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VoIP: The Convergence of networks

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The development of telecommunications, especially of the Internet, has made possible the use of technologies such as IP (Internet Protocol) telephony for both business and leisure. The problems arising from the diversity in number of existing telecommunication nets are motivating the study of systems to promote the homogenisation of means of voice and data transport. This paper presents the VoIP (voice over Internet Protocol) solution as a likely solution for this problem. It also develops a first approach to the concept and terminology of net convergence, and later, it establishes a comparison between IP telephony and conventional telephony. The paper also supplies a detailed analysis of requirements of IP telephony, of its legal situation and of the different standards used in its development.

1. INTRODUCTION

The convergence of the present telecommunications nets implies finding the suitable technology to use the same line for voice and data. This obliges us to establish a model or system that enables us to "pack" voice so that it can be transmitted along with data. Taking into account that Internet is the "net of nets", the development of a worldwide technology leads us clearly to the Internet Protocol (Goncalves, 1998), and to find a method to transmit voice and data on that Protocol. This problem has an "easy" solution: VoIP (voice over Internet Protocol).

The aim of this article is to analyse whether the path to convergence is appropriately designed, to discover its advantages and disadvantages and to present the conclusions of a real situation where the latest technology has been used. If convergence is possible, the interpretation of this article and the adaptation to different situations today can be the first step to prepare for future scenarios.

This is not as simple as might appear, and to prove this we only have to consider the evolution of the different commercial developments, standards, nomenclatures and acronyms that experts in the field use.

This article, based on a real VoIP experience, tries to analyse the present situation regarding previous studies and investigations, to be able to offer future options and their application to business reality.

Although different investigations on advanced algorithms for voice digitalisation have been known since 1970 (VoIP'99), is was not until February 1995 when the company Vocal Tec (Canto, 1999) marked the beginning of the race showing, with its product Internet Phone, the actual possibilities of establishing telephone calls from PC to PC a software packet installed in the PC was used and the transmission system was Internet. Thus surged what is today known as IP telephony.

Its evolution was unstoppable and in 1996 the first calls from a telephone to a PC were made (VoIP'99) and from telephone to telephone. From 1997 on, new appliances and methods started to appear, which led us to keep up to this day the term XoIP ("X" over Internet Protocol) (VoIP'99) as the real future option, or better, as the door to the convergence of nets. In this acronym "X" stands for any content that can be transmitted over a net (D= data, V= voice, F= fax, M= multimedia, etc)

This maze of technologies, commercial interests and future options leads, as every "revolution", to the confusion and weariness of the public in general. The immediate consequence is the habitual FAQs (Frequently Asked Questions): why IP? What is the difference between IP telephony and voice on IP? Is VoIP the same as VoFR (Voice on Frame Relay)? What does XoIP really mean? etc.

It is therefore necessary to define in a clear and simple way the present situation to be able to identify clearly both the terms and the elements that take part in the different levels of development in the convergence of nets. Some terms that possibly identify the way to VoIP services are:

• Telephony: Telecommunication services supplied on the Commuted Telephonic Net, on both the Basic Telephonic Net and the digital Net of Integrated Services, except data communication.

• Voice through Internet: Telephony services supplied on the global public net made up by the interconnection of nets of communication of IP based packets.

• Voice on IP: Telephony services supplied on "private" IP nets, without interconnection with the Commuted Telephonic net.

- Voice on Frame Relay: Telephony services supplied on nets supported on Frame Relay circuits, orientated to data transmission.

- Voice on Asynchronous Transfer Mode: Telephony services supplied on ATM nets, where the possibility of quality of service is offered.

- Multimedia on IP: multimedia services (video, audio, image, etc) supplied on IP nets.

- Fax on IP (FoIP): Fax transmission services supplied on IP nets.

- XoIP: in global terms "everything on IP". The letter "x" can be substituted by any other letter that identifies any service on IP nets (F= fax, M= multimedia, V= voice, D= data, etc.)

In conclusion, we can deduce that if the future is IP (due, mainly, to its present coverage, its acceptation by users and the imminent arrival of the protocol IPv6), and if X is the global integration of all the present and future services, XoIP is the real way that can open the doors to the Convergence of Nets. This convergence will imply, in economic terms and authentic "revolution", which will affect both the business world and homes. The reduction in expenses on every area can be considered extraordinary.

