Voice Over IP (VoIP)

What is Voice Over IP (VoIP)?

Voice over IP (VoIP), or Internet Telephony, is technology that uses data packets to transmit voice over the Internet. It consists of 3 steps:

- the conversion of the analog voice into IP packets, using a digital signal processor (DSP) embedded in the phone or using a PC in case of soft phones, which are software programs used with a hand set or headset connected to a PC,
  - transmitting IP packets over a packet or data network (where the packets travel along with other VoIP packets, data packets, Video over IP packets, etc.), which are sent to appropriate destinations by routers and switches,

conversion of IP packets back to voice (using DSPs or computers).

With the use of appropriate gateways (devices that do appropriate conversions), telephone calls can be made from a

- soft phone to a soft phone,
  - soft phone to a POTS (plain old telephone system, i.e., regular) phone and vice versa, and

POTS phone to a POTS phone

The traditional voice network assigns a dedicated circuit to a telephone call for its duration. Although this produces very reliable telephone calls, assigning a dedicated circuit to each telephone call makes very inefficient use of bandwidth. VoIP uses compression techniques (to make very efficient use of available bandwidth. For example, by using the G.723 compression protocol, a VoIP call can be transported using one-tenth the bandwidth of a traditional voice call, which uses a 64 kbs channel.

VoIP packets must be delivered to the receiver in real time and reliably. Sending VoIP packets over public IP networks (as opposed to private IP networks) typically generates voice connections of a very poor quality. This is because voice packets are intermingled with (typically low priority) data packets and both kinds of packets have to compete for bandwidth. Voice packets may be delayed which causes jitter. Network congestion may lead to some packets being dropped which causes clicking and popping sounds. Good VoIP implementations must alleviate these problems.

The public IP network was not designed for real-time packet transmission and typically does not reserve extra bandwidth to ensure that there are no transmission delays. Data traffic is bursty and can be very large (as is the case when downloading a large PowerPoint file). In such cases, voice packets are delayed, as they have to compete for bandwidth with the data packets and may not be delivered in real time.

To address this issue, VoIP packets must be given a high priority as compared to data packets. In other words, voice packets should be given better quality of service (QoS). Several QoS mechanisms have been proposed:

- diffserv (uses a byte that tells the router on how to treat the packet),
    - IPv6 (uses priority bits and flow labeling that tells the router on how to treat the packet), and

  RSVP (uses bandwidth reservation).

Sending voice and data over the same network facilitates the building of multimedia applications such as converting voice mail to email, unified messaging, enables easier multi-media interactions with customers (simultaneous establishment of a voice and Web connection with a single click).

Use of VoIP telephony or simply IP telephony requires that the sender and receiver have one of

- IP phones (phones that convert voice to IP packets before sending them on to the network)
    - soft phones (Microsoft's Windows XP operating system will provide built-in support for IP telephony).

  access to gateways that convert analog voice to IP packets and then back to voice – IP packets are transmitted between the gateways.

What is the Advantage of VoIP?

VoIP benefits users/customers, service providers, and equipment makers:

- Users benefit by new applications based on converged voice and data networks. For example, VoIP
    - eliminates long distance charges over LANs and WANs.
    - no need to maintain separate voice and data networks in the enterprise,
    - enables web call centers,
    - supports multi-media conferencing and collaboration, and
    - provides unified messaging (using a phone or PDA to access both email and voice mail; appropriate conversions are performed using text-to-speech and speech recognition systems).
- Service providers benefit with more efficient use of their networks. For example,
    - voice packets can be sent along with data over a single network,
    - compression techniques can be applied to send more voice packets over the same bandwidth thus enabling more VoIP connections (normal voice channels require 64 kbs; this bandwidth can be reduced to 6.4kbs by using compression techniques such as G.723; further compression has a noticeable negative effect on voice quality), and
    - a single network can be used for transmitting both voice and data.

- Equipment makers will benefit from the new types of equipment and software. For example,
  - VoIP gateways for converting analog voice to IP packets and vice versa. This allows the use of PSTN for the last mile connection to the phone users, and
  - IP phones (phones which convert voice to IP packets and vice versa; no an IP connection is needed for such phones – a voice connection is not required).

Where will VoIP be used?

VoiP will be used, for example, in

- backhaul carrier networks,
  - to avoid very high tariffs on normal international voice connections over the public telephone networks (many countries have very high telephone tariffs) – VoIP packets go over the data networks and are not subject to voice telephony tariffs; using VoIP over private networks can save up to 80% in costs,
  - converged data and voice networks,
  - converged data and voice appliances such as IP PBXs, and

always-on cellular phones.

Some Standards Used in VoIP

- G.711: an international standard used for encoding (packetizing) telephone voice at either 56kbs or 64 kbps – this is uncompressed digitized voice.
  - G.723: a protocol for compressing voice to 6.4 kbs or 5.3 kbs. The compression quality is very good with voice quality as good as normal telephone voice quality. It is supported by virtually all IP telephone equipment.
  - H.323: Signaling & telephone services protocol for the transmission of IP packets representing any combination of voice, video and data. H.323 is designed for operation over existing IP networks. Includes facilities call setup signaling and media control. Allows VoIP equipment to interoperate.
  - SIP (Session Initiation Protocol): A signaling & telephone services protocol similar to, but simpler than, H.323.

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