

Voice over IP: Products, Services and Issues

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Abstract:

Once you are aware of the benefits and applications of Voice over IP, it is too good to resist. Perhaps that is why vendors are flooding the market with VOIP products and services. The following paper analyzes the various issues in the evolving VOIP technology and the challenges in the development of VOIP products. It then presents the features of few VOIP Products offered by the leaders in this field, how well they handle the issues and some services currently available.

See also: [VOIP Protocols](#) | [Voice over ATM](#) | [Voice over IP](#) (Lecture by Dr Jain) | [H.323 and Associated Protocols](#) | [VOIP References](#) | [Books on Voice over IP and IP Telephony](#)

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1. INTRODUCTION

"Migrate to IP or risk being left behind." This seems to be the idea in the minds of vendors who have been using circuit switching infrastructures for the transportation of voice. As you are reading this article, the Internet is being modified to support voice traffic and products are being made to link the data and voice networks. Eventually the Internet and the telephone network will be one and the same.

Internet Telephony is an emerging technology and has a number of technological and evolutionary issues. The technological issues are mainly because the Internet was not designed for real time traffic such as voice and video. The evolutionary issues stem from the fact that a variety of vendors develop their products according to market demands and supplies. It will take time for all these products to converge and inter work with the same reliability as the circuit switched networks. However 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.

The following sections describe the benefits of this technology, the issues related to the technology, the challenges ahead and also present a survey of the current VOIP products in the market, the services provided and how well they handle the issues.

1.1 Benefits of the technology

- **Integration of Voice and Data**

The integration of voice and data traffic will be demanded by multi application software. The inevitable evolution will be web servers capable of interacting with voice, data and images.

- **Simplification**

An integrated infra structure that supports all forms of communication allows more standardization and lesser equipment management. The result is a fault tolerant design.

- **Network Efficiency**

The integration of voice and data effectively fills up the data communication channels efficiently, thus providing bandwidth consolidation. The idea is to move away from the TDM scheme wherein the user is given bandwidth when he is not talking. Data networks do not do this. It is a big saving when one considers the statistics that 50% of a conversation is silence. The network efficiency can be further boosted, by removing the redundancy in certain speech patterns.

- **Cost reduction**

The Public Switched Telephone Networks' toll services can be bypassed using the Internet backbone, which means slash in prices of the long distance calls. However these reductions may slightly decrease when the Federal communications Commission (FCC) removes the Enhanced Service Provider (ESP) status granted to Internet service providers (ISPs) by which they do not have to pay the local access fees to use the telephone company (TELCO) local access facilities. Access fees form a significant part of all long distance calls. But in spite of this, the circuit switched telephony would be expensive because of lack of bandwidth consolidation and speech compression techniques. **1.2 New Applications**

- **Directory Services over Telephones**

Ordinary telephones can be enhanced to act as an Internet access device. Directory services could be implemented by submitting a name and receiving a reply.

- **Inter Office trunking over the corporate intranet**

The tie trunks between company owned PBXs could be replaced by an Intranet link and would provide large savings at a good quality of service.

- **Remote access to the office from your home**

One's home could be converted to a home office and gain access to the company's voice, data and fax services using the company's Intranet.

- **IP-based call centers**

With the advent of the Internet, companies have experienced large increase in their web site inquiries. These may not result in immediate financial transaction but atleast people get to know about their products. This is the beginning of E-commerce. With VOIP there can be interaction with the customers.

- **Fax over IP**

Real time facsimile transmission is an immediate application of Voice over IP. Facsimile services which use dial-up PSTN services are affected by high cost for long distance, analog signal quality and machine compatibility. Instead a fax interface unit can convert the data to packet form, handle the conversion of signaling and controlling protocols and ensure complete delivery of the data in correct order. [Back to Table of Contents](#)

2. IDENTIFICATION OF MAJOR SYSTEM COMPONENTS

2.1 Gateways

The gateways are the devices that communicate between the telephone signals and the IP endpoint. The IP endpoint usually speaks H.323 for media stream and more recently Session Initiation protocol (SIP). The gateways usually perform the following 6 functions

- **Search function**

When an IP gateway is used to place a call across an IP network, it receives a called party phone number. It converts it into the IP address of the far end gateway, possibly through a table lookup in the originating gateway or in a centralized directory server.

- **Connection Function**

The originating gateway establishes a connection to the destination gateway, exchanges call setup, compatibility information and performs any option negotiation and security handshake.

- **Digitizing function**

Analog telephone signals coming into a trunk on the gateway are digitized by the gateway into a format useful to the gateway, usually 64 kbps PCM. This requires the gateway to interface to a variety of Telephone-signaling conventions.

- **Demodulation functions**

With some gateways the gateway trunk can accept only a voice signal or a fax signal but not both. But sophisticated gateways handle both. When the signal is a fax, it is demodulated by the DSP back into the original 2.4-14.4 kbps digital format. This is then put into the IP packets for transmission. The demodulated information is remodulated back to the original analog fax signal by the remote gateway, for delivery to the remote fax machine.

- **Compression functions**

When the signal is determined to be voice, it is usually compressed by a DSP from 64K PCM to a 5.3 Kbps signal, which is the G.723.1 standard.

