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IP Telephony – Today/Tomorrow/Ever?

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SUMMARY

Voice communication over the Internet debuted in mid-1994 with a shareware program, ‘Internet Voice Chat’ for simple PC-to-PC connections. Subsequently, the market has evolved into an Internet protocol (IP) telephony market with a number of applications such as voice telephony, voicemail, fax, desktop videoconferencing, application sharing and document sharing.

Telephony carried over IP networks has generated much attention from end-users, ISPs, data vendors, telephony vendors and existing telecom operators. This attention is centered around the promises of very inexpensive telephony services. By routing voice traffic over IP telephony users can bypass long-distance charges completely, paying only for a local connection.

While the IP telephony technology is still in its infancy, it is expected to mature within the coming years leading to phone-to-phone, PC-to-PC, PC-to-phone, phone-to-PC and fax-to-fax services over IP networks. Ultimately, this is expected to be Internet-based multimedia communication and collaborative tools integrated with the enterprise PABX and call center applications.

Today the IP Telephony market is quite small, but International Data Corporation (IDC) has predicted that the Internet telephony market will grow to $560 million by the end of 1999. According to Frost & Sullivan, software and hardware manufacturers are expected to realize $1.89 billion in global IP telephony revenues by the end of 2001.

These are only predictions, and should of course be treated as such, but they do indicate the potential for IP telephony.

What is IP Telephony?

IP telephony uses the Internet protocol to transmit voice as packets over an IP network. This means that IP telephony can be achieved, at least in principle, on any data network that uses IP, such as the Internet or any of a growing number of Intranets and local area networks (LANs). This is in contrast to traditional circuit switched telephony, where an end-to-end circuit is set up between two telephones. A circuit switched connection is established for the duration of every telephone call, with a fixed bandwidth (64 kbit/s) reserved even during silent periods.

In an IP telephony connection, the voice signal is digitized, compressed and converted into IP packets, which are transmitted over the IP network and shared with other IP traffic.

Figure 1.

A normal telephone call is connected through an end-to-end circuit with a fixed bandwidth. With IP telephony, a packet-based network is used where a number of calls share the same network link.

The data networking technology moves information at a much lower cost by making better use of the network capacity. Not only is a packet-based shared network more efficient than a fixed 64 kbit/s circuit switched connection, but it also compresses the voice signal.

To accomplish this, a gateway provides the connection between the telephone network and the IP network.

Figure 2.

Typically IP telephony is handled by a voice gateway that is placed between the PSTN and the IP network.

The Voice Gateway provides the physical interface between the telephone network and the IP network. The Gateway handles the signaling to and from the telephone network, reception of telephone numbers, conversion between telephone numbers and IP addressing in the IP network, as well as voice processing.

The voice processing implies reception of the voice signal, compression and packetization, echo cancellation, silence suppression, etc. The gateway compresses the voice signal for two reasons: to reduce the amount of bandwidth required in order to reduce cost and to reduce the delay impact from the network.

Generally, users dial a telephone number of the gateway, the gateway will respond with

