VoIP (voice over IP - that is, voice delivered using the Internet Protocol) is a term used in IP telephony for a set of facilities for managing the delivery of voice information using the Internet Protocol (IP). In general, this means sending voice information in digital form in discrete packets rather than in the traditional circuit-committed protocols of the public switched telephone network (PSTN). A major advantage of VoIP and Internet telephony is that it avoids the tolls charged by ordinary telephone service.

VoIP is therefore telephony using a packet based network instead of the PSTN (circuit switched).

During the early 90's the Internet was beginning its commercial spread. The Internet Protocol (IP), part of the TCP/IP suite (developed by the U.S. Department of Defense to link dissimilar computers across
many kinds of data networks) seemed to have the necessary qualities to become the successor of the PSTN.

The first VoIP application was introduced in 1995 - an "Internet Phone". An Israeli company by the name of "VocalTec" was the one developing this application. The application was designed to run on a basic PC. The idea was to compress the voice signal and translate it into IP packets for transmission over the Internet. This "first generation" VoIP application suffered from delays (due to congestion), disconnection, low quality (both due to lost and out of order packets) and incompatibility. Vocal Tec’s Internet phone was a significant breakthrough, although the application's many problems prevented it from becoming a popular product. Since this step IP telephony has developed rapidly. The most significant development is gateways that act as an interface between IP and PSTN network.

What is Voice over IP?

Voice over IP (VoIP) is a blanket description for any service that delivers standard voice telephone services over Internet Protocol (IP). Computers to transfer data and files between computers normally use Internet protocol.

"Voice over IP is the technology of digitizing sound, compressing it, breaking it up into data packets, and sending it over an IP (internet protocol) network where it is reassembled, decompressed, and converted back into an analog wave form." The transmission of sound over a packet switched network in this manner is an order of magnitude more efficient than the transmission of sound over a circuit switched network.

As mentioned before, VoIP saves bandwidth also by sending only the conversation data and not sending the silence periods. This is a considerable saving because generally only one person talks at a time while the other is listening. By removing the VoIP packets containing silence from the overall VoIP traffic we can reach up to 50% saving. In a circuit switched network, one call consumes the entire circuit. That circuit can only carry one call at a time.

In a packet switched network, digital data is chopped up into packets, sent across the
network, and reassembled at the destination. This type of circuit can accommodate many transmissions at the same time because each packet only takes up what bandwidth that is necessary. Internet Telephony simply takes advantage of the efficiencies of packet switched networks.

- Gateways are the key component required to facilitate IP Telephony.

- A gateway is used to bridge the traditional circuit switched PSTN with the packet switched Internet. The gateway allows the calls to transfer from one network to the other by converting the incoming signal into the type of signal required by the network it is required to send it on. For example, A PC user wishes to call someone using a conventional phone. The PC sends the IP packets containing digitized voice to the gateway.

**Requirements of a VoIP**

The requirement for implementing an IP Telephony solution to support Voice over IP varies from organization to organization, and depends on the vendor and product chosen. The following section aims to identify the fundamental requirements in the general case and is split into 3 sections:

- Software Requirements
- Hardware Requirements
- Protocol Requirements

**Software Requirements**

The software package chosen will reflect the organizational needs, but should contain the following modules as defined in the Technology Guide Series - Voice Over IP Publication, and other sources.

- Voice processing module
- Pcm Interface:
- Echo Cancellation
- Idle Noise Detection
- Tone Detector
- The Packet Voice Protocol
- Voice Playback Module.
• Call Signaling Module
• Packet Processing Module.

The most obvious requirement is the existence (or installation) of an IP based network within the branch office gateway is required to bridge the differences between the protocols used on an IP based network and the protocols used on the PSTN.

➤ Protocol Requirements

There are many protocols in existence but the main ones are considered to be the following:

H.323 is an ITU (International Telecommunications Union) approved standard which defines how audio/visual conferencing data is transmitted across a network. H.323 relies on the RTP (Real-Time Transport Protocol) and RTCP (Real Time Control Protocol) on top of UDP (User Datagram Protocol) to deliver audio streams across packet based networks.

G.723.1 defines how an audio signal with a bandwidth of 3.4KHz should be encoded for transmission at data rates of 5.3Kbps and 6.4Kbps. G.723.1 requires a very low transmission rate and delivers near carrier class quality.

G.711. The ITU standardized PCM (Pulse Code Modulation) as G.711. This allows carrier class quality audio signals to be encoded for transmission at data rates of 56Kbps or 64Kbps. G.711 uses A-Law or Mu-Law for amplitude compression and is the baseline requirement for most ITU multimedia communications standards.

Real-Time Transport Protocol (RTP): it is the standard protocol for streaming applications developed within the IETF (Internet Engineering Task Force).

Resource Reservation Protocol (RSVP): it is the protocol which supports the reservation of resources across an IP network.