- **Decompression and Remodulation functions**

At the same time that the gateway performs steps 1-5, it is also receiving packets. Hence this function is required **2.2 Gatekeepers**

Terminals are the LAN client endpoints that provide real time two way communications. When an endpoint is switched on, it performs a multicast discovery for a gatekeeper and registers with it. Thus the gatekeeper knows how many users are connected and where they are located. The collection of a gatekeeper and its registered endpoints is called as **azone**. A gatekeeper is required to perform the following functions:

- **Address translation**

Translation of an alias address to a Transport Address using a table updated via Registration messages.

- **Admissions control**

Authorization of LAN access, using Admissions Requests or Confirm and Reject (ARQ/ARC/ARJ) messages. Access is based on call authorization, bandwidth or some other criteria.

- **Bandwidth management**

Support for Bandwidth Request, Confirm and Reject messages, or a null function that accepts all requests for bandwidth changes.

- **Zone management**

The Gatekeeper provides the above functions for terminals, MCUs, and Gateways, which are registered in its Zone of control. **2.3 IP Telephones**

These are devices, which replace the existing telephones by providing enhanced services suited to VOIP. At the same time they should retain the capabilities of the original phones to keep the user comfortable.

2.4 PC Software phones

This arrangement consists of a microphone connected to a PC interfaced by a card and running a software which permits voice and multimedia transfer over the Internet. Microsoft NetMeeting is an example.

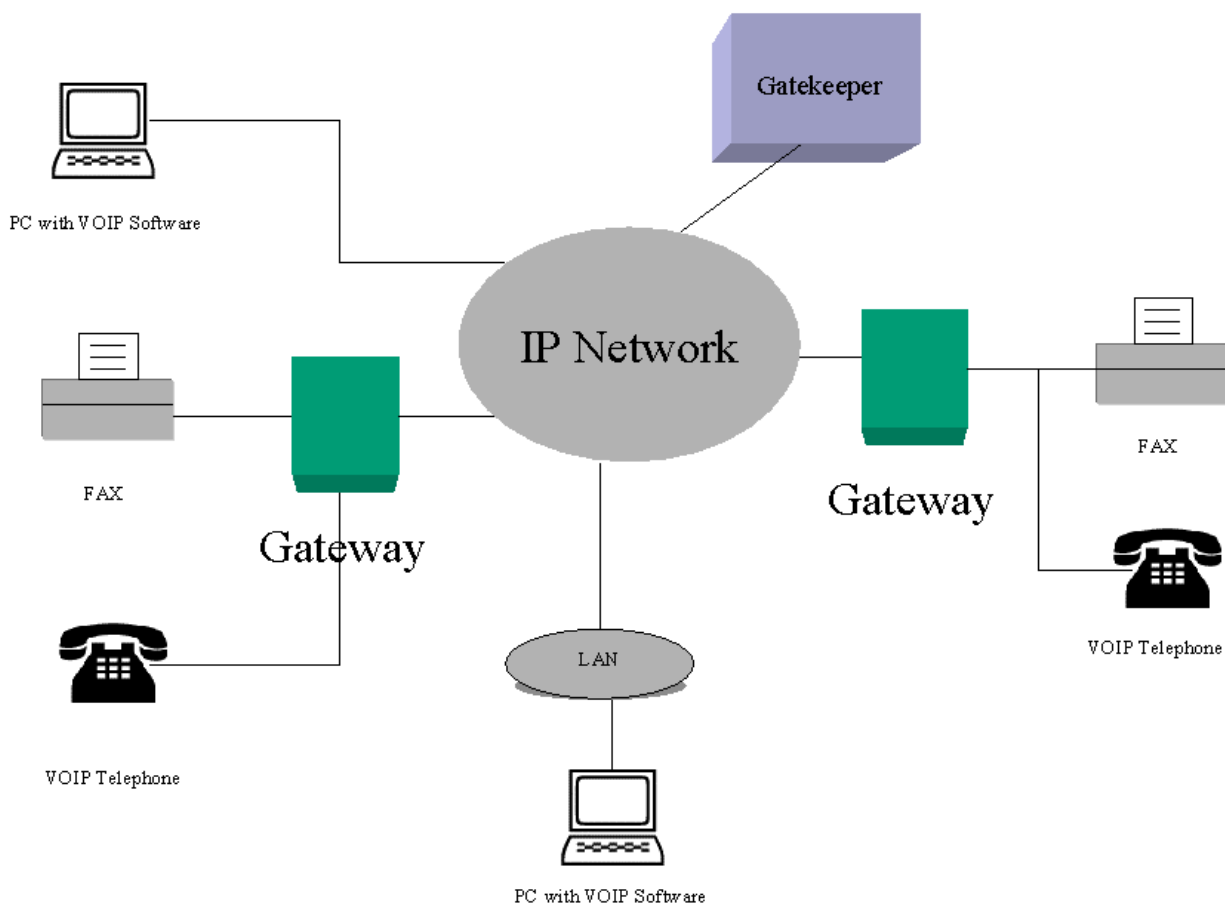


Fig.1 Components of a VOIP system

A range of the above products launched by different vendors is discussed later. But before that, the major development issues regarding these products are discussed.