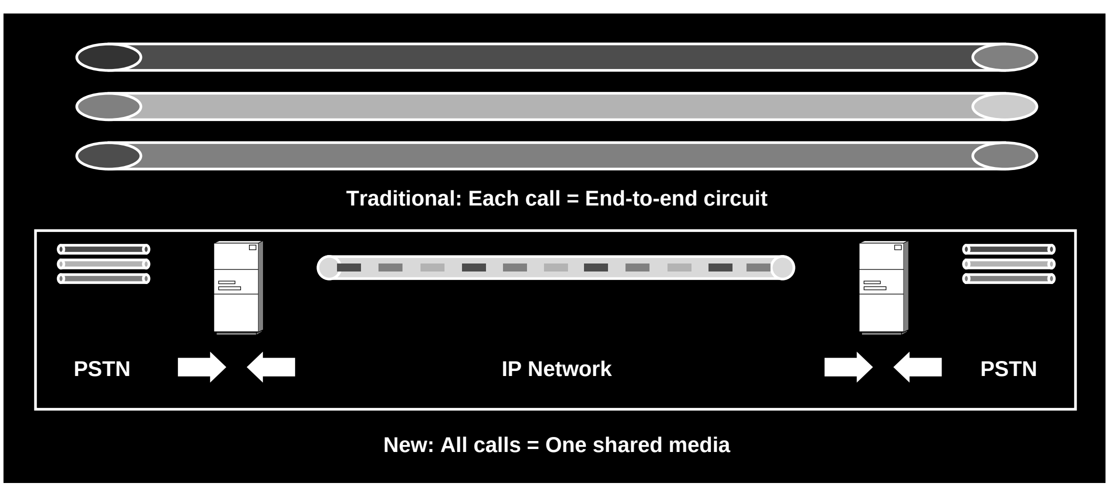

Figure 1

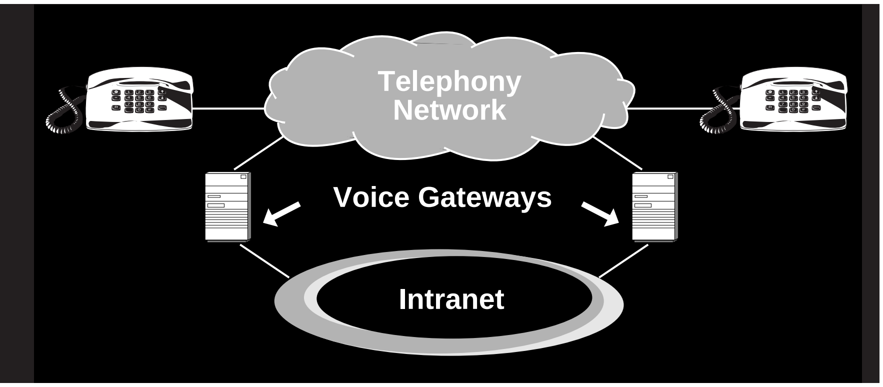

Figure 2

an audio request for the user's destination telephone number, and a routing table will identify which gateway is located closest to the destination telephone network. The IP address of that gateway is then used to route the telephone call as datagrams through the IP network.

The gateway also reverses the operation for packets coming in from the network and going out of the phone. Both operations (coming from and going to the phone network) can take place at the same time, allowing a full-duplex conversation.

IP telephony encompasses a number of services including phone-to-phone, PC-to-phone, phone-to-PC, PC-to-PC and fax-to-fax, as well as video conferencing and collaboration from the desktop.

Figure 3.
In an IP telephony solution, it is possible to have a combination of PC-based telephony applications and PSTN connected telephones.

In a phone-to-phone scenario, the gateway will, in real time, perform the functionality needed in order to send and receive voice over the IP network.

In a PC scenario an IP telephony client is needed. The client digitizes, compresses and packetizes the voice signal and transmits it over the IP network. Standard telephone calls are connected to a voice gateway and IP telephony calls connect to a telephone or a PC.

IP telephony client software may allow users with multimedia PCs to video and audio conference, share documents and use a white board, enabling a more efficient work environment.

Fax-to-fax IP telephony includes both real-time fax and store and forward fax. Real-time fax is characterized by:

- The sending and receiving fax machines directly perform the required "handshaking"
- The fax is sent directly from the sending machine to the receiving machine
- The sender receives a confirmation as soon as the transmission is completed

