Edwin van Tricht and Cor Quist

Voice-Over-IP for Corporate Users

A solution in search of a problem?

What is the real value of voice over IP (VoIP) for corporate users? This is the key question of this paper. Five application areas of VoIP for companies are discussed: VoIP over the local area network (LAN), VoIP over the wide area network (WAN), VoIP for customer contact, VoIP for telecommuters and multimedia collaboration over IP. Competing technologies exist for all five areas. There is, however, a strong tendency towards IP-based solutions and VoIP does offer some unique advantages. The sometimes-poor voice quality of VoIP solutions is a serious barrier for its success. It is also important to note that current VoIP solutions do not yet offer all the well-known features of the classic PBX. It is expected that solutions will become available to solve these limitations.

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Introduction

According to some vendors, every company will have implemented voice-over-Internet-protocol (VoIP) solutions in a few years' time, regardless of the application or field of operation.

The following five application areas of VoIP for corporate users are distinguished: VoIP over the local area network (LAN), VoIP over the wide area network (WAN), VoIP for customer contact, VoIP for telecommuters and multimedia collaboration over IP.

For each application, the pros and cons of VoIP are discussed and compared to competing technologies. Based on this comparison, conclusions are drawn regarding the actual value of VoIP for the different application areas. The paper starts with a brief explanation of VoIP and of the quality it can offer.

What is VoIP?

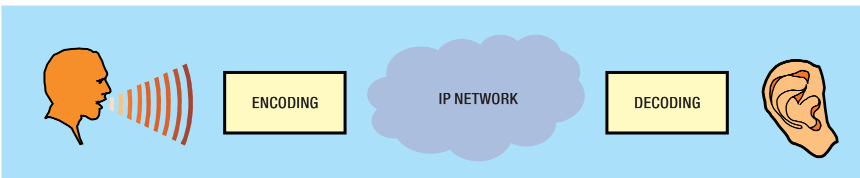
VoIP is a technology that transports voice over an IP network. With the help of special coding methods, voice is converted into IP packages, which are then transported over an IP network. The encoding software and hardware are known as *codecs*. The receiver decodes the packages and converts them back into voice. Figure 1 illustrates this process.

The great advantage of using IP for data or voice transmission is the fact that it can be transported over almost any other transmission protocol. For example, a user who is linked to an Ethernet LAN and someone working at home with a public switched telephone network (PSTN) line can communicate using the same IP application.

Another advantage is the compression that can be applied during the encoding. At the moment, the most popular codecs can compress a 64 kbit/s voice connection by a factor of four, making it a 16 kbit/s connection. Some of the more advanced codecs can even compress to a bit rate of about 5 kbit/s. By way of comparison, global system for mobile communications (GSM) encoding compresses a 64 kbit/s voice signal to 9·6 kbit/s. One should, however, bear in mind that compressed audio data need to be encapsulated in IP packets. This introduces a significant amount of overhead. We found that—depending on the parameters sample frequency, silence suppression and header compression—the overhead can easily result in the required bandwidth being doubled.

There are also disadvantages in using IP networks for real-time applications such as voice or video. IP was originally not developed for real-time applications, which means that the voice quality is lower than the

Figure 1—With the help of codecs, voice is converted into IP packages and transported over an IP network and subsequently decoded



PSTN quality. There is a risk of delays, echo and the loss of various parts of the conversation—if you are having a conversation over the Internet at a bad time, you could end up in a kind of walkie-talkie exchange. On the other hand, laboratory demonstrations have shown that voice quality is satisfactory when using VoIP on a well-dimensioned and well-managed IP network.