➤ Hardware Requirements

The gateway takes a standard telephone signal and digitizes it before compressing it using a Codec. The compressed data is put into IP packets and these packets are routed over the network to the intended destination.
network. RSVP can be used to indicate the nature of the packet streams that a node is prepared to receive.

**How VoIP works**

**Part 1**

Let us look at very simple VoIP call. Consider two VoIP telephones connected via an IP network. In this example both VoIP telephones are connected to a local LAN. Sally’s phone has an IP address of 192.168.1.1, Bill’s phone is 192.168.1.2, the IP addresses uniquely identify the telephones. Both our phones are configured to use a widely used VoIP standard called H.323.

Bill wants to talk to Sally and his phone knows the IP address of Sally’s phone. Bill lifts the handset and ‘dials’ Sally, the phone sends a call setup request packet to Sally’s phone, Sally’s phone starts to ring, and responds to Bill's phone with a call proceeding message. When Sally lifts the handset the phone sends a connect message to Bill's phone. The two phones will now exchange the data packets containing the speech. At the end of the call Bill replaces his handset and phone stops sending voice data sends a disconnect message and Sally's phone responds with a release message. The call is now complete. All the messages contain the Q931 ISDN protocol.

Having introduced VoIP I will now talk about three main 'types' of VoIP installed in the market place today.

**part 2 : The Protocols.**

I have made an assumption that both ends of a VoIP telephone conversation are compatible. This compatibility only happens if both ends agree to use the same protocol. All manufacturers who claim to be producing industry standard voice over IP either support SIP or H.323 protocol.

**So what is H.323 ?**

Over the next few years, the industry will address the bandwidth limitations by upgrading the Internet backbone to asynchronous transfer mode (ATM), the switching fabric designed to handle voice, data, and video traffic. Such network
optimization will go a long way toward eliminating network congestion and the associated packet loss. The Internet industry also is tackling the problems of network reliability and sound quality on the Internet through the gradual adoption of standards. Standards-setting efforts are focusing on the three central elements of Internet telephony: the audio codec format; transport protocols; and directory services.

**H.323 CALL SEQUENCE**
part 3: Encoding

The call control part of H.323 sets up the parameters for the full duplex voice path between source telephone and destination telephone. I will continue with my analogies to explain how your voice gets transported across the Internet.

In terms of H.323 there is a trade off between call quality and bandwidth, in general the higher the quality the greater the bandwidth required.

During the call setup portion of H.323 the phones have to decide which speech encoder/decoder to use when they send the speech to the other phone, Bill and Sally both have phones that support G.723.1, G.711 and G729.

part 4: Hear the Quality.

The performance of the speech encoders at each end, the number of packets lost on route, Latency and Jitter.

I have already talked about the encoders in the previous section. I also bundle into the encoding process echo suppression. In the early days of voice calls via satellite there would be an annoying echo. As the technology improved the echo disappeared. Echo suppression is very key to good quality VoIP calls. I do not dwell on the subject since the mathematics is beyond my comprehension. Good echo suppression makes for quality calls.

Be warned that because a manufacturer has a G.723.1 encoder it may not sound the same as another manufacturer who claims to have G.723.1, quality does vary. As a general rule the occasional lost packet will not affect too drastically the quality of a call, but lose 5 in a row and an entire word is lost and this will be a problem. So if you are going to have lost packets make sure they are only lost in a regular distributed manner. 5% lost packets distributed evenly will not result in the loss of words lose 5% of the words by clustering the packets and the effect is bad.

Advantages of VoIP

- Single network infrastructure.
- VoIP uses "soft" switching
- Simple upgrade path
- Bandwidth efficiency.
➢ Only one physical network is required to deal with both voice/fax and data traffic instead of two physical networks.

➢ Lower physical equipment cost

Disadvantages of VoIP

➢ The Internet is not the best medium for real time communications.

➢ Solutions which aim to negate the degradation suffered due to transit delays.

Opportunities

Many vendors offer the ability to incorporate Virtual Private Networking (VPN) with relative ease into the IP Telephony solutions they provide. This allows any transmission to be encrypted using a number of cryptographic techniques and providing security by transmitting the communications through a 'tunnel' which is set up using PPTP (Point-to-Point Tunneling Protocol) before commencing communications.

IP Telephony allows companies to exploit Computer Telephony Integration to its full extent. The convergence of communications technologies allows greater control over communications, most vendors provide logging and accounting facilities whereby all usage can be monitored.

Conclusion:

Without a doubt, the data revolution will only gain momentum in the coming years, with more and more voice traffic moving onto data networks. Vendors of voice equipment will continue to develop integrated voice and data devices based on packetized technology. Users with ubiquitous voice and data service integrated over one universal infrastructure will benefit from true, seamless, transparent interworking between voice and all types of data.

Reference: