The Effect of Voice Over Internet Protocol (VoIP) For Business Communication Services and Applications

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Abstract

VoIP as a technology has developed against a broader advancement in technology- namely convergence of voice, video and data into a single network. This is a development with huge potential in many ways- saving telecommunication costs, enhancing productivity and enabling mobility and flexibility for the people. VoIP can work across two networking situations. First, IP telephony can work over a local area network (LAN). This can be a good option for communications within the office. It can also extend to voice transmissions between people of the same organization working from different locales, over the company's network. VoIP can also be carried over a private or public wide area network (WAN). The private WAN is a more secure medium for communications than a public WAN. This VoIP is a technology system that transmits telephone calls over a data network, giving the miss to traditional telephone lines. The idea sounds simple, but this technology advancement has tremendous implications for business communications.

Keywords: *Asynchronous Transfer Mode (ATM), International Telecommunications Union (ITU), Quality Of Service (Qos), Resource Reservation Protocol (RSVP), Spectralink Voice Priority (SVP)*

1. INTRODUCTION

VoIP stands for 'V'oice 'o'ver 'I'nternet 'P'rotocol. As the term says VoIP tries to let go voice (mainly human) through IP packets and, in definitive through Internet. VoIP can use accelerating hardware to achieve this purpose and can also be used in a PC environment. VoIP has revolutionized the way we use telephony, with serious implications for gadget makers, service providers as well as the end users. This technology can considerably bring down the telephone/ long distance costs for individuals as well as businesses. VoIP is a set of technologies that enables voice to be sent over a packet network. While few corporations use VoIP today, its usage for messaging is expected to explode in the coming two years.Users can communicate using VoIP as easily as they do with today's PBXes and public phone network. By leveraging the existing data network, companies can save significant amounts of money by using VoIP for toll-bypass, which is particularly important for multinational corporations. VoIP will also speed the adoption of unified messaging by transmitting voice, fax and e-mail messages. VoIP is also known as IP telephony [1]-[16].

2. CONCEPT

The possibility of voice communications traveling over the Internet, rather than the PSTN, first became a reality in February 1995 when Vocaltec, Inc. introduced its Internet Phone software. Designed to run on a 486/33-MHz (or higher) personal computer (PC) equipped with a sound card, speakers, microphone, and modem (see Figure), the software compresses the voice signal and translates it into IP packets for transmission over the Internet. This PC-to-PC Internet telephony works, however, only if both parties are using Internet Phone software [17]-[28].