KPN Research performs numerous tests on the VoIP quality. Voice quality is not easy to measure. The relevant parameter is the perceived quality by the end-user, expressed as the mean opinion score (MOS). A live conversation has an MOS of 5, PSTN corresponds to MOS 4 and mobile telephony using GSM corresponds to MOS 3. KPN Research has developed a method to predict the MOS factor of voice in an IP network based on the technical parameters of the IP network. During a test carried out in 1998, the highest MOS score obtained was 3·5. So even under perfect conditions, the speech quality is still below PSTN quality. Five parameters determine the perceived audio quality: bandwidth, delay, jitter, echo and packet loss. We found that as the delay increases, the echo problem also increases. The delay of VoIP services will always be higher compared with the delay of the public switched circuits.

VoIP Applications within a Company

This paper discusses five applications of VoIP (see Figure 2):

- VoIP over the LAN,
- VoIP over the WAN,
- VoIP for contacts with customers,
- VoIP for telecommuters, and
- multimedia collaboration

These five application areas are discussed below. First, there is a short description of each application's architecture, then the existing products are described and finally, the competing technologies are highlighted.

VoIP over the LAN

Architecture

The architecture for VoIP over the LAN is illustrated in Figure 3. A growing number of terminals is available for the use of VoIP over the LAN. The terminals can be seen from left to right in Figure 3. On the left is a classic telephone connected directly to a router which has an integrated VoIP-gateway. Secondly, multimedia PCs can be used as telephones. These can include a regular desktop PC or a laptop communicating over a wireless LAN. In the middle you can see a special IP telephone connected directly to the LAN (in combination with an IP PBX). An IP PBX is a telephony server that provides already existing telephony features, such as completion of calls to busy subscriber (CCBS) and call diversion, as well as new computer telephony integration (CTI) features based on the integration between the telephony and the PC/data environment. For example, the number and name of an incoming call can be displayed on the PC screen. At the right side of Figure 3 you can find a solution for mobile telephony: wireless IP telephones that use wave LAN technology to communicate with other terminals.

This IP architecture consists of one single network for both data and voice, which means that only one infrastructure has to be managed. It is also possible to integrate the configuration management. When adding a new user, the new telephone number is created in much the same way as a new e-mail address is created. In this way, user data can be created and modified more efficiently, thereby reducing management costs.

Existing products

Every PBX vendor and IT vendor is currently active in this field. The products they sell can be divided into two categories: IP telephones and (small) gateways for normal telephones. IP telephones can be plugged directly into any Ethernet interface on the LAN. Selsius, meanwhile taken over by Cisco, was the first vendor of IP telephones and IP PBXs. (Small) IP gateways are integrated within the router and standard analogue telephones can be plugged into the router. At the moment, only a small number of services is supported. Frequently-used features such as call completion on busy subscriber are not supported at the moment.

Figure 2—Five applications areas of VoIP within a company

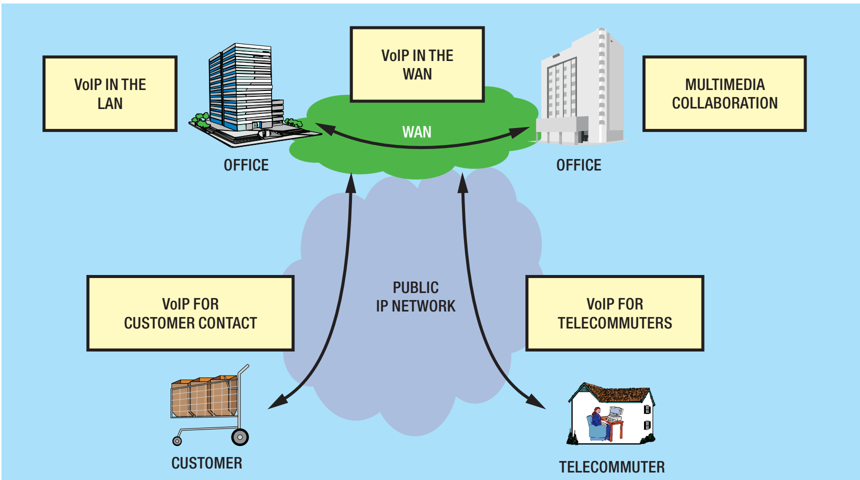
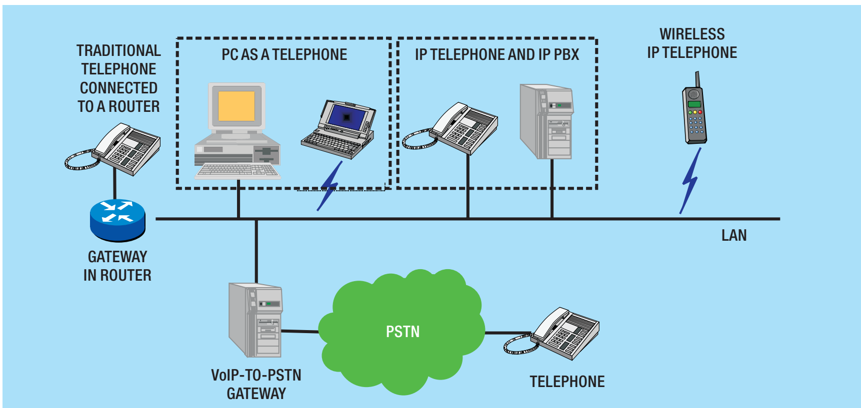