The next sections present a comparison between IP and traditional telephony and a description of the advantages and disadvantages of IP services. Section 4 shows a successful application of IP telephony in the commercial area and finally the conclusions are presented.

2. IP TELEPHONY VS. TRADITIONAL TELEPHONY

Although IP Telephony takes advantage of the existing telecommunications facilities (Figure 1), it also needs new elements, as shown in Figure 2.

In Figure 1, we can see the situation today, an environment where the nets of an organisation co-exist. On one hand there is a data circuit and alongside a voice circuit.

Conversely, in Figure 2, it can be appreciated how both nets can be unified, with the addition of the Voice on IP Gateway (VoIP GW), and therefore the convergence is achieved.

IP telephony needs a device which transforms the voice waves into digital data and besides divides them in packets that can be transmitted using the IP protocol. This device is known as Digital Signal Processor (DSP), which is already available, and which use IP telephones or the same gateways that transmit the IP packet once the voice has been packetised. When the packets reach the gateway, the inverse process occurs through the DSP and thus the receiver will be able to receive the analogical signal of the sender's voice.

The transmission of packets of voice in the aforementioned system is similar to the transmission of an e-mail from origin to destination. The problem is that IP transmissions can be unsuccessful, and if the mail is not legible or if a packet "gets lost" it is necessary to request a new transmission and its recovery is possible. However, this is not the case with transmission of voice, as the necessity to receive the packets in the correct order, the necessity of ensuring there is no loss and of getting a minimum rate of transmission make the implementation of quality-of-services systems essential. These systems are nowadays a big challenge for the industry, and the guarantee of quality of service over IP will mean the immediate implementation of voice transmission systems.

All in all, the actual problem nowadays is that Commuted Telephony establishes a dedicated virtual circuit between the origin and the destiny and so the quality is undeniable and safe, on the contrary the transmission of voice over IP shares the circuit and the width of band with the data, and the packets may cross many nodes before reaching its destiny, which causes deficiencies in the transmission of the packets of voice.

Next, some other questions referring to this technology are considered. They must be taken into account when the actual implementation of an IP telephony system for commercial or professional use is carried out.

2.1 Required bandwidth

Until recently the bandwidth required for the transmission of voice and video in real time was considerably high, which made this sort of communication impossible over data nets that didn't guarantee quality of services such as Internet or nets based on IP protocol.

At present, the voice received by a gateway is digitalized and compressed by different algorithms (GSM, G.723.1, G.711, G.729) (VoIP'99) whose main characteristic is to be able to obtain bigger compression ratios to the detriment of latency time (the necessary time to uncompress the voice so that it can be understood). Some algorithms manage to compress the packets of voice in approximately 8 Kbps. The IP protocol adds the packet a series of checked sums for its suitable transport in a net, which increase the necessary width of band up to 16 Kbps.

We also have to take into account the parameter called "silence suppression" (VoIP'99) when this parameter is activated, it is possible to reduce the transmission of packets (use of the width of a band) to those situations when the agents are speaking. The rest of the time (when there is no voice to transmit) the bandwidth is available. Taking this into account it can be affirmed that the average size of a packet of voice during a conversation is 8 Kbps.

Taking all the previous discussion into account it can be seen that on a B channel of any ISDN (Integrated Services Digital Network: 2B channels and lD channel) whose bandwidth is 64 Kbps a communication of eight simultaneous calls can be supported. This situation coincides with the size of the switchboard of any small or middle-size company. This demonstrates that the bandwidth requirement for this kind of applications is affordable by nearly every company.

2.2. Quality in the transmission of voice

Regarding the quality in the transmission of voice, all manufacturers and research refer to three determinant factors (Minoli et al., 1998):

• Voice encoders: they influence the digitalisation of voice in packets of data that contain the voice and that will be transmitted over the IP net. They also have an influence due to the delay necessary for the decompression of the packet of voice, which adds an extra delay to the communication.

• Cancellation of echo: this is a necessary requirement for any communication over IP telephony. It eliminates automatically and in real time, any possible echoes without this feature the communication would be unintelligible.

• Latency: it is necessary time to allow the voice to travel from one end to the other, including the time to compress transmit and uncompress. This amount of time tends to reduce, but it will never be suppressed at present, the latency time is 120ms approximately.

2.3 Standards

At present, there are standards that regulate this kind of communication. They were set by international organisations, such as the International Telecommunications Union (ITU), which has established some rules for the interconnection of the different elements that take part in a communication over IP telephony.