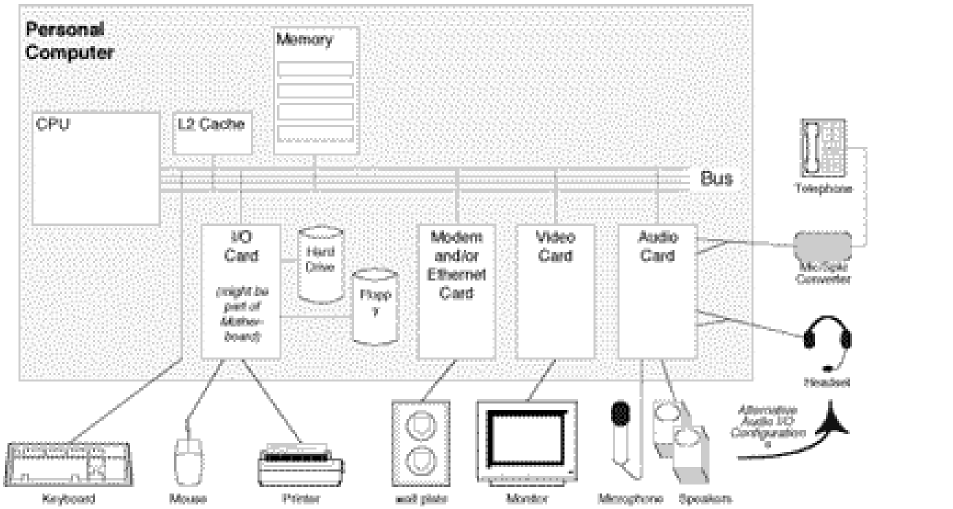

Figure 1. PC Configuration for VoIP

In the relatively short period of time since then, Internet telephony has advanced rapidly. Many software developers now offer PC telephony software but, more importantly, gateway servers are emerging to act as an interface between the Internet and the PSTN (see Figure). Equipped with voice-processing cards, these gateway servers enable users to communicate via standard telephones.