Figure 3—Architecture of a company network using VoIP over the LAN



Voice quality

The present IP installed base within companies is usually not sufficient to guarantee the quality of a telephone call. This means that a conversation might be temporarily interrupted if your neighbour starts up his word processor from the network. Quality guarantees for company networks can only be given if all equipment (for example, routers) supports the same protocol. Preparing your network for high-quality voice therefore often requires upgrading it entirely.

Competing technologies

IP telephony over the LAN will have to compete with existing solutions for business telephony. The PBX is still the preferred solution. The PC-based PBX is a fairly recent development. The various solutions are discussed and compared below.

The PBX is currently the most popular solution for business telephony. In a PBX, the switching matrix for setting up calls between the various extensions and the management software are completely integrated. The system is completely closed—it is not possible, for example, to develop a customer-specific application for a PBX. CTI has been developed to overcome this problem. CTI solutions link a server to a PBX via the hardware.

With a PC-based PBX (PCBX), the link between the telephony environment and the IT environment is not created using the hardware, but using the software. This PCBX is equipped with hardware cards that contain PBX functionality and can be controlled by the software in the PC. The telephone at the desktop is directly connected with the PCBX, while the PC is linked to the PCBX over the LAN. The software connection allows virtually every functionality desired by the user to be built. All of the existing PBX features can be realised, and new services such as video-conferencing and application sharing can be used. The drawback of a PCBX is its limited reliability, which will make it difficult to equal the 99.999% reliability of the traditional PBX.

Use of VoIP over the LAN

The best solution for a company depends greatly on its individual needs. If it wants to make calls using additional services such as voice mail, then a PBX is still the most suitable solution. This will be the case for the vast majority of businesses. Companies that place high demands on functionality will prefer a PCBX solution. By developing a customer-specific PCBX application every desired functionality could be achieved. For the next two or three years, VoIP over the LAN will only be used in niche markets, for example at locations that have a data network but no telephony infrastructure in place.

VoIP over the WAN

Architecture

VoIP also offers new ways of connecting PBXs over an IP network. This option uses VoIP-to-PSTN gateways. The gateway digitises and compresses the voice and splits the audio-bitstream into IP packets. It then sends the data over an IP network to another gateway, where the audio is decoded again. Figure 4 illustrates the architecture.

Integrating voice and data across the same network can offer cost savings—only one network has to be managed by the company, and only one service has to be supplied by the network provider.

Existing products

Current gateways were not specifically developed to link PBXs but to be used in the public network. Transmission of special PBX signals between two gateways is not supported by any supplier at present.