The standard that rules this kind of communication is the H.323 by the ITU (ITU standards, 1998). This rule is actually a set of rules for the transmission of multimedia data (audio, video, data) over nets that do not guarantee a quality of service (IP networks).

The functions covered by H.323 are about the control of calls, the use of voice encoders and rules by other organisations that specify the transmission of packets of voice in real time.

The H.323 protocol has been adopted by nearly all leading companies in this field, such as Netscape, Microsoft, Intel, and Vocaltec. The adoption of this standard makes possible the interconnection of equipments and software of any manufacturer that has adopted it.

It is then a logical conclusion that any company that wants to work in VoIP services must adopt this standard in all their developments. This will guarantee a perfect integration with hardware and software platforms from different manufacturers those products follow the rule.

2.4 Government regulations

The governments are concerned about the voice services over IP, due to the fast growth of Internet and the possibility of offering a number of services of an added valve on this net, such as IP telephony.

The European Union (EU) has established different criteria that IP telephony has to fulfil before being under these regulations (Colchero, 1999):

• The communications must be available commercially.

• The communications must be supplied to the general public.

• The communications must have their origin and destiny in sites of the commuted telephonic public net.

• The communication must imply transport and commutation in real time.

The European Union considers these four criteria are not fulfilled at present, and that's the reason why the European Union has decided not to regulate IP telephony. This shows that the EU has underestimated the possibilities for development of this technology, as these characteristics are already possible with the present technology.

It is important to make clear that this legal situation is liable to change any moment according to the criteria of the competent regulating organisations.

2.5. Applications

With all the aforementioned, a series of highly demanded applications can be set in motion, which will generate significant savings in prices.

Call centres:

Call centres can use IP telephony, improving the quality of the information exchanged in each session. For example, a user could navigate over on-line information before consulting and operator, the user could work with a document shared over the screen. In this way, it is possible to obtain systems with high quality services, and to reduce considerable the expense on telephonic lines and on Automatic Call Distributors (ACD).

Private virtual voice networks:

This application consists of the interconnection of the telephonic switchboards over the corporate IP network, so as to be able to make a call from an extension in office A to another extension in office B over the company's data network. This call is free of charge as is uses the existing data infrastructure. A clear example of this service would be banks and their network of branches.

Call centres over the WEB:

If a company has its information available in a Web in Internet, the users that visit that Web will not only be able to visualise the information that company offers, but it could also establish a communication with a person in the Sales Department without cutting off the connection. When the Sales Operator answers the call he will have on his screen the same information the user has. This application has the following advantages:

• It is a call over Internet, so it won't have any additional cost for the user as it uses the call established to transmit data to transmit voice also. This will allow the companies to offer a service similar to the "lines 900".

• The user will be able to stay on-line while he speaks with a Sales Operator.

• The customer deals with human operators who can advise him. This feature will undoubtedly improve the results of an electronic commerce system.

• The operator can close the sale in the easiest way. The users are quite unwilling to give the details of his credit card in a Web page due to well-known security problems. However there will be no objection to give those details to a Sales Operator, as the user has the full guarantee that his details are protected.

FAX applications:

It is possible to transmit faxes over IP telephony networks in the same way as voice. In this way the cost in fax transmission of a company can be significantly reduced. In this case, the user that receives the fax has not need to use special equipment, as he will not have to receive the faxes through a conventional fax machine. A typical application of this feature is the massive remittance of faxes. The user will only send one copy of the fax and a list of the recipients' telephone numbers and the system will send them diverting the faxes to the sites from where the call is the most economical.

Multiconference:

IP telephony allows the connection of three or more users simultaneously, sharing voice conversations or even documents. All the members of the multiconference can participate in its revision, which is very useful for those companies that hold virtual meeting as it saves the company the travelling expenses.

3. ADVANTAGES AND DISADVANTAGES OF IP SERVICES

In this section, we will analyse separately both, the advantages and disadvantages in the use of IP services. We will also analyse the most relevant of these services:

3.1. Advantages

VoIP services offer a variety of advantages in every aspect. Its enumeration and explanation must be simple and clear to be able to make the future users aware of the advantages of its implantation in a non-distant future. It is crucial to avoid the contusion and premature rejection of something that emerges as a universal solution but that is not fully understood.

There are three major areas:

Business environment:

1. Big reductions in the cost of the telephonic bill. The cost of all calls will be the same as a local call, so the reduction in the cost of the traffic of voice will be very important from all points of view.