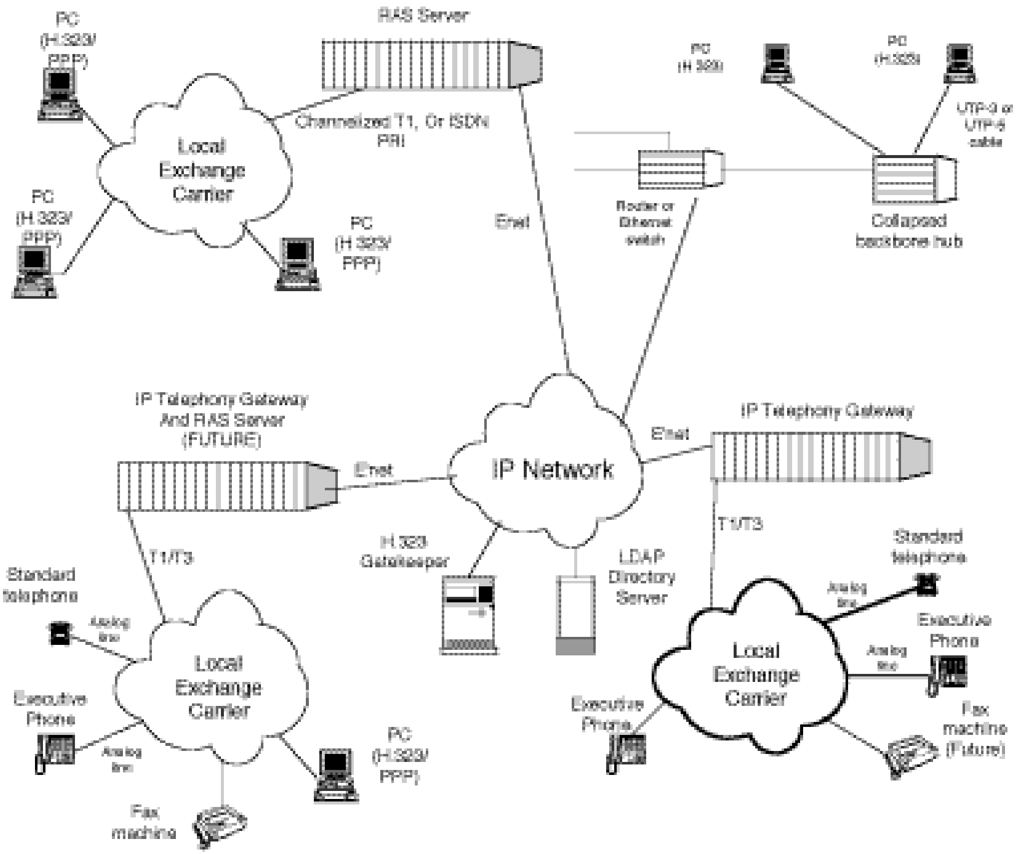

Figure 2. Topology of PC-to-Phone

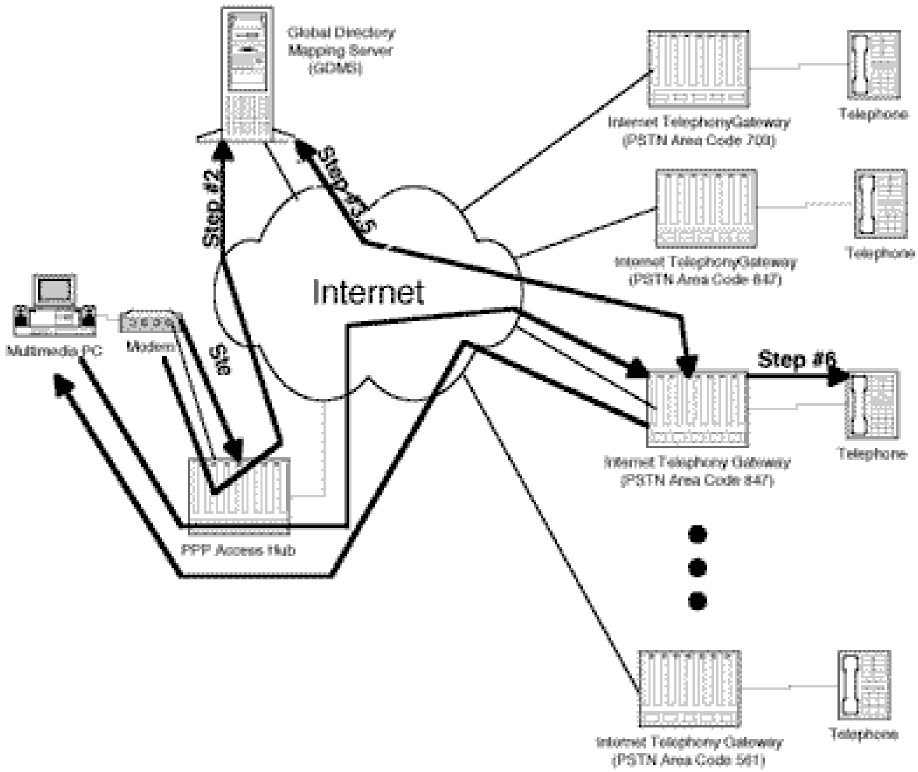

Figure 3. Sequence of VoIP Connection: PC-to-Phone

A call goes over the local PSTN network to the nearest gateway server, which digitizes the analog voice signal, compresses it into IP packets, and moves it onto the Internet for transport to a gateway at the receiving end (see Figure). With its support for computer-to-telephone calls, telephone-to-computer calls and telephone-to-telephone calls, Internet telephony represents a significant step toward the integration of voice and data networks [21]-[25] [27]-[30].