The various suppliers of VoIP-to-PSTN gateways can be divided into three groups: individual gateway suppliers, router suppliers who want to integrate the gateway into the router, and PBX suppliers who try to integrate the gateway into the PBX. The first group is by far the largest, yet this solution is less interesting for businesses, because a stand-alone system cannot be integrated in nor managed with the telephony environment or the data environment. Cisco is a clear example of a company that integrates the gateway into the router, whereas Nortel has chosen to integrate the gateway into the PBX. It is difficult to predict which of the two solutions will prevail in the long term.

Voice quality

Relatively speaking, it is easier to give quality guarantees for VoIP over the WAN than for VoIP over the LAN because less equipment has to be upgraded or replaced. Only the actual connections with the WAN have to be modified. Depending on the type of network and the available bandwidth, the quality of the voice connection is roughly the same as GSM quality. This quality will increase in the coming years through the use of new protocols, which will make it possible to give quality guarantees.

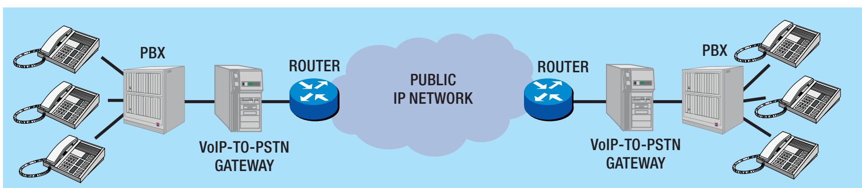
Competing technologies

The idea behind VoIP over the WAN is that voice and data are integrated over one network. Older technologies that do the same thing include voice over frame relay and voice over ATM, both of which are further developed. For example, the transport of special PBX signalling between sites is already supported for ATM and frame relay, but not for IP.

Neither the IP protocol nor frame relay were originally developed to transport high-quality voice. ATM was. This means that it is easier to obtain a high-voice quality compared to frame relay or IP. Besides, IP always requires a transmission mechanism such as ATM or frame relay. This begs the question why a company would send its voice traffic over IP when it could just as well do so over the underlying transport layers like frame relay or ATM. IP does offer four important advantages:

- IP can be used as an overlay protocol in companies that use different transmission protocols.
- IP is carried to the desktop of the end user. This enables functional integration between the PC/data-environment and telephony applications (CTI).
- The data industry is currently focussed on IP, allocating large research and development budgets to IP-related developments. Furthermore, IP products are improving very rapidly.

Figure 4—Configuration for connecting PBXs over an IP network



- The advent of *all-IP operators* can result in cheaper IP transmission.

Use of VoIP over the WAN
VoIP over the WAN will be used by businesses that simultaneously use several transmission technologies in their network. The main offices can be linked to an ATM network, and the branch offices to a frame relay network. The IP protocol serves as a bridge between the various protocols. Businesses that implement it in the years ahead will have to accept that voice quality will be somewhat poor and that the various PBX features will not be transparent throughout the network. Businesses that use only one transmission technology in the network will probably not use VoIP but direct voice over frame relay or voice over ATM instead.

There will be a steady increase in the number of providers offering IP networks to link sites. The company will be supplied with an IP plug at each site. In this situation, companies may well choose a VoIP solution if they want voice-data integration.

VoIP for customer contact

Architecture
VoIP can be used as an extra way for businesses to come into contact with their customers, for example when a customer is surfing the company's web page and wants to ask a question about a certain product. Using VoIP, the customer can call the company without first having to break the connection. This method uses a single VoIP-to-PSTN gateway that converts the voice into IP packages and vice versa. The customer will need a multimedia PC with a sound card, a microphone and a loudspeaker. The VoIP-to-PSTN gateway can be managed either by the company itself or by a service provider. In the latter case, the gateway may be used by several businesses at the same time. This architecture is illustrated in Figure 5.