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3.VOIP PRODUCT DEVELOPMENT ISSUES

In this section we discuss the points that manufacturers have to take note of while developing their products.

3.1 Voice Quality

The voice quality should be comparable to what is available using the PSTN, even over networks of varying levels of QoS. If a company thinks that reducing the bills is the criteria and adopts a poor quality VOIP service, then the only people using that service would be the Managing Director and the Accounting Officer. The employees will not compromise quality to reduce the company's bills.

The following factors decide the VOIP quality:

- **Use of a Quality CODEC**

Codec stands for Coder Decoder. It should give good voice quality and low delay. The International Telecommunication Union's (ITU's) officially recommended CODEC for all wide area networking applications is G.729

- **Echo cancellation**

When a two-wire telephone cable connects to a four-wire PBX interface or a telco central office interface, a special electric circuit called a hybrid is used to do the conversion. But in them a small percentage of telephone energy is not converted but instead reflected back to the caller creating an echo. If the delay is more than 10mS the caller hears the echo and this has to be avoided.

- **Delay**

- **Total Transmission Delay**

Total transmission delay is the sum of the compression, decompression delays, processing delay, the buffering/Queueing delay, the transmission delay and the network delay. The network delay is variable while the others can be fixed pre hand to less than 130ms. When this total delay exceeds 200ms, the two speakers have to make sure that when one speaks the other listens and pauses to make sure that the speaker is done. Bad timing may result in stepping on the other's message.

- **Delay Jitter**

Delay jitter is the variability in arrival time of a packet. When a packet does not arrive in time to fit into the voice stream going out of the far end gateway, it has to be discarded. It cannot be re transmitted, as it would delay proceedings too much. If this happens too often, then the listener will perceive reduced voice quality.

- **Delay management**

- **VOIP Packet Prioritization**

The reason VOIP works well over a corporate IP network is due more to the corporate network's low jitter than low delay. Corporate routers usually prioritize voice/fax packets either by explicit programming of the router or by using a prioritization protocol like Resource ReserVation Protocol (RSVP).

- **IP Packet Segmentation**

This is an important step required to ensure that a very long data packet does not delay the voice packet from exiting the router in a timely manner. This is achieved by programming the router to segment all out bound data packets according to the WAN access link.

- **Packet replay technique**

To allow for variable packet arrival time and still produce a steady outgoing stream of speech, at the far end the speech is not played as soon as the first packet arrives. Instead it is held in the jitter buffer for some time and then played. This adds to the overall delay. The lesser the jitter, smaller the jitter buffer time and lower the delay.

The combination of the above three techniques produces a VOIP friendly IP network. Such IP networks are called as **Managed IP networks**.

- **VOIP Forward Error Correction (FEC)**

The public Internet has substantial packet corruption and loss. Packet replay may not suffice. For this FEC can compensate for the corrupted or missing packet.

- **Intra Packet FEC**

Here extra bits are added, thus allowing the receiving end to determine which of the bits were corrupted, yielding a packet ready for play out.

- **Extra packet FEC**

Here extra information is added to each packet that allows the receiving gateway to extrapolate from the previously received good packet and reconstruct the missing or severely corrupted packet

3.2 High Bandwidth consumption

A telephone quality call or a toll quality call requires atleast 64 kbps/call. This bandwidth is impossible to dedicate on a data network for voice.

Speech compression techniques as the G.729 reduce this to around 8kbps. The IP router overhead is around 7 kbps. Thus it is 15 kbps. But modern compressors make use of an important technique called **assilence suppression**. In a typical full duplex phone conversation, only 35-40% is active. There are significant pauses between words, phrases etc. The bandwidth consumption is thus reduced by silence suppression. Ultimately voice requires only 5-6 kbps.

Silence suppression renders the line absolutely silent to the listener so much so that it sounds absolutely dead. But by inserting **Comfort Noise** or even better, by periodically sampling the background noise and regenerating it for the listener, the line sounds active.

3.3 Transparency to the user

The user need not know what technology is being used for the call. He should be able to use the telephone as he does right now.

- **Ease of configuration**

An easy to use management interface is needed to configure the equipment. A variety of parameters and options such as telephony protocols,

compressing algorithm selections, dialing plans, access controls, PSTN fall back features, port arrangement etc. are to be taken care of.

- **Addressing / Directories**

Telephone numbers and IP addresses need to be managed in a way that it is transparent to the user. PCs that are used for voice calls, may need telephone numbers. IP enabled telephones IP addresses or an access to one via DHCP protocols and Internet directory services will need to be extended to include mappings between the two types of addresses. **3.4 The TCP/UDP issue**

The voice packet is constructed as a UDP/IP packet, to avoid TCP/IP's attempt to retransmit the corrupted packet. However TCP could be a better alternative for Fax transmission simply because if lost packets occur during the negotiation of a page, the fax could be terminated. When TCP/IP is used and the host software hides the retransmission from the fax machine, there will be no impact.