2. New possibilities of direct marketing and boosting of the customers' attention service. The philosophy of "Push 2 Talk" could be implanted. It consists of an icon on a web page through which the navigator could talk to qualified company staff while he continues navigating in the web.

3. Boosting of telework and the teleworkers. With only one connection, it will be possible to gain access to corporative applications, the vocal mail, receive calls or seek information on new projects.

• End users:

1. At present, the user that is using his domestic telephonic line to transmit data cannot receive communications at the same time, as the line is busy. The new VoIP services will not only allow to receive calls simultaneously but also to know who is calling to be able to admit or reject a call, or even divert it

• Service suppliers:

2. XoIP will be their new commercial argument. X implies the possibility of offering voice, data, fax or any service capable of being transmitted over an IP network. The clearest example is the new American aspect called Internet telephony Service Providers (ITSPs) that offers all kind of services over IP networks.

3.2. Disadvantages

If everything is so clear, if the technology exists, if the standards are already validated by imitational organisations (such as H.323 set by the ITU), if there is no legal objection and besides the international consultants introduce this solution as the actual business alternative in the year 2005, it is logical to think that the implementation of XoIP will be immediate. However, the real problem can be summarised in three letters: "QoS".

Quality of Service: It is not possible to guarantee quality of service over an IP network due to delays and bandwidth available. Once the voice has been digitalized and packetised, it is sent to the transmission channel and once there, there are no solutions that enable establishment of bandwidths order of packets or possible delays in its transmission. One possible solution could be to differentiate the packets of voice and the packets of data, giving priority to the transmission of the packets, of voice, and make sure that transmission of packets does not surpass in any case 150 mse (ITU recommendation) (Caputo, 1999)

Different organisations and manufacturers are starting to define solutions and standards but its applications or implementation is not considered to be possible in a minimum of 2 or 3 years.

The present work line and the solutions developed up to this day are based on:

- Bandwidth:

Table 1 shows the relation between the different algorithms of compression of voice and the bandwidth required.

- Delays:

Once the delays in he processing and transmit have been established the conversation is considered acceptable of under 150 ms

VoCodecs	Bandwidth
G.7ll PCM	64Kbps
G.726 ADPCM	16, 24, 32, 40 kbps
G.727 E-ADPCM	16, 24, 32, 40 kbps
G.729 CS-ACELP	8 kbps
G.728 LD-CELP	16 kbps
G.723.l CELP	6.3/5.3 kbps

Table 1: bandwidth required by present VoCodecs

- Echo:

The echo is due to a reflection, habitually due to a maladjustment if impedances.

- QOS:

The present work lines to obtain quality service IP transmission is based on:

a) Suppression of silence and VAD (voice activity detection): to establish the difference between voice and silence, and not to transmit packets of silence generating silence on the other end.

b) Compression of headers: assumption of the RTP/RTCP standards (VoIP'99). RTCP (Real-Time Control Protocol): it gives feedback on quality.

e) Bandwidth reserve: implantation of the RSVP standard of the IETF (Internet Engineering Task force) (Schmidt et al., 1998). RSVP incorporates bandwidth reserve and delay and it establishes a dynamic of access from one end to the other. Its main deficiencies are its defective growth (it's a valid solution in small networks) and its deficient authorisation and authentication. We must also bear in mind that the present infrastructures don't consider it.

d) Ranking: There are different tendencies such as:

1. CQ (Custom Queuing) (VoIP'99): assignment of a percentage of the bandwidth available.

2. PQ (Priority Queuing) (VoIP'99): it establishes priority in queues.

3. WFQ (Weight Fair Queuing) (VOIP'99): it assigns priority to traffic with the least weight.

4. Diffserv: defined in a rough copy by the IETF, it avoids jams in intermediate routers and establishes decisions of routers for every packet.

e) Congestion Control: it uses the RED (Random Early Discard) protocol (VoIP'99). This technique forces random discards.

f) Use of Ipv6: bigger addressing space and possibility of Ipv6 and tunnelling.

4. AN INDUSTRIAL APPLICATION

This section present an industrial application in which have been tested and demonstrated the possibilities of convergence (Valiño, 1999), always taking into account that the first step is the unification in one network of the voice and data transport. Obviously, the network is an IP network.