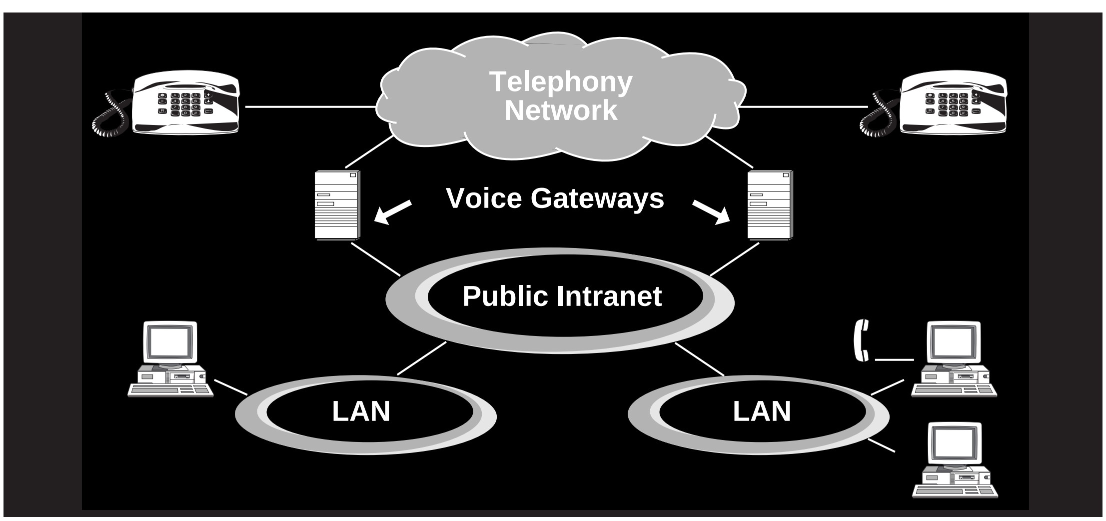

Figure 3

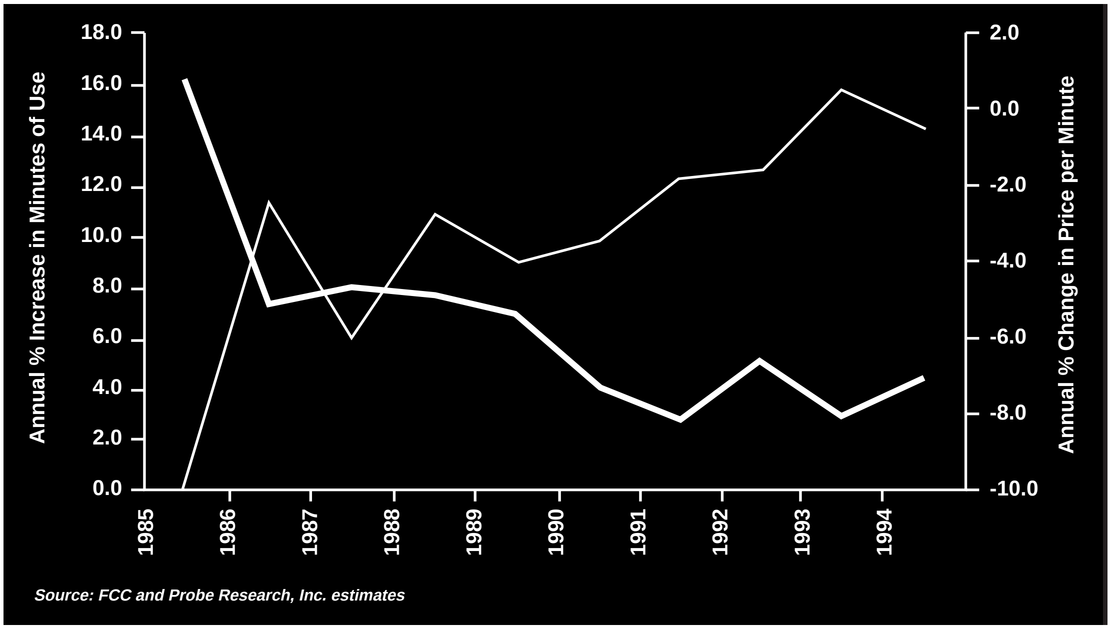

Figure 4

Store-and-forward faxing connects a server to the gateway. This server acts as the destination fax, storing the fax until it is re-transmitted to the real destination.

Using this method, faxes can be held back when there are high network loads. This also requires a great deal of storage capacity. The advantage of real-time faxing is its similarity to PSTN faxing with practically no time delay.

Why IP Telephony?

Today most of the interest in IP telephony is generated by promises of cost reduction for long distance telephone calls. Other benefits include:

- Cost reduction due to toll-bypass or toll-reduction
- Demand for multimedia communication
- Demand for integration of voice and data networks

Statistically, telephone usage seems to be very much related to price, as illustrated in the figure above from Probe Research.

Figure 4.

Historically, price sensitivity in long distance U.S. telephone traffic has been linked to minutes of use.

This linkage would most likely mean that the overall telephony usage would increase if IP telephony offers a cost-saving alternative. IP telephony may cannibalize some PSTN usage, but it will more likely motivate higher telephone usage.

The reaction that can be expected from the traditional telephone company will most likely be a reduction in long distance tariffing as a competitive equalizer, and perhaps even an entrance into the IP telephony market. Entering the IP telephony market will be motivated partly by a desire to capture this new market segment and a desire to accelerate the regulation of IP telephony.

Figure 5.

Figure 5 illustrates the growth in the total amount of telephone traffic by introduction of a lower cost service choice.