3.5 Deployment of the Gateway: Trunk Contentions

At a remote site there are normally 2 to 4 VOIP connections (or trunks) from the VOIP gateway to the PBX allowing 2 - 4 simultaneous phone/fax connections between the remote site and other corporate locations. The actual number of trunks, depend upon the number of calls made per day and the total amount they consume. The number of the head quarters trunks is decided by the total number of phone calls between head quarters and the remote sites and the total number of simultaneously active calls. Usually, head quarters have a fraction of the total trunk count. The trunk contention ratio is the ratio of total remote site trunks to head quarter trunks.

3.6 Security

- **Authentication/ Encryption**

VOIP offers the potential for secure telephony by making use of the services available in TCP/IP environments. Access controls can be implemented using authentication and calls can be made private using encryption of the links.

- **Security implementation**

Security features are usually implemented using four primary components: Packet Filtering Router, Connection gateway, Address Translating firewall and Application proxy. [\[Mercer '99\]](#)

Achieving security is a complex issue. An H.323 call is made up of many different connections. In addition addresses and port numbers are exchanged within the data stream of the next higher connection. this makes it particularly difficult for address translating fire walls which must modify the addresses inside those data streams.

The firewall must be able to stand under a large number of simultaneous connections also. Detection of intruders should be possible on the inside and the outside of the firewall. **3.7 Accounting / Billing**

VOIP gateways must keep track of successful and unsuccessful calls. Call detail records should be produced. But the major issue is the suitable billing model selection. A number of billing models have been suggested

- Time-based - Metered by flow duration, time-of-day, time-of week
- Destination, distance, carrier-based IP - Rated by called and calling station IDs associated with the sequence of stages used to support the call
- QoS-based Voice over IP - reflecting established service parameters such as priority, selected QoS, and latency.

3.7.1 Future Billing Models [\[Mercer '99\]](#)

Directory-based billing applications will streamline the process of customer registration, authorization, and service provisioning without human intervention. Directory-based billing applications store user profiles, service profiles, and service policy information in the directory instead of a private datastore. That way, the directory service can maintain the security and integrity of the data in a physically distributed environment.

Other billing models currently being developed include:

- Secure Active Directory services for storage and replication of static and dynamic data
- Integrated Domain Name System (DNS) and Dynamic Host Configuration Protocol (DHCP) services for associating IP address pools with user and application profiles
- Directory-based event services for propagation of application and network events
- Cross-platform application programming interfaces for enabling disparate billing, provisioning, and management applications to securely produce and consume directory-based data

In addition to all the above points, in a public networking environment different products will need to inter work if any to any communications is to be possible. The gateway between the telephone and the VOIP needs to be highly reliable and fault tolerant. Sufficient capacity must be available in the VOIP systems to minimize the likelihood of a call blocking and mid call disconnects. The gateways must allow every device to be accessible, especially when there is mapping across different protocols and signaling systems. VOIP is likely to get very popular. In that case, the components should be flexible enough to grow to very large user populations, to allow a mix of public and private services and to adapt to legal regulations.

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4. MARKET PRODUCTS

In this section some important market products categorized as gateways, gatekeepers, IP phones and PC based software phones are discussed. Two important VOIP support products, which do not fall into these categories, are discussed in the end of the section. These are the Motorola Vanguard 6560 and Lucent Softswitch.

4.1 Gateways

4.1.1 [MICOM V/IP Gateway](#)

Features

- Uses the company's current LANs, routers and WANs
- Easily integrates into any server or desktop PC running DOS, Windows 95, Windows NT or Netware
- Flexible analog/digital connections and operating platforms
- V/IP interfaces with all the current communications equipment, from telephones and PCs to servers and routers. The benefits of Voice/ data integration are obtained at no risk of losing data or the expense of re provisioning of the network.

Operation

- The V/IP access number, destination office number and remote extension number triggers a "calling out" signal which travels from the telephone through the PBX system.
- The "calling out " signal goes into either an analog or digital V/IP voice interface card in the gateway PC.
- The V/IP does call setup based on the digits entered. The V/IP's phone data base maps the destination office number to the remote V/IP gateway's remote address.
- V/IP establishes availability of an open channel on the remote gateway. If a priority protocol such as RSVP is available, it is requested for allocation of bandwidth.
- The call is connected within 1 or 2 seconds.
- In the course of the conversation, the voice signal is digitized and compressed into IP Packets. The voice packets are sent over a router. The router treats the packets as priority IP traffic over the WAN.
- When the call is terminated, V/IP automatically deallocates bandwidth, logs call accounting records and recycles for the next call.

4.1.2 Nortel Networks CVX SS7 Gateway

Features

- Scalability

The CVX SS7 Gateway supports up to 100,000 circuits, 2,048 route/trunk groups, and 32 SS7 links (16 link sets). Because of this, service providers can grow without bothering about Signaling System 7 (SS7) hardware changes, if any.

- Cost-Efficiency

Makes it easier to leverage existing SS7 trunks, which are typically less expensive and readily available.

- Compliance to standards.

Bellcore certified to Network Equipment Building Standard (NEBS) Level 3, Earthquake Zone 4, and is Year 2000 compliant.

- Availability

Provides a highly available, Bellcore- and industry-proven platform. It includes fully redundant hardware components. Hence they are resistant to single failures.