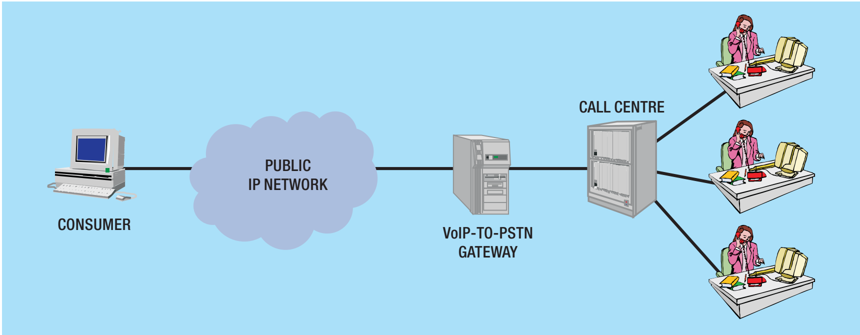


Figure 5—VoIP architecture for customer contact

Existing products
VoIP-to-PSTN gateways are available as individual gateways. Call centre suppliers are also busy integrating these gateways into their equipment.

Voice quality
Before a company can offer this kind of service to its customers, it must first examine the quality of the voice connection very carefully. If communication with the customer is rendered impossible by a poor connection, this will certainly not be good for the relationship with the customer. The customer will probably blame the company rather than the IP network.

Competing technologies
There are two alternatives for this application:

- call the call centre by a normal telephone once the web surfing session is finished, or
- the use of the so-called *click-to-dial* button whereby a visitor of a web site clicks this button and the call centre call him/her back over the regular PSTN at a specified time.

The advantage of these alternatives is that the telephone call is a regular PSTN telephone call and the quality will be high. The advantages of the VoIP-application are that

- the customer can speak to the call centre agent directly;
- it offers the team-browse feature—the customer and the call centre agent can discuss a specific web page; and
- the customer is still on-line.

Use of VoIP for contact with customers
VoIP for contact with customers is an addition to the existing communication possibilities using web pages. Companies can experiment with this kind of application without big investments if they start with one central gateway managed by a service provider. If the response is satisfactory, the call centre at the customer location can be extended with a gateway.

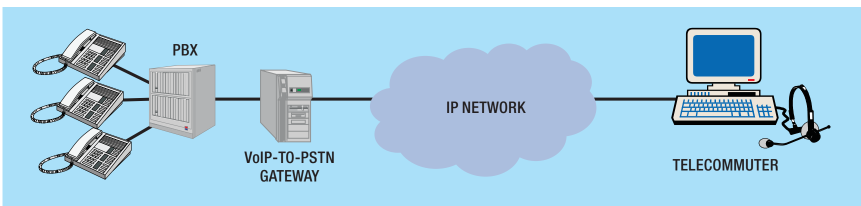
The potential problem of low voice quality might be avoided by giving the customer a choice: using VoIP which might result in a lower quality, or dialling back using the normal high-quality telephone. The latter option has the disadvantage that customers with only one PSTN telephone line have to disconnect from the Internet first.

VoIP for telecommuters

Architecture
Telecommuters often use a dial-in connection to connect to the company intranet. This same IP connection can be used for voice. When the telecommuter logs in to his/her intranet, a special software program on his/her PC logs on the VoIP-to-PSTN gateway. An incoming call, addressed to this telecommuter, is forwarded by the PBX to the VoIP-to-PSTN gateway. The gateway translates the voice into IP packets and sends them to the telecommuter's PC. The telecommuter can also use his/her PC at home to originate telephone calls. Even some PBX features can be supported. This architecture is illustrated in Figure 6.

Existing products
Virtually all gateway suppliers support telecommuter functionality.

Figure 6—VoIP architecture for telecommuters



The architecture is very similar to the VoIP architecture used for customer contact. The only difference is that telecommuters have to log on to the gateway before they can use their PCs as telephones.