IP is used increasingly not only as a network platform but also as a defacto application platform. This differentiates voice over IP from voice over frame relay and ATM, which are used only as transport technologies.

Therefore, carrying voice over IP leads to a variety of possibilities for enhancing or

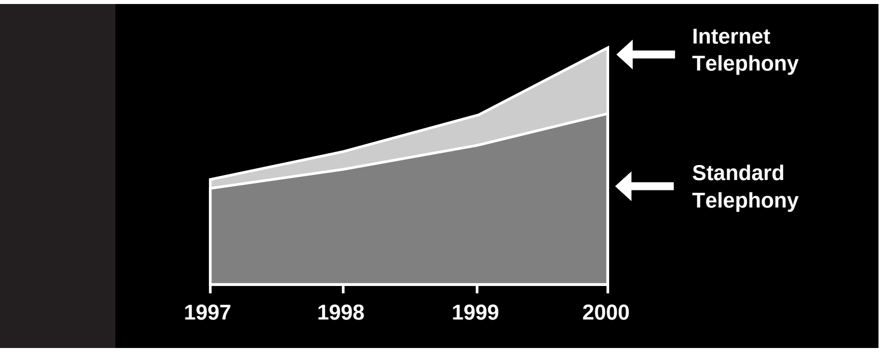

Figure 5

integrating the communication with other applications such a video, white boarding, document sharing, application sharing, teleworking, directory services, calendar tools and back office tools in general.

Additionally, communication can be integrated with other types of equipment. For example, users could dial up to a microwave oven from a phone and activate it with DTMF signaling or control appliances over a home LAN, stretching the Internet into a Homenet.

The possibilities are immense and most industry insiders agree that IP telephony is no longer just a matter of "cheap calls" but service integration and service differentiation.

Service Integration would mean that a large number of services could be supported via one network, which would enable the possibility for reduction in the operational costs and the possibility for new advanced services.

Service Differentiation is the option for operators and service providers to offer services, or bundled services, which can be used as a value-added differentiator.

Who could benefit from IP Telephony?

Since IP telephony is carried over an IP network, virtually everybody with an IP-based network could benefit from IP telephony.

Voice over IP can be offered by a traditional ISP looking for a value-added service. In such a case the ISP may be called an ITSP (Internet telephony service provider). But an increasing number of companies are emerging that only provide telephony services over IP, without offering any typical ISP services.

These telephony over IP operators are called the Next Generation Telcos. These operators typically offer phone-to-phone, fax-to-fax, phone-to-PC and PC-to-phone services, competing with the traditional switched telephone operators.

The following groups potential users into four segments with a description of the type of service required for IP telephony.

- Business/Corporation
  - Connecting sites over longer distance, e.g., Melbourne to Perth, reducing usage cost for both voice and fax.
  - Integrating the data and voice networks into one network, reducing management and operational costs, and enabling multimedia collaborative tools in order to facilitate a more efficient work environment.
  - The possibility for "Zero Administration for Telephony."
  - With a LAN-connected IP phone, any moves, additions or changes are easy tasks, simply a matter of plugging in the phone to the LAN.
  - The possibility to integrate IP telephony with web-based click-and-dial features providing both e-commerce and call center applications.

- ISP/ITSP
  - May offer IP telephony as an extra value-added service to the existing service offering.
  - May offer IP telephony as part of a bundled offering.

- Next Generation Telco
  - May offer IP telephony as a cost-efficient alternative to the existing telco offerings.

- Telco
    - May offer IP telephony in order to capture a new market segment.
    - May offer IP telephony as a reaction to competition from Next Gen Telcos or ISP/ITSPs.
    - May provide wholesale IP telephony to small operators.
    - May offer IP telephony as part of a migration strategy where voice will migrate to a data network, most likely carried over IP.

These market segments have already started to emerge, with a growing number of vendors offering a diverse range of solutions.

What are the requirements?

Each of the defined groups have their own specific requirements for an IP telephony system, and even within a specific group the requirements may vary from customer to customer.

An enterprise or corporation that wants to use IP telephony (voice and fax) between locations or to integrate the data and voice network would not necessarily require an accounting/billing function. Services such as PBX integration, call center applications and features such as Call Back and Diversion would be common requirements for these corporations.