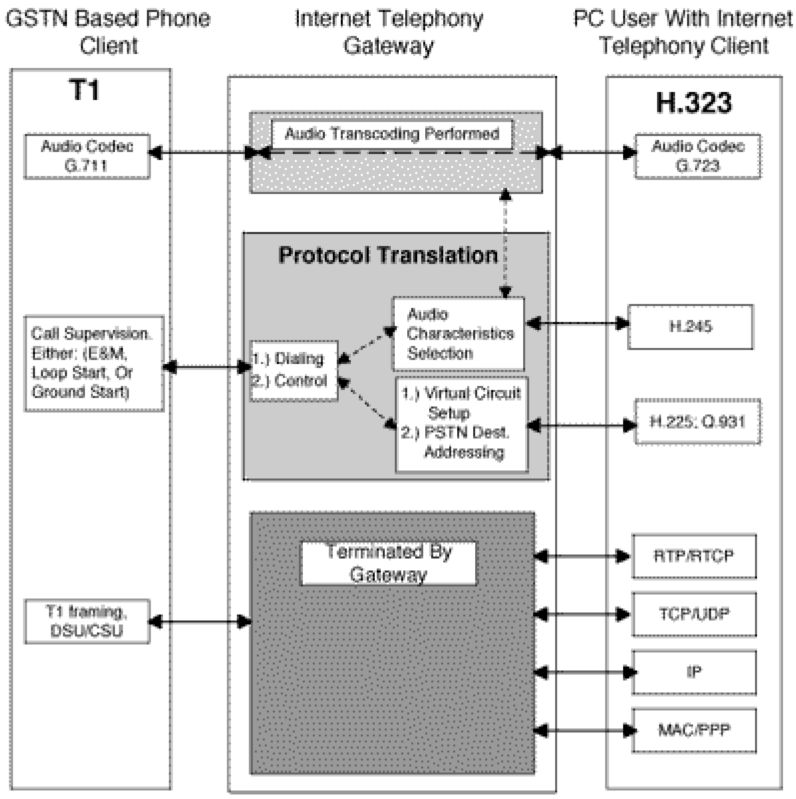

Figure 4. Sequence of VoIP Connection

Originally regarded as a novelty, Internet telephony is attracting more and more users because it offers tremendous cost savings relative to the PSTN. Users can bypass long-distance carriers and their per-minute usage rates and run their voice traffic over the Internet for a flat monthly Internet-access fee.

