string; PN: integer)

Allows to modify an existing software solution: functionalities, compatibility with hardware devices, requirements.

f) Function **del_harddev** (name, manuf: string; PN: integer)

Allows to delete a hardware device. It will also delete relationships with hardware extensions and QoS functions. If either of those is no more related to any hardware device, it will also be deleted.

g) Function **del_QoS** (name: string)

Allows to delete a QoS function.

h) Function **del_hard_ext** (name, manuf: string)

Allows to delete a hardware extension.

i) Function **del_soft_func** (name, ss_name, manuf: string; ss_pn: integer)

Allows to delete a software (i.e. telephony) function.

5) Cost evaluation module

Parameters to be given to this module: situation code number, sit1 and sit2 records

a) HMI-related functions

i) HMI_inst_cost

(1) Function get_inst_cost

_ → *integer*

pre: window initialised

post: returns the total cost of installation

ii) HMI_equip_cost

(1) Function get_equip_cost

_ → *integer*

pre: window initialised

post: returns the total cost of equipment

iii) HMI_unifmsg_ben

(1) Function get_unifmsg_ben

_ → *integer*

pre: window initialised

post: returns the estimated benefit due to the unified messaging

iv) HMI_skills

(1) Function **get_skills**

_ → *skills_array*

pre: window initialised

post: each skill record of the array contains a name and a cost. Names are: VoIP_support, PBX_support, VoIP_formation, PBX_formation. If no skill or formation is required, associated cost is zero.

Type: `skills_array = array[1..4] of skills;`

v) HMI_ext_maint

(1) Function get_ext_maint_cost

_ → *integer*

pre: window initialised

post: returns the estimated cost of outsourced maintenance.

b) Processing functions

i) Function install_cost (): integer

Calls HMI_inst_cost to retrieve information from the user. Installation cost is dependent from the parameters:

- *kilometres of cable to install*
- *hardware devices to install*
- *initial configuration*

ii) Function equip_cost(code: integer; S1: sit1; S2: sit2) : integer

Calls HMI_equip_cost to retrieve the equipment cost. This equipment cost is based on information retrieved by the situation-dependent analysis module, which may lead to different possible choices. At this moment, the marks obtained by each of those solutions may be presented to the user to help him do the choice.

According to S1 and S2 information, questions and propositions about the equipment to choose will differ.

For example, if S2.VoIP_enab = true, the application will need to know what is the current equipment of the user and give him the according hardware extension. It also depends of interoperability needs (if the user requires PRA interoperability, it is useless to propose POTS modem cards).

iii) Function maint_cost

Calls HMI_maint_cost to retrieve the maintenance cost from the user. Maintenance cost is dependant from the following parameters:

Either external or internal (outsourced) maintenance: initial choice to make with checkboxes. At least one must be checked.

If internal, skills to acquire : call to Skills function.

If external, cost of the maintenance outsourcing: uses HMI_ext_maint.

A mix of external and internal maintenance is also possible. In this case, both forms are shown.

iv) Function Skills () : skills list

_ →skills_array

Pre: /

Post: the output list is an array of required skills and related cost, as described here above.

Type:
```
skills =        record of  skill : string;
                           cost  : integer;
                end;
```

Calls: SKILLS, get_skills

This function will ask the user for skills to acquire, whatever the situation is. If there is already a data network, an additional person may be required to support the VoIP tasks.

v) Function Total_cost (install, equip, unfmsg, maint: integer)

install x equip x unfmsg x maint → _

pre : all parameters have a non-zero value

post: the user has the total cost of the chosen telephony solution

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Hot links about VoIP: http://www.protocols.com/voip.htm

VoIP Gateway software offers an integrated MGCP and H.323 based solution for enterprise and carrier networks supporting Voice over IP, Fax over IP and Internet Offload Applications. : http://www.hssworld.com/products/voip/VoIP_home.htm

VoIP: Title Page: http://www.bell-labs/mailing_lists/iptel

Companies :

http://www.cisco.com;

http://www.lucent.com;

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