Carrier-grade networks, on the other hand, have strict requirements for scalability or interoperability since they provide IP telephony to hundreds of thousands of subscribers spread over very large geographical areas. Such a system would not only have to carry a large amount of traffic, but also support a large and diverse set of network based services, e.g., Call Forwarding, Universal Personal Numbering, Mobility, 800 Services and other Intelligent Network services, and network capabilities such a Re-Routing and Accounting between different IP telephony operators.

Carrier grade IP telephony solution characteristics:

- Highly scalable (performance, port number and network size)
- Network management (single point management of all network nodes, with monitoring and statistics collection)
- SS7 network signaling and call control (e.g., re-routing, announcements, timers, supplementary service signalling)
- High reliability, high availability (>99.9%)
- NEBS compliant (or near NEBS)
- Real-time billing/accounting capability
- Interoperability between vendors (standards based)
- Interoperability with other carrier networks (accounting, routing, etc.)
- Very end-user friendly
- High voice quality

The voice gateway is an important component of scalability, but not the only component of a scalable IP Telephony solution.

The network architecture supported by the control server (the gatekeeper) is a more important element of scalability in a large network deployment.

The gatekeeper is the entity that performs traffic handling, e.g., routing of calls from the originating voice gateway to the destination voice gateway. The distribution of traffic control is necessary to avoid bottlenecks, requiring signaling gateways between different carriers’ networks for interoperability and exchange of accounting. The gatekeeper is the logical entity for access control, i.e., authentication of users and control of what services the users are granted.

The voice gateway is a bridge between the PSTN network and an IP network. It allows telephone calls to be passed from the circuit switched telephone network to an IP packet switched data network and vice versa.

The voice gateway performs voice-processing necessary for setting up such

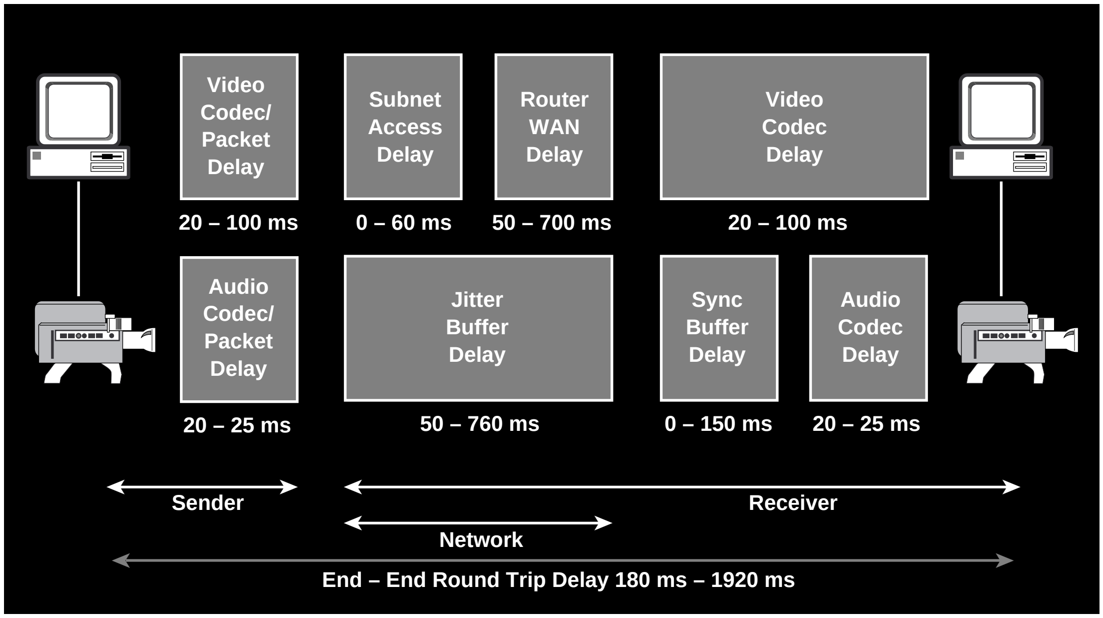

Figure 6

calls, and signaling to/from the PSTN and the gatekeeper.

One of the most important aspects of voice-processing is the real-time compression of a voice signal from a PCM G.711 format and decompression back to a G.711 format. The type of codecs used and the capability of the processing unit to handle the codec with an acceptable quality affects the voice quality.