- Network Management

Web-based network management interface (SS7View) is provided to enable fast and easy provisioning, supervising and troubleshooting. **4.1.3**

Lucent Technologies Pathstar Access server

This integrates the following components into a single system.

- Digital loop carrier
- Telephony system
- Voice over IP gateway
- Remote access server
- Digital Subscriber Line (DSL) Access multiplexer
- Edge router

It is an open platform with support for industry standard protocols such as H.323, Q.931, Signaling System 7 (SS7), Open Shortest Path First (OSPF), Border Gateway Protocol (BGP) and IP multicast.

4.1.4 [CISCO systems DE-30+ Gateway](#)

The Cisco DE-30+ digital gateway provides a connection path between the Cisco AVVID (Architecture for Voice, Video and Integrated Data) packet

telephony network and the Public Switched Telephone Network (PSTN) or a PBX, which uses digital Primary rate Integrated Services Digital Network (PRI ISDN) trunks. The DE-30+ supports 30 voice channels on an E1 interface. Gateways are administered through the required [Cisco CallManager](#). The DE-30+ gateway consists of a single PCI bus-based card and mounts in any PCI bus-PC (where it only draws power).

The DE-30+ gateway supports up to 30 simultaneous channels of voice over IP (VoIP) packet to circuit switched adaptation, G.711 encoding (A-law or m-law), dual tone multifrequency (DTMF) detection/generation, signaling, and line echo cancellation. 30 simultaneous channels of G.723.1 encode/decode are also supported.

4.1.5 3Com Gateway

Features

- 3Com's total control IP telephony gateway promises to deliver a high density, scalable platform that performs all H.323v2 compliant functionality including real time voice and call processing.
- Its modular design allows interface and application cards to be inserted and removed while the chassis remains on line, minimizing downtime.
- 3Com gateway is designed to inter operate with H.323 compliant gateways, gatekeepers and legacy or third party back end services.
- Each gateway can support up to 312 concurrent DS0 channels via T1 infrastructures or 390 calls via E1.

4.1.6 VocalTec Series 2000 Gateway

Features

- Advanced Audio Capabilities

Advanced voice packet handling strategies such as reconstruction redundancy provide enhanced sound quality. It also includes a jitter buffer (0-300 msec with controlled automatic tuning mechanism), interpolation of bad frames, Voice Activity Detector (VAD), Comfort Noise Generator (CNG) and 16/32 ms G.165 adaptive echo canceler. The G.168 adaptive echo canceler is up coming. Additional audio features include input/output gain control and selectable G.711 u-law/A-law interface.

- Management Capabilities

A standalone application, which runs on Microsoft WindowsNT, provides network management. It offers an advanced Graphical User Interface (GUI). Management can be launched from VocalTec Network Manager on a single console.

- Failure Handling and Redundancy

Failure handling, based on VocalTec Gatekeeper (VGK), ensures that Call Detail Record (CDR) information will be saved locally until the VocalTec Gatekeeper connection can be renewed. A dynamic gatekeeper search algorithm (DNS or IP-based) allows quick relocation of an available gatekeeper.

- System maintenance and monitoring

The system topology, various statistics, current equipment status are displayed. An event history browser with event log is also provided. Output relay stops traffic in case of an alarm. Terminal tumble switches allow the administrator to monitor time slots.

- Open Interfaces

The open architecture of VocalTec Telephony Gateway Series 2000 offers the option of interfacing to third party systems. Billing, Quality of Service (QoS), and Authorization, Authentication, and Accounting (AAA) Software Development Kit (SDKs) are available for third-party developers. AAA is also supported via VocalTec Gatekeeper in VocalTec Telephony Gateway Series 2000.

- Scalability

VocalTec Telephony Gateway Series 2000 supports up to 16 E1/T1 trunks, with up to 480/384 ports per shelf and 3 terminals per cabinet.

- Standards Compliance

VocalTec Telephony Gateway Series 2000 uses industry standard codecs (G.729A, G723.1, G.711, G.726, G.727, VHQC at 6.4, 7.2, 8, 8.8, 9.6 Kbps). It is compliant with the International Telecommunication Union H.323v2 standard, helping to achieve interoperability in a multi-vendor environment.

- Security

H.323 (specifically H.235) token based authentication and authorization procedures maintain network security. **4.1.7 Nuera Solutions Access plus F200 IP**

Features

- Advanced voice compression
- High bandwidth efficiency
- Call routing
- Flexible voice interfaces
- SNMP network management.

- High density, scalable architecture.
- MGCP (Media gateway Control protocol) protocol used.

4.2 Gatekeepers

4.2.1 Ericsson H.323 gatekeeper

Features

- Provides Least Cost Routing
- Provides Admission Control using Access Control lists and User profiles.
- Tracks bandwidth usage
- Provides Billing and Customer Care using a god database management system
- Service Management is web based. It is done through extensive monitoring and logging. It includes alarm and debug facilities.
- Registration Admission Support (RAS), Q.931 and H.245 signaling support
- Non RAS client support (e.g. Microsoft NetMeeting)
- Scalability through architectural design

4.2.2 VocalTec Gatekeeper

Features

- Dialing Plan Management

It provides flexible, rule based dialing plan management to ensure full control over call routing to all VocalTec Telephone Gateways. Routing can be configured using permissions, restrictions or hours of service. Least Cost Routing is supported, by assigning priorities to termination gateways. Load Balancing ensures even distribution of call load between available gateways.

