# INTRODUCTION

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

# HOW IT WORKS

VOIP technology involves the following stages:

* **Digitizing sound**
* **Compressing digitized sound**
* **Breaking up** **digitized sound**  **into data packets**
* **Sending the packets over an IP** **(internet protocol) network**
* **Reassemble**
* **Decompression**
* **Digital to analog conversion (**converted back into an analog wave form**)**.

VoIP can also be used to make calls to the traditional circuit switched PSTN. **Gateways** are 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, if 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 VOIP

The requirements for implementing an IP Telephony solution to support Voice over IP are categorized as follows:

* **Software requirements**
* **Hardware requirements**
* **Protocol requirements**

## Software Requirements

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

**Voice Processing Module** - This aspect of the software is required to prepare voice samples for transmission. The functionality provided by the voice processing module should support:

* **A PCM Interface** - required to receive samples from the telephony interface (e.g. a voice card) and forward them to the Voice Over IP software for further processing.
* **Echo Cancellation** is required to reduce or eliminate the echo introduced as a result of the round trip exceeding 50 milliseconds.
* **Idle Noise Detection** is required to suppress packet transmission on the network when there are no voice signals to be sent.
* **A Tone Detector** is required to discriminate between voice and fax signals by detecting DTMF (Dial Tone Multi frequency) signals.
* **The Packet Voice Protocol** is required to encapsulate compressed voice and fax data for transmission over the network.
* **A Voice Playback Module** is required at the destination to buffer the incoming packets before they are sent to the Codec for decompression.
* **Call Signaling Module.** This is required to serve as a signaling gateway which allows calls to be established over a packet switched network as opposed to a circuit switched network (PSTN for example).
* **Packet Processing Module**. This module is required to process the voice and signaling packets ready for transmission on the IP based network.
* **Network Management Protocol**. Allows for fault, accounting and configuration management to be performed.

## Hardware Requirements

The list below highlights the most general hardware required.

* **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. 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.
* **The PC's attached to the IP based network.** These require the voice/fax software outlined above. They also require Full Duplex Voice Cards which allow both communicating parties to speak at the same time - as often happens in reality.
* **IP Telephones** (As an alternative to installing Voice Cards on PC’s) can be attached to the network to facilitate Voice Over IP.
* **A secondary gateway**. This should be considered as a backup in the event of the failure of the primary gateway.

## 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. The VoIP Forum as the baseline Codec for low bit rate IP Telephony has chosen this encoding technique.
* **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)** is the standard protocol for streaming applications developed within the IETF (Internet Engineering Task Force).
* **Resource Reservation Protocol (RSVP)** is the protocol which supports the reservation of resources across an IP network. RSVP can be used to indicate the nature of the packet streams that a node is prepared to receive.

# PROS AND CONS OF VOIP

## Advantages of VoIP

* **Single network infrastructure.** When installing VoIP in the office only a single cable is required to the desk, for both telephone and data, thus eliminating separate telephone wiring.
* **VoIP uses "soft" switching** which eliminates most of the legacy PBX equipment, thus reducing the cost of installing a communications infra-structure and the maintenance cost once installed.
* **Simple upgrade path.** The VoIP PBX technology is software based. It is easier to expand, upgrade and maintain than its traditional telephony counterparts.
* **Bandwidth efficiency**. VoIP can compress more voice calls into available bandwidth than legacy telephony. IP Telephony helps to eliminate wasted bandwidth by not transporting the 60% of normal speech which is silence.
* **Supported by most platforms** and is independent of the transport protocol used.
* **Only one physical network** is required to deal with both voice/fax and data traffic instead of two physical networks. Having only one physical network has advantages which include **lower physical equipment cost** and **lower maintenance costs**.

## Weaknesses of VOIP

* **The Internet is not the best medium for real time communications.** Individual packets can take different routes and varying delays can be encountered and packets lost in transit. Waiting for delayed packets or retransmission of lost packets can result in considerable degradation of quality.
* **Heavy congestion on the network** can result in considerable degradation of service as IP is not good at providing QoS (Quality of Service) guarantees.