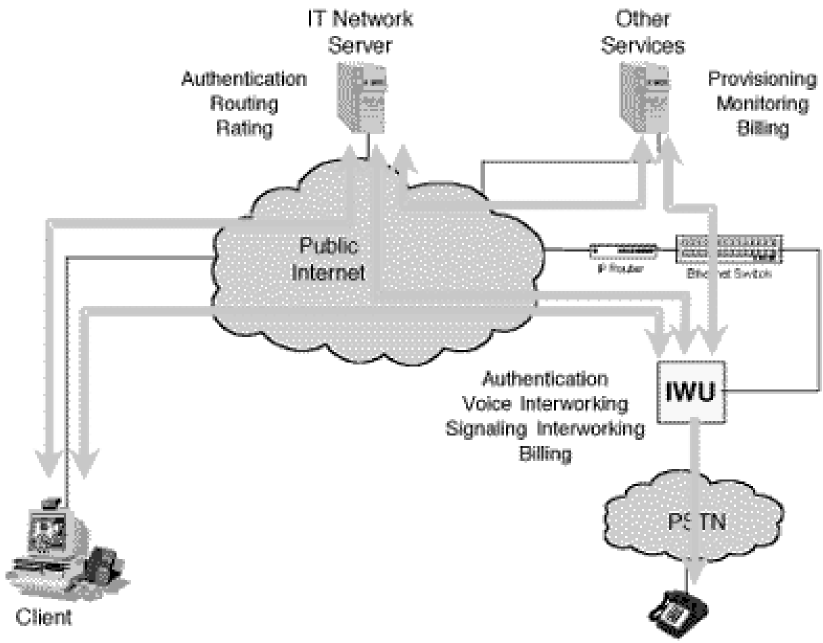

Figure 5. PC-to-Phone Connection

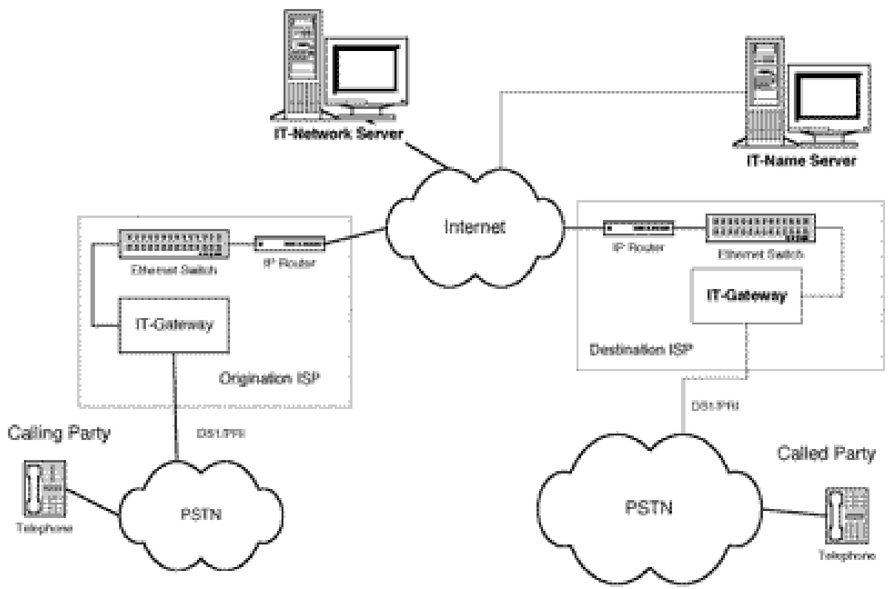

Figure 6. Phone-to-Phone Connection

3. STANDARDS

Over the next few years, the industry will address the bandwidth limitations by upgrading the Internet backbone to asynchronous transfer mode (ATM), the switching fabric designed to handle voice, data, and video traffic. Such network optimization will go a long way toward eliminating network congestion and the associated packet loss. The Internet industry also is tackling the problems of network reliability and sound quality on the Internet through the gradual adoption of standards. Standards-setting efforts are focusing on the three central elements of Internet telephony: the audio codec format; transport protocols; and directory services. In May 1996, the International Telecommunications Union (ITU) ratified the H.323 specification, which defines how voice, data, and video traffic will be transported over IP–based local area networks; it also incorporates the T.120 data-conferencing standard (see Figure). The recommendation is based on the real-time protocol/real-time control protocol (RTP/RTCP) for managing audio and video signals [17]-[30].