- Network Security

It authenticates user ID/passwords and authenticates users who want to access the IP telephony system. Cryptographic access tokens allow secured control to network elements in compliance with the International Telecommunication Union (ITU) H.235 standard.

- Centralized Accounting and Billing is maintained.
- Database Management and Backup is done through an Oracle Database.
- It promises to accommodate networks with thousands of lines and millions of subscribers.
- Network manager can establish a gatekeeper hierarchy for networks managed by separate organizations (domains). Each gatekeeper defines its own view of the network and communicates with other gatekeepers when necessary to contact a destination outside its span.

4.2.3 Nortel networks' IPConnect

IPConnect is an Internet Telephony solution from Nortel Networks for full featured telephone services and advanced data/multimedia services.

IPConnect promises a full featured, PSTN-grade telephony over multi-service IP networks. IPConnect allows customers to take advantage of the cost efficiency, open standards, and time to market for new services promised by IP networks without sacrificing the values of traditional telephony: service richness, quality, reliability, scalability and manageability. The idea is, providing PSTN equivalency is the first step in moving to a highly advanced, integrated voice/data communications based on an IP network.

4.2.4 Elemedia H.323 gatekeeper GK2000S

The Elemedia® H.323 Gatekeeper Platform is a software package that enables rapid development of high-performance H.323 Gatekeeper applications. This modular software provides the components necessary to build H.323 version 2 compliant gatekeeper applications. It is designed to interface easily to existing systems. It also provides stand-alone services for the H.323 environment.

4.3 IP telephones

4.3.1 CISCO's IP Phones

The Cisco 30 VIP voice instrument is marketed as a full featured IP telephone for executives and managers. It provides 30 programmable line and feature buttons, an internal, high-quality, two-way speakerphone with microphone mute, and a transfer feature button. The 30 VIP also provides a large 40-character LCD display consisting of two lines of 20 characters each. The display provides features such as date and time, calling party name, calling party number, and digits dialed. An LED associated with each of the 30 feature and line buttons provides feature and line status.

4.3.2 [Selsius IP phones](#)

The Selsius IP Ethernet telephone is a device, which connects to the standard ethernet LAN jack. It gives audio quality comparable to that of a PBX telephone and is easy to use with single button access to line appearances and features. The IP telephone has many characteristics of a PC in that it can operate as a standard IP device and has its own IP address. Because the IP phone is compatible with H.323, it can talk to other H.323 devices like Microsoft NetMeeting.

4.3.3 Nokia Systems' IPCourier

IPCourier is an ethernet business telephone that delivers PBX functionality to the desktop without the PBX. Its features are multiple line appearances, speakerphone capability, programmable buttons for memory dialing and LCD display.

IPCourier also supports advanced call features such as call waiting, caller ID, forward, transfer, mute and do not disturb.

4.4 PC based software Phones

4.4.1 [VocalTec iPhone v.5.01](#)

- PC-to-Phone Communication: requires signing up with an Internet Telephone Service Provider (ITSP) and then regular telephones around the world can be called. Community Browser serves as a virtual neighborhood in Cyberspace. Direct Calling makes calling as simple as entering an e-mail address. Caller ID, Call Waiting, Muting, Blocking and Directory Assistance offer the amenities of a full featured phone.
- Live Motion Video: One can actually see the person with whom he is speaking. (No additional hardware is required to receive video).
- Audio Conferencing Support: Can talk with up to 100 people using the VocalTec Conferencing Server.
- White boarding lets one share and edit documents, photos, and drawings with other users. Text Chat lets fingers do the talking.
- Multitasking and Auto Accept Calls let Internet Phone run in the background while working.

4.4.2 [Netscape's CoolTalk](#)

CoolTalk is a real time desktop audio conferencing and data collaboration tool specifically designed for the Internet. Not only does CoolTalk provide real-time audio conferencing at either 9600 baud, 14.4k or 28.8k modem speeds, but also includes a full function White board, text based chat tool, and answering machine.

4.4.3 White Pine's CU-SeeMe Pro

- Directory Service lets one see a list of all of the users published on a particular ILS server, whether they are using CU-SeeMe Pro, CU-SeeMe Version 3.1.2, or Microsoft® NetMeeting.
- Conference Companion lets the user locate associates, friends or family online and call them without needing to know their IP addresses
- One can view up to 12 video images simultaneously
- Integrated T.120 data collaboration for sharing applications, white board, and file transfer for multi-user collaboration during conferences
- A choice of video and audio codecs for best performance over a variety of network speeds
- It is H.323 compatible So one can make point-to-point calls to users of Microsoft NetMeeting, Intel ProShare and other H.323 clients
- It is available for Windows® 95/98, Windows NT®

4.4.4 Microsoft Net Meeting

Overview

NetMeeting for Windows 95 and Windows NT is an award winning product that provides the most complete conferencing solution for the Internet and corporate Intranet. Its features let one communicate with both audio and video, collaborate on virtually any Windows based application, exchange graphics on an electronic white board, transfer files or use the text based chat program. Using the PC and the Internet, one can now hold face-to-face conversations with friends and family around the world. NetMeeting works with any video capture card or camera that supports Video for Windows.