Voice quality

When telecommuters dial directly into the company's intranet, only privately owned network capacity is used and the voice quality approaches that of GSM telephones. However, sometimes public IP networks are used as dial-in infrastructure. In that case, the company does not have to invest in its own modem pool, but uses the modem pool of an Internet service provider. When using the public infrastructure of an Internet service provider, part of the network can no longer be controlled by the company. When using the Internet, voice quality becomes unpredictable.

Competing technologies

Traditional PBX vendors use proprietary telephones to support telecommuter functionality. These telephones are relatively expensive and require a second telephone line. The VoIP solution assumes that the telecommuter uses a multimedia PC which is permanently logged on to the network.

Another competing technology is GSM. Given the fast-growing penetration of mobile telephones, the need for fixed telephones is decreasing. The telecommuter's fixed telephone at the office is forwarded to his/her mobile telephone, without the need for extra gateway investments. When using extra fixed-mobile integration services, it is even possible to dial short numbers on the mobile telephone.

Use of VoIP for telecommuters

VoIP technology is available to automatically transfer calls made to fixed telephones at the office to the PC at home. A voice quality similar to GSM is certainly possible. This can be an interesting solution for telecommuters that have a multimedia PC at home, yet it might come too late, considering the rapid penetration of GSM.

VoIP and multimedia collaboration over IP

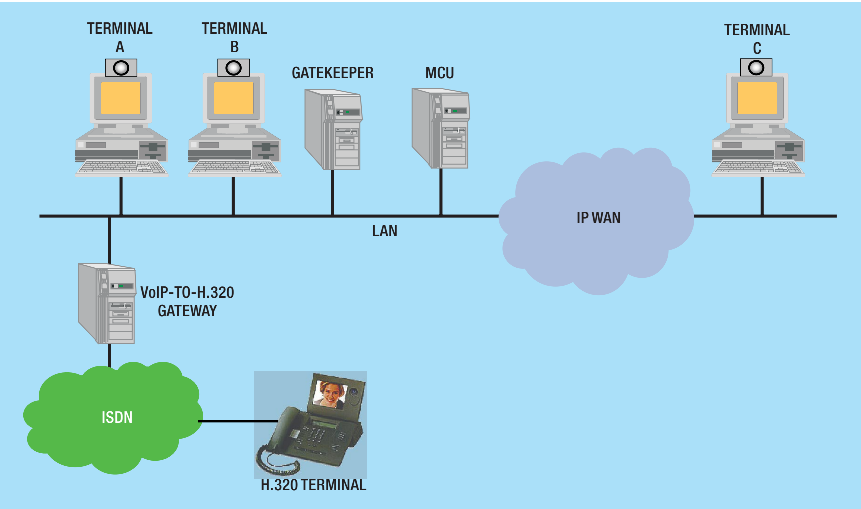
Architecture

It's a done deal: IP is the standard protocol for multimedia communication. Multimedia communication at the desktop will become the standard in the corporate environment. The three traditional key barriers are disappearing: bandwidth becomes available on the LAN and the WAN, desktop PCs are becoming increasingly powerful and the prices of equipment are dropping.

The ITU standard for VoIP, H.323, also supports application sharing and video communication.

A possible architecture is illustrated in Figure 7. We foresee that the desktop multimedia PC will become the de facto multimedia terminal. The gatekeeper is a necessary management tool to control the bandwidth usage and to translate the name of end users into IP addresses on the network. Bandwidth management is a crucial function as real-time traffic (voice, video) is very demanding. The MCU is a multipoint conferencing unit. This server enables multimedia conferences between three or more terminals. And finally, there is the H.323/H.320 gateway, which translates the information of a H.323 terminal to a H.320 (ISDN videoconferencing) terminal and vice versa. This gateway is only useful if you want to communicate with H.320 terminals. It can also be used to make a voice call from any H.323 client to a regular telephone on the PSTN.