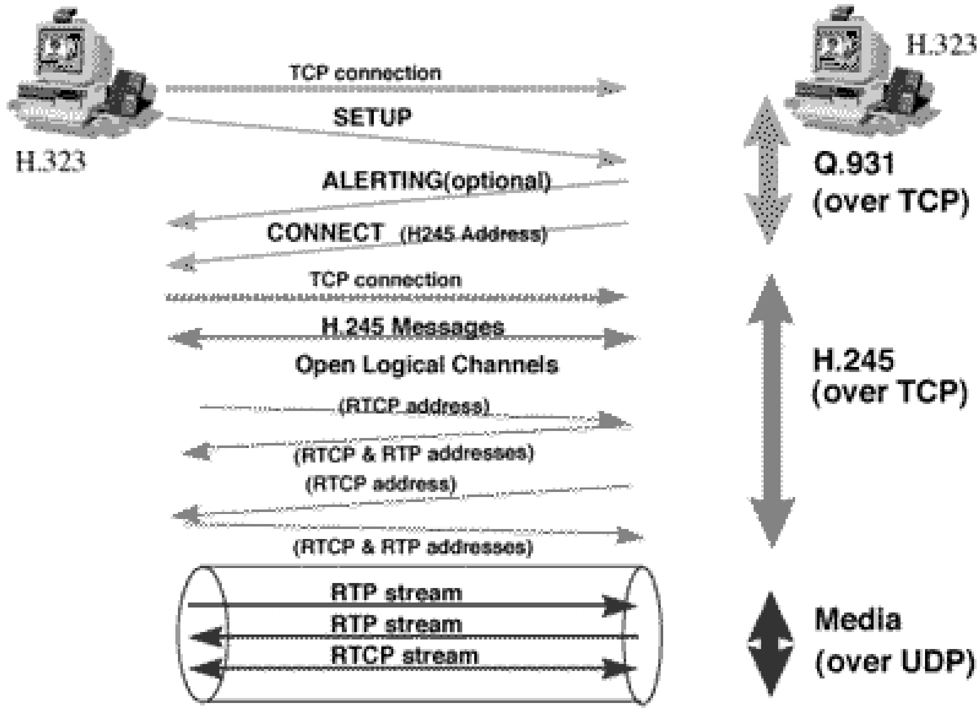

Figure 7. H.323 Call Sequence

As such, H.323 addresses the core Internet-telephony applications by defining how delay-sensitive traffic, (i.e., voice and video), gets priority transport to ensure real-time communications service over the Internet. (The H.324 specification defines the transport of voice, data, and video over regular telephony networks, while H.320 defines the protocols for transporting voice, data, and video over integrated services digital network (ISDN). H.323 is a set of recommendations, one of which is G.729 for audio codecs, which the ITU ratified in November 1995. Despite the ITU recommendation, however, the Voice over IP (VoIP) Forum in March 1997 voted to recommend the G.723.1 specification over the G.729 standard. The industry consortium, which is led by Intel and Microsoft, agreed to sacrifice some sound quality for the sake of greater bandwidth efficiency—G.723.1 requires 6.3 kbps, while G.729 requires 7.9 kbps. Adoption of the audio codec standard, while an important step, is expected to improve reliability and sound quality mostly for intranet traffic and point-to-point IP connections. To achieve PSTN–like quality, standards are required to guarantee Internet connections.

The transport protocol RTP, on which the H.323 recommendation is based, essentially is a new protocol layer for real-time applications; RTP–compliant equipment will include control mechanisms for synchronizing different traffic streams. However, RTP does not have any mechanisms for ensuring the on-time delivery of traffic signals or for recovering lost packets. RTP also does not address the so-called quality of service (QoS) issue related to guaranteed bandwidth availability for specific applications. Currently, there is a draft signaling-protocol standard aimed at strengthening the Internet's ability to handle real-time traffic reliably (i.e., to dedicate end-to-end transport paths for specific sessions much like the circuit-switched PSTN does). If adopted, the resource reservation protocol (RSVP), will be implemented in routers to establish and maintain requested transmission paths and quality-of-service levels. Finally, there is a need for industry standards in the area of Internet-telephony directory services. Directories are required to ensure interoperability between the Internet and the PSTN, and most current Internet-telephony applications involve proprietary implementations. However, the lightweight directory access protocol (LDAP v3.0) seems to be emerging as the basis for a new standard [3]-[12].

4. BENEFITS OF VOIP

VoIP offers all the features of traditional telephony such as call waiting, caller ID, unified messaging and directory services. As VoIP is a standard open protocol, businesses can easily integrate their own audio applications or other custom made applications with it. VoIP lends itself to easy expansion at little extra cost or requirement of elaborate hardware or software infrastructure.VoIP gives multiple options- it can be used in the form of a VoIP phone that is quite like the conventional desk phone. VoIP can also be used as ‘softphone’ loaded into desktops, laptops or PDAs. Users can use their VoIP phone number from any location. VoIP supports the convergence of data, video and voice services. Adopting VoIP will help companies benefit from a large range of multimedia applications being developed constantly [4]-[17].

5. WHY A BUSINESS WOULD CONSIDER VOIP

Let us first take a look at the traditional ways companies manage their voice communications. Large companies have PBX systems, which are private phone systems within the facility. In PBX, users share a certain number of telephone lines to make outgoing calls. Some companies have a variation of PBX called Centrex. Centrex or central office exchange service has all its PBX routing done at the local telephone company. Key Systems are another model of very small PBXs, ideal for small offices.

None these systems are of help in saving call costs. The costs can be enormous when it comes to long distance calls. And consider that long distance calls are becoming an integral part of business communications in the expanding global scenario.

Further, these conventional systems are difficult to change or expand as the company grows without going in for huge investments. They cannot support more advanced communication technologies. Moreover these are proprietary systems, so companies have only the limited facilities that their phone company provides.

These are precisely the drawbacks that VoIP can get rid of. It is both cost effective and easy to implement and upgrade.