Benefits

- **Multipoint Data Conferencing.**

Allows sharing of any Windows based application or folder with several other participants using standards based T.120 data conferencing. There is also an electronic white board, text based chat as well as file transfer capabilities.

- **Internet Audio/Video Conferencing.**

With a sound card, microphone, and speakers, NetMeeting lets one place standards based H.323 audio calls over the Internet or a corporate Intranet. Addition of a video camera permits face-to-face communication.

4.5 Motorola Vanguard Series

[Motorola Vanguard 6560](#)

This is an award winning expandable network access and concentration platform that integrates LAN, analog/digital voice and future multimedia traffic. The Vanguard 6560 Multimedia Access Device features Dual Core routing and Bridging, which results in low response times, bandwidth efficiency, quality voice transmission, and Multimedia transport capability.

It has following Voice Support features:

- 8/16 Kbps compression minimizes network bandwidth requirement
- Support for analog and digital voice port connections

Other benefits include low response times and high bandwidth efficiency

4.6 Lucent Softswitch

Overview

The Lucent technologies Softswitch is a programmable, multi-protocol software system that allows communication between different signaling systems such as SS7, SIP, H.323 and Q.931. It thus serves as a mediator between telephony and IP connectivity. The system promises to be a scalable one. It restores the simplicity for the user who is used to the current style of placing telephone calls, fax and other voice services. It also solves interoperability problems between gateways from different vendors caused by signaling and protocol incompatibilities. Thus it permits an incremental evolution of services. New protocols can be supported, by defining new device servers.

Working

It sets up a point-to-point connection across IP and PSTN networks by providing interoperability between SS7 and IP protocols. It is written in Java and works on several platforms available. Internet Service Providers can use it to offer multi-protocol telephony or telephony enabled services and both.

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5. VOIP SERVICES

With a whole range of products being launched in this field, there are a variety of services being provided to the end user. The service basically involves transferring voice from one end to the other. There are different ways though.

5.1 PC to Phone Services

These Services require a gateway on the receiving side to convert the IP packets back to Telephone signals.

- **VocalTec Surf&Call**

A good example would be the VocalTec Surf&Call. It enables Web to Phone Call center applications, promoting e-commerce. The web user sees an icon of Surf&Call and when he clicks on that he is connected to the phone on the other side through the internet via VocalTec gateway bypassing the PSTN.

- [Dialpad.com](#)

Dialpad.com has started an online VOIP service at www.dialpad.com. This offers free of cost long distance calling service without any installation of software through the Internet. Its revenue comes from online advertising. **5.2 PC to PC services**

These can be provided without a gateway on either side.

This service is obtained by a variety of software products such as

- **Microsoft NetMeeting**
- **VocalTec Iphone**
- **TaoTalk.com.**

It promotes video conferencing applications, Application share, White board etc.

5.3 Phone to Phone Services

A large number of Companies are providing long distance phone call services by means of VOIP at reduced rates. Examples are:

- AT&T's 7cents per minute any day any time offer for long distance calls in the United States. It also offers discounted international calls on purchase of the above offer.
- [America On-line](#) offers 9cents/minute service.
- [IDT Corporation](#) introduced a service, which costs 8cents/minute in US. UK-18cents, Australia 20cents, Japan 29cents/minute. These rates are 95% less than before.

A variety of calling card services to talk over long distances from anywhere, including different countries. However in many of these services which offer low rates, the quality is poor. But there are some, which use good gateways and reliable billing mechanisms.

Examples:

- **AcculinQ :**

This offers local Access in 5 Major US Cities including: Austin, Dallas, Fort Worth, Houston Texas & Denver Colorado at an extraordinarily low long distance rate of 5.9 cent per minute.

Calls to France and Germany are 11.9 cents per minute.

- **USATEL VIA ONE Prepaid Calling card:**

This card does not charge the FCC pay phone access fee. It charges 14 cents per minute in Continental USA.

[Qwest](#), AT&T, [Deutsche Telecom in Germany](#), France Telecom, [MCI](#), [Sprint](#), [Cyberlink](#), VoiceNet Global cardare some other examples.

5.4 Network Services

Here we talk about services being offered to improve the quality of transfer of IP packets. VOIP in a company Intranet is currently much better than that over the public Internet. While talking about issues, we talked about the Managed IP Network. It is believed that fiber networks will improve the quality of transfer.

5.4.1 Level 3's IP Crossroad Service

It is a nation wide IP network. This service is intended to give better multimedia transfer across the network at reduced rates. The customer is charged depending on the origination and the destination of traffic.

5.4.2 QWest

QWest Virtual Network Service enables building a virtual private network system for call networks to meet individual business needs. It is built with Qwest Macro Capacity Fiber network as a backbone and advanced architecture and includes features desired by most private users.

QWest Dedicated Internet Access provides reliable Internet connection by means of OC-48 packet over SONET IP backbone

5.5 Services for the Service Providers

5.5.1 ITXC

The company's customers and affiliates are traditional telephone companies, new competitive carriers, ISPs, prepaid calling card companies, call back companies, and newly formed Internet telephony service providers.