Figure 7—Architecture for multimedia collaboration over IP



Existing products

Multimedia clients come in several forms. An increasing number of H.323 multimedia terminals is available. If you have a multimedia PC, you can use Microsoft Netmeeting, a public domain H.323 client that supports multimedia communication. However, if you prefer to have a better video quality you need to buy special H.323 clients; for example, PictureTel's client. This client uses about 700 kbit/s for the video images. Multipoint conferencing units (for example, PictureTel, VideoServer) and H.323/H.320 gateways have been available since 1998.

Voice quality

The voice quality strongly depends on the characteristics of the network and on the multimedia client. The voice quality can be greatly improved by using a headset instead of a microphone and speakers. Multimedia communication, including video communication, is possible over a well-designed and well-managed LAN/WAN. Special attention needs to be paid to the audio and video synchronisation if video is used. Each client uses a different approach to synchronise the media streams.

Competing technologies

Multimedia collaboration can also be used over the integrated services digital network (ISDN) or asynchronous transfer mode (ATM). Videoconferencing systems for ISDN (H.320 terminals) and ATM are available. Multimedia over IP has one great advantage: IP can be used from the desktop, whereas ISDN and ATM are mainly used to connect videoconferencing studios. An alternative for multimedia collaboration is to use standard telephones for audio conferencing while using the desktop PC for a synchronised video communication and data-/application-sharing session. Although this is feasible from a technical point of view, it is not very popular in the marketplace.

Use of Multimedia collaboration over IP

Multimedia collaboration is particularly useful for multi-site companies with an intranet. It allows employees to participate in multimedia meetings from their desktop PC. Data/application sharing can enrich audio conferences. Video is optional and can be useful for certain types of meetings. Managed bandwidth in the LAN and WAN is required, especially when video is added to the conference. Employees need to have access to multimedia PCs.

Conclusion

This paper argues that using VoIP on the LAN will be a niche market for the forthcoming years. If companies adopt VoIP on the LAN, a gradual migration from their traditional PBX to VoIP on the LAN is expected.

Voice-data integration in the WAN will allow cost savings. This can be achieved using VoIP but also through voice over frame relay and voice over ATM technologies, which are slightly older and have made greater progress than VoIP. VoIP can, however, be a good solution if different tranportation technologies are used within the company network.

VoIP is the only solution for companies dealing with on-line customers who must be able to *push-to-talk* and have access to the Internet simultaneously. Companies should seriously consider the speech quality they can provide to their customers using this technology. Low quality will result in a bad image. To prevent this, companies can give customers two options: dial back using normal telephone lines or VoIP.

VoIP for telecommuters can be a cheap solution compared to proprietary PBX solutions. It enables telecommuters to automatically transfer calls made to fixed company telephones to their PCs at home. The PC can be used as a telephone that even supports some PBX features. However, the need for this solution is decreasing with the fast-growing penetration of mobile telephones.

IP is expected to become the platform for multimedia collaboration applications. Voice, video and data will become completely integrated. The available bandwidth on the public Internet will not be sufficient for these applications to grow substantially. Managed LANs and WANs certainly provide a solution here.

Biographies

Edwin van Tricht
Synergetics IT Consultancy

Edwin van Tricht has worked as a consultant within KPN Research for three years in the field of business communications. Within KPN Research, new upcoming technologies are evaluated, which enables KPN Telecom to offer new services. Edwin worked on several projects to evaluate new technologies, such as VoIP, PC-based switching and multimedia call centres. In the summer of 1999, he left KPN Research and is now working for Synergetics IT Consultancy.

Cor Quist
KPN Research

Cor Quist is a consultant within KPN Research in the field of multimedia and systems integration. He has been working on multimedia communication services, interactive multimedia applications, tele-education and voice-over-IP services. He has been following the developments in voice over IP ever since this evolving technology first presented itself. Over the past few years, he has been closely involved in the strategic discussion on VoIP within KPN Telecom and has contributed to several VoIP pilots and services that KPN Telecom has recently introduced.