4.1. Aim

The aim of this project is to demonstrate the viability of a VoIP solution. Two technologically advanced entities: The PTG (Galician Industrial State) and the Cesga (Galician Supercomputing Centre) have participated in this project in collaboration with the University of Vigo. The infrastructure of both entities was suitable to develop the project in a frame complex in itself and different in each spot.

4.2.Initial stage

As it is shown in Figure 3, the PTG had a Siemens telephonic Central Hicom 300, equipped with 3 primary accesses, with capacity for 960 extensions and 200 operative extensions. Besides, there was a data network base on vertical fibre wiring, horizontal wiring on intertwined pair and an FDDI (Fibre Distributed Data Interface) along its urbanisation (www.ptg.es)

In the Cesga, the base was a voice network supported on a Lucent Definity telephonic Central equipped with one primary access and two basic accesses and a data network base on ATM Technology (www.cesga.es).

4.3. Requirements for convergence

To reach convergence, i.e. to integrate the voice and data network that were independent up to now, it was estimated that the necessary equipment was the following:

• Gateway to primary access in PTG: PC Pentium 300, 64 MB, and Windows NT.

• Gateway for secondary access in Cesga: PC Pentium 300, 64 MB, and Windows NT

• Voice encoder protocol: G720

• PTG switchboard: voice encoder card with a 75 ohms connector. (2 BNC)

• Cesga switchboard: voice encoder card with a 120 ohms connector. (1 RJ45)

• Switchboard compatibility study: definition of a common protocol.

• Connection protocol: EURO-ISDN

• Switchboard configuration mode: terminal mode.

• Gateway configuration mode: network mode.

• Dialling groups definition: incorporation of route server to identify users.

• Dialling mode: election of certain prefixes.

4.4. Final Stage:

In Figure 4, we can observe that the incorporation of the aforementioned equipment allowed reaching network convergence and to transmit voice and data, with the following conditions imposed by the entities that took part:

• The possibility to make telephone calls from Orense to La Coruña using the private network as a support.

• Possibility to make telephone call from La Coruña to Orense over the private network.

• Possibility to make free telephone calls from an extension in Cesga to an extension in PTG and vice versa over the private network.

• Possibility to make any kind of calls over Internet.

4.5 Next steps

Considering the success of the project, two new and immediate steps that supported the results were considered. The new actions considered of the incorporation of new agents to the project (new nodes) and the implantation of G 711 protocol, which allowed transmitting 64 kbps voice packets to obtain a better quality of transmission in the packets of voice.

5. CONCLUSIONS

At present, VoIP over private nets with a suitable design is a completely viable and operative solution. Companies are eager to apply it and its immediate incorporation on Intranets is feasible as it greatly improves the quality/price ratio. The present solution comprises design and a combination and optimisation of the various tools and resources available in each organisation.

Regarding the implementation of this solution on public nets such as Internet, the solution is viable, but the cost is very high in respect of loss of packets and intelligibility of conversations. This is the reason why the market is in expectation, and it is the user companies that will determine the rhythm of development and the implementation of solutions to guarantee QoS.

It is also remarkable that there are two relevant facts that for the time being establish the biggest obstacle or convergence: QoS applied to VoIP and the legal aspects to be taken into account.

Bibilography

Proceedings of the VoIP'98 Conference, (1998), International Institute Research, October, Madrid, Spain.

Canto, J., (1999), Byte Magazine, April, Madrid

Colchero, D. (1999), Computing Magazine, April Madrid

Proceedings of the VoIP'99 Conference, (1999), International Institute Research, October, Madrid, Spain.

Minoli, D., Minoli, E., (1998) Delivering Voice over IP Networks, John Wiley &Sons, March.

Held, G., (1998) Voice over Data Networks; Covering IP and Frame Relay, McGraw-Hill, June.

Goncalves, M., (1998) Voice Over IP Networks, McGraw-Hill, October

Caputo, R., (1999) Cisco Packetized Voice & Data Integration, McGraw-Hill, June.

ITU (1998), Table of Contents and Summary of Reconunendation H.323, ITU, Augnst.

Korpi N. (1998) Call Centers and CTI, Pocket Technology, June

Schmidt A., Minoli D.(1998) Multiprotocol Over ATM Building State of the Art ATM Intranets Utilizing RSVP, NHRP, LANE, Flow Switching, and WWW Technology, Prentice Hall, October.

Valiño M. (1998) VoIP: ¿La puerta hacia la convergencia?. Proceedings of the VoIP'98 Conference. International Institute of Research, October, Madrid, Spain.

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