ITXC's WWeXchange Service networks different carriers and links every telephone in the world by using a combination IP and PSTN.

5.5.2 IP Telephony for carriers by Delta Three and Ericsson

This service, combines Delta three's IP network with Ericsson's networking hardware and software. The service will be marketed to fixed line carriers and Internet Service Providers.

5.5.3 Cisco and VocalTec to jointly provide hybrid end-to-end services to carriers and service providers

This agreement is claimed to put both companies in a unique position for offering scalable, manageable, and flexible end-to-end solutions for customers seeking innovative new services over cost-effective networks. It is supposed to combine the best of both worlds by bringing together Cisco's experience as a leading manufacturer of data networking and voice gateway equipment and VocalTec's strong reputation as a software provider and focused research and development in the area of voice services.

5.5.4 Cisco AVVID

Cisco Architecture for Video, Voice and Integrated Data is an Open systems architecture proposed by Cisco to bring about converged networking. It proposes 3 building blocks for this

- Infrastructure such as Switches and Routers
- Applications such as call control.
- Clients such as IP telephones, H.323 Video conferencing equipment and PCs

It has applications in unified messaging, Desktop IP Telephony and CISCO IP Contact centers.

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6. CONCLUSION

VOIP is growing fast. The very knowledge of the applications of this technology is enough for users and manufacturers to flock towards it. It is ideal for computer based communications and at the same time bringing down the cost of multimedia transfer. Hence VOIP products and services have flooded the market. The above paper presented the features of the products of a few major game players in the field of VOIP and how well they handle the issues.

A list of gateway vendors and links to their web sites can be found in Appendix A. A list of Group Conference Software Vendors is provided in Appendix B.

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APPENDIX A: LIST OF GATEWAY VENDORS

Vendor	Product	Vendor	Product
Cisco systems	7500 Series, Access path-VS3 systems	Ericsson	IPT

ECI Telecom	ITX 120	GlobalTel	Portal CPCI Gateway
Franklin Telecom	Typhoon	Hypercom	IP.tel 6000
Internet Telecom	Telecommunication Pro	MasterMind Technologies	MasterVox
Media Gate	Edge Commander	Global gateway group	Local Exchange Server
Innomedia	InfoGate	Netrix	Network Exchange 2410
Inter Tel	InterPrise Series, VocalNet Series	Interline	Digital Gateway
NeTrue	NeTrueCom	NetPhone	IPBX, Connect
Nokia	IP Telephony Gateway	Nortel Networks	CVX 1800 Gateway
StarVox	Stargate Server	Latic	Latnet Gateway Server
Telogy	Golden Gateway	VegaStream	Vega100, Vega200
VipNet	Telco-In-a-Box	World Telecom labs	INX
3Com	TotalControl Gateway	Ascend	Multivoice for the MAX
Cheap Call	Cheap call	Coyote Technologies	Carrier IP gateway
Computer Protocol	CpIP Gateway	Clarent	Clarent Gateway
Array Telecom	Series 3000	EFusion	Estream
DigiEurope	NetBlazer 8500	Intelliswitch	iSwitch gateway
Lucent	Packetstar	MockingBird	Nuvo500
Nuera	F200ip	Quescom	Qbox V series
VocalTec	VocalTec Gateway	Vsys	Vswitch
ViveSynergies	MultiVoIP	AreINet	i-Tone
Micom	Micom V/IP gateway	VocalData	IP*Star

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APPENDIX B: GROUP CONFERENCE SOFTWARE VENDORS

Vendor	Product	Vendor	Product
Cine Com	Cine Video	DataBeam	Meeting Tools
Engineering Consulting	ClearPhone	Dywco	Conference System
Hani Abu Rahmeh	Internet Multimedia	Honey transfer Com	HoneyCom
BoxTop	iVisit	NetSpeak	WebPhone
Tribal Voice	PowWow	VocalTec	IPhone
Wintronix	XtX Visual conference	Wincroft	DigiPhone, VideoTalk
Vox Phone	Video Vox phone	White Pine	CU-SeeMe-Pro
Labtam Communications	CollabOrator 2000	Microsoft	NetMeeting
Multitude	FireTalk	INRIA	Free Phone
John Walker	Speak Freely, http://www.speakfreely.org	Netscape	CoolTalk

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3. Lucent Technologies, <http://www.lucent.com>

4. Nortel Networks, <http://www.nortel.com>

5. VocalTec, <http://www.vocaltec.com>

6. Nuera, <http://www.nuera.com>

7. Ericsson, <http://www.ericsson.com>

8. Qwest, <http://www.qwest.com>

9. ITXC, <http://www.itxc.com>

10. Motorola, <http://www.motorola.com>

11. Delta Three, <http://www.telephonyworld.com/service/delta3/>

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ACRONYMS

BGP - Border Gateway Protocol

NEBS - Network Equipment Building Standard

OSPF - Open Shortest Path First

PSTN - Public Switched Telephone Network

SIP - Session Initiation Protocol

SS7 - Signaling System 7

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Note: This paper is available on-line at http://www.cse.wustl.edu/~jain/cis788-99/voip_products/index.html