The ITU H.323 standard is recommending the G.723, which compresses the voice signal to 5.3 or 6.3 kbit/s. A number of vendors have started using GSM compression algorithm (compressing to 13.3 kbit/s) because it appears to provide superior quality compared to G.723.

A number of different gateway types exist, as described in the following list:

- PC server-based gateways built using specialized DSP cards designed for IP telephony
- Chassis-based gateways using DSP technology
- PC-based gateways built using the CPU for voice processing
- Multipurpose NIC cards with telephony capability
- Stand-alone IP telephones

Obviously these gateways have very different capabilities, both with regard to processing power, and especially with regard to scalability. A PC-based gateway using the CPU for all the voice processing would not be very scalable compared to chassis-based gateways with DSP boards. An IP PC-based gateway also might not be able to provide the same voice quality as a DSP-based gateway.

This indicates that there has been both a fast evolution in the gateway development and that gateways do have market segmentation.

The following indicates some of the requirements for carrier-grade voice gateways.

Carrier Grade Voice Gateway:

- Very high port density per chassis (number of simultaneous calls)
- High processing power for voice compression and packetization
- Hot swap cards (repair during operation)
- Redundant DC and AC power

- Passive backplane (elimination of single point of failure)
- Redundant cooling fans
- Echo cancellation
- Silence suppression
- Low latency
- Very precise timing or clocking

It is obvious that the larger the gateway becomes, the more important the control and monitoring capabilities become. A gateway may consist of a large number of trunks, and the gatekeeper may need capabilities to identify specific trunks and connect calls over a specific trunk. Bellcore has recently proposed a protocol, Simple Gateway Control Protocol, which defines a control model for gateways and call connections via a gateway.

What are the issues?

As an emerging technology, IP telephony has a number of technological and evolutionary issues.

The technological issues are to a large extent related to the fact that the Internet Protocol was developed for data traffic, not real-time traffic, such as voice and video. The benefits of using IP as a generic platform for both data and real-time applications are compelling enough to encourage resolution of these issues.

Evolutionary issues stem from a variety of vendors developing their product portfolios according to market demands and market trends. This of course means that it takes time before the products will cover all possible market segments and interwork with the same reliability as the circuit switched network.

Voice quality

IP-based networks are developed for data applications, and do not provide true real-time capabilities for services such as voice and video. This does not mean IP telephony cannot be used without proper voice quality. It means that when the IP networks get heavily loaded, the queuing in the network routers have no natural technique for controlling the queuing delay for the voice traffic.

The end-to-end delay comes from a number of sources, but usually it's the voice gateway (or a client PC if such is used) and the routers causing delay.

Figure 6.
A number of components may contribute to the delay and jitter in a voice or video over IP session.

The voice gateway induces delay because of compression and decompression and plus packetization and depacketization. The compression algorithm often introduces a large delay because of a lack of availability of sufficient processing power.

The router-induced delay will depend on the capacity of the router and the number of router hops from originating voice gateway to terminating voice gateway.

From an end-user point of view, the delay encountered in the communication has to be below a certain threshold; otherwise, the usability of the communication will be reduced dramatically.

Figure 7.
The end-to-end delay must be below a threshold in order to provide acceptable communication for the users.

The IETF is developing features such as MPLS, Differentiated Services and Assured Services that may improve quality. These are different techniques for reducing delay and jitter in real-time traffic over a large, routed network.

In smaller networks, it is possible to use existing techniques such as RSVP (incorporated into H.323 version 2) or the IP precedence indicator (TOS) in the IP header. Monitoring and traffic engineering even out the load in the network in order to prevent any part of the network from reaching a load level where the voice quality deteriorates by routing the voice traffic through the part of the network with the smallest router delay. Techniques such as TCP Rate Control, which is used by the PacketShaper from Packeteer, controls the data traffic on behalf of the voice traffic.

Obviously, the codec set for the voice compression and the actual implementation of that algorithm is very important for the voice quality. This is an area that is constantly improving with large variations between different vendor products. This depends of course on the hardware platform being used – an advanced algorithm requires a large amount of processing. PC-based platforms, or DSP platforms with a low MIPS, will have problems handling advanced algorithms satisfactorily.

Interoperability

Interoperability relates to products from different vendors as well as different carrier networks. Interoperability issues exist mainly because standardization is not yet mature in this area. No standard exists for signaling, call or accounting and billing.

For a carrier, accounting is an issue for mainly two reasons:

- There is no standard agreement on accounting records or billing methods for IP telephony.
- A large number of operators all over the world have to establish bilateral agreements on accounting. Even in normal circuit switched telephony this can be expected to become an issue as deregulation spawns more and more new operators.

Today there seems to be a number of ways to handle the issues:

1. One is not to bother at all, and just offer IP telephony to a limited number of destinations.
2. Another is to act as a Next Generation Telco, such as Delta Three, which offers connectivity to operators via a franchise agreement for Delta Three's network.
3. A third is to use an organization such as ITXC, which would provide third-party authentication, routing and accounting. ITXC provides settlement rates and arbitrage against pre-defined international accounting rates. IP Telephony operators are thus seen as service providers to ITXC.

Accounting between the operators may be one problem, but billing of the operators' customers may prove to be just as large of an issue. The challenge for the ITSP will be to obtain accurate, timely, usage-based end-user billing, settlement and reconciliation. Today most ISPs provide a flat-rate cost scheme mainly due to the cost of real-time back office tools. The IP telephony solution should incorporate such features due to ease of management and potential cost reductions.

These issues are being handled in the standardization organizations through maturing standards such as H.323 and Tiphon from ITU.

Security

Security for voice over IP has been somewhat overlooked, but it is now being raised as a very important issue by carriers and business users.

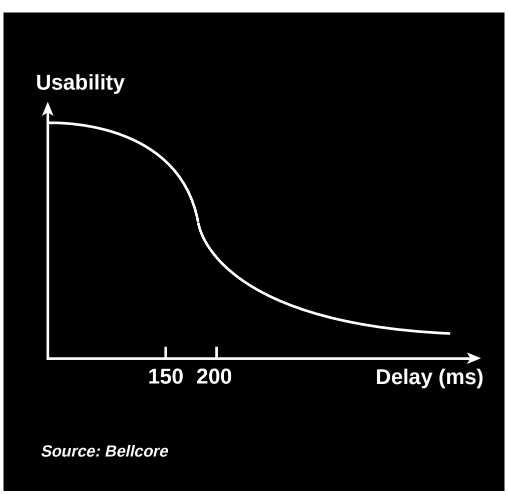

Figure 7

The basic security issues are the same as for e-commerce:

- User and data authentication
- Data privacy (integrity and confidentiality)
- Access control
- Policy management

Network security is of course related to the routing side of IP. In a public Internet, the packets can traverse through any router and can be intercepted by anyone. Acceptable security can be obtained by encryption and tunnelling.

Tunnelling can be done in a number of manners, but IEFT has decided to standardize the Layer 2 Tunnelling Protocol (L2TP). This can be used to establish a secure tunnel or VPN, between two gateways or gatekeepers.

By encrypting the payload in the packet, it is possible to secure confidentiality. The most common encryption method used today is the secure sockets layer (SSL), which may be replaced by the new transport layer security (TLS). Both of these operate on a transport level, i.e. based on an end-to-end session. This may be suitable for gatekeeper-to-gatekeeper signaling, but not if UDP transports the voice.

The IETF IPsec working group describes a security architecture for IP protocol, i.e., network level security that makes it suitable for secure VPN. IPsec authenticates the users (end users and gateways), encrypts the payload and keeps track of who has changed the packet. ITU has incorporated this into H.323 Version 2 via H.235 also called H.Secure.

Although the security can be handled by key cryptographic means, there are political (military) issues in a number of countries, which place restrictions on the export of cryptography (what size key may be used). Recently the U.S. Chamber of Commerce announced automatic approval for banks and financial institutions, with no key length restrictions or back-door provisions. This indicates that encryption will soon be generally available.

It should also be noted that the stronger the encryption, i.e., algorithm and keycode, the more processing power will be needed. This may place a security limitation on less advanced hardware platforms.

Integration with PSTN

The major issue for integration of IP telephony and PSTN is making the PSTN and the IP telephony network appear to be one network to the end-user and easy to manage by the operator.

Today IP telephony is a separate network that the end-user often must access via a two-stage dial-up, i.e., dial the gateway and receive a voice prompt for the destination E.164 number. This is not the most user-friendly mode of operation.

The IP telephony network is not yet ubiquitous, i.e., it is not possible to call anywhere via IP telephony, and the user will often need to know to which destinations the operator offers IP telephony.