6. VOIP OVER WIRELESS

1. Voice over Wireless IP
   Combining VoIP with 802.11 wireless LANs to create a wireless telephone system for offices is an emerging market segment. VoWIP enables businesses to leverage their wireless LANs to add voice communications, enabling companies to deploy and manage voice and data over a single wireless backbone. From a network perspective, VoWIP applications require some reservation of bandwidth to support the real-time nature of voice. Proprietary standards like Spectralink Voice Priority (SVP) are today's solution; however, the IEEE is developing the 802.11e standard for quality of service as a long-term solution.

2. The Voice over Wireless LANS
   When most people think of wireless LANs (WLANs), they generally only consider transferring data while using applications such as a Web browser, e-mail client, for file transfer, etc. It's possible, however, to use a WLAN as the transport system for carrying telephone traffic from mobile users as well. A significant benefit of mixing telephone traffic

with data on a WLAN is to provide mobility and make use of a common infrastructure. The support of a common system for both data and voice traffic is generally simpler and less expensive than two separate entities.

3. VoIP and Wireless Networking
   Voice over Internet Protocol, or commonly known as VOIP, is basically a type of software and hardware which allows people to use the Internet as the transmission medium for telephone calls by sending voice data in packets, or datagrams rather than by traditional circuit transmissions of the PSTN. It enables you to communicate with anyone who has a regular phone number. VOIP has emerged as a viable way to drastically cut costs of telephone calls, since telephone calls over the Internet do not incur a surcharge on what the user is paying for internet access. One may wonder why there exists a demand for wireless VOIP. Again, money is the simple answer here. Businesses take great interest in decreased communication costs while making use of a reliable and good quality service. Wireless VOIP can be classified as VoWiFi, or Voice over Wireless Fidelity [5] [9] [17] [19].

4. The pros and cons VoIP Over Wireless
   Voice over IP (VoIP) technology allows companies to cut the telco cord and make phone calls over the internet. Now, with VoIP over wireless - also called VoW, VoWiFi, wVoIP, and a number of other acronyms - organisations can cut all the cords. Numerous vendors are offering Wi-Fi IP phones that operate on the same 802.11 technologies used for wireless networks. According to a study from Infonetics Research, sales of Wi-Fi IP phones will likely reach \$3.7bn (£2bn) by 2009. The market already totalled more than \$125m in 2005, and it's growing steadily, particularly in business fields that have many mobile workers. That includes people who work in hospitals and factories, on sales floors, etc.

7. FUTURE OF VOICE-OVER-INTERNET PROTOCOL

Several factors will influence future developments in VoIP products and services. Currently, the most promising areas for VoIP are corporate intranets and commercial extranets. Their IP–based infrastructures enable operators to control who can—and cannot—use the network. Another influential element in the ongoing Internet-telephony evolution is the VoIP gateway. As these gateways evolve from PC–based platforms to robust embedded systems, each will be able to handle hundreds of simultaneous calls. Consequently, corporations will deploy large numbers of them in an effort to reduce the expenses associated with high-volume voice, fax, and videoconferencing traffic. The economics of placing all traffic— data, voice, and video—over an IP–based network will pull companies in this direction, simply because IP will act as a unifying agent, regardless of the underlying architecture (i.e., leased lines, frame relay, or ATM) of an organization's network. Commercial extranets, based on conservatively engineered IP networks, will deliver VoIP and facsimile over Internet protocol (FAXoIP) services to the general public. By guaranteeing specific parameters, such as packet delay, packet jitter, and service interoperability, these extranets will ensure reliable network support for such applications [1]-[15] [19]-[27].

8. CONCLUSION

VoIP products and services transported via the public Internet will be niche markets that can tolerate the varying performance levels of that transport medium. Telecommunications carriers most likely will rely on the public Internet to provide telephone service between/among geographic locations that today are high-tariff areas. It is unlikely that the public Internet's performance characteristics will improve sufficiently within the next two years to stimulate significant growth in VoIP for that medium. Over the next several years, companies will deploy VoIP in conjunction with 802.11 wireless LANs, enabling workers to have WLAN-based mobile phones when in the office.

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