It is not yet possible to route between the public telephone network and the Internet (or intranet) based on class of service, cost of service, and quality of service, leaving a number of decisions to the end-user.

ITU is addressing this issue in Tiphon, where issues such as network architecture, numbering and supplementary service integration are being specified.

Some of these issues may not seem very serious considering the increasing number of standard telephony operators. But from a carrier point of view, it will be a requirement to provide uniform and easy-to-understand service offerings.

The H.323 version 2 has incorporated handling supplementary services as specified in H.450. H.450 is based on QSIG and can be used for PBX-to-PBX signaling and implementation of services such as Call Transfer and Call Forwarding. This will facilitate Call Center applications as well as supplementary services.

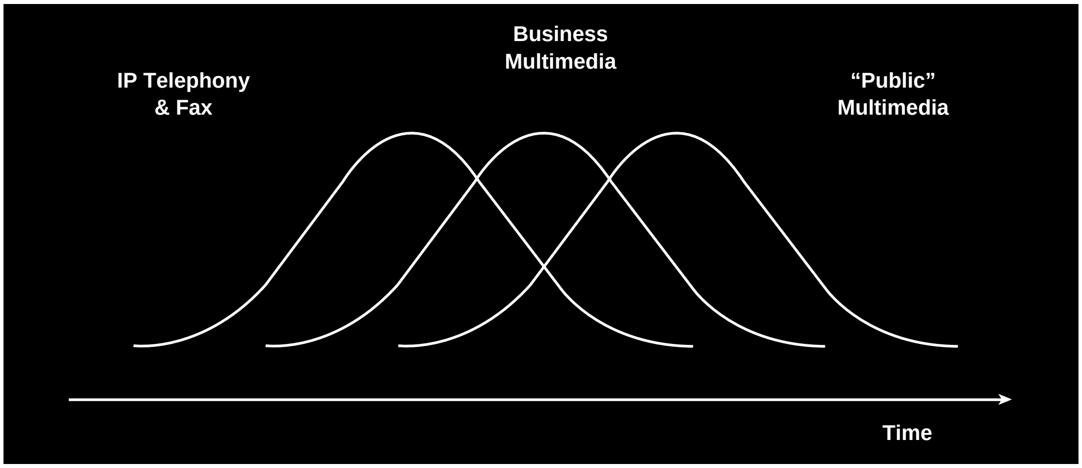

Figure 8

Scalability

Scalability relates to both the voice gateway and the gatekeeper architectures. Most gateways on the market have PC-based hardware with a limited number of ports, and most of the gatekeepers have limited capabilities for handling large hierarchical networks. For an enterprise or an ISP, this may not be the major concern, but it certainly has to be for carriers.

What will be the evolution of IP telephony?

IP telephony is a new technology and the previous chapter indicates a number of issues to be handled. This does not mean, however, that the issues cannot be handled properly. There are products appearing that address this and outline product evolutions and network migrations toward large-scale IP telephony operations.

IP telephony is leading-edge technology, being developed as the market trends and market requirements get clearer. This evolution of the IP telephony market can be expected to go through three phases, as illustrated in figure 8.

Figure 8.

The market for real-time services over IP will evolve from the current arbitrage market, to vertical business applications and eventually widely deployed multimedia applications.

This does not mean that the technology is too immature to provide benefits under the right circumstances.

A large and increasing number of the so-called Next Generation Telcos and ITSPs are offering telephone services via IP telephony. This includes operators such as Latic Communications, the Global Exchange Carrier Company (GXC), IDT, OzEmail Interline, Delta Three, Qwest, and USA Global Link, which operate with both Internet and Intranet services. Traditional telecom operators such as AT&T, Telenor and MCI have started utilizing IP telephony for both telephone and fax traffic.

In a properly designed network, the quality can be controlled enough to provide an acceptable service. Operators such as Delta Three, Qwest, IDT, MCI, etc., are all large IP telephony providers with successful operations.

Likewise, corporations around the world have started using IP telephony (both phone and fax) and IP-based video conferencing between office locations. And as e-commerce becomes more accepted, IP telephony will become an integral part of e-commerce applications with services such as web calling (calls initiated from a company's web page).

IP telephony may be a medium-term to long-term strategy, but the potential cost savings and the capability for innovative applications is an attractive proposition for both businesses and carriers. This is being enabled today, and the experience gained today is the business advantage of tomorrow.

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