Voice over IP
White Paper
Revision 1.0

Introduction

Historically, telephone calls were made through Public Switched Telephone Networks (PSTN), which provided high-quality voice transmission between two or more parties. However, since the demand for data traffic is growing faster than voice traffic, we've seen a gradual shift towards packet-based networks like IP, ATM and Frame Relay. Packet-based networks provides high cost-benefit ratio and an increasing number of businesses are realizing the value of transporting their voice circuits over IP networks to reduce expenses. Saving, coupled with exceptional Quality of Service (QoS), are synonymous with Voice over IP (VoIP).

Voice over IP technology

VoIP systems digitize and transmit analog voice signals as a stream of packets over a digital data network. IP networks allow each packet to independently find the most efficient path to the intended destination, thereby using the best network resources at any given instant. Packets associated with a single source may take many different paths to the destination when traversing the network. With the different paths, arrivals will vary greatly due to delays; they may arrive out of sequence or possibly not arrive at all. At the destination, the packets are re-assembled and converted back into the original voice signal. VoIP technology insures proper reconstruction of voice signals, compensating for echoes made audible due to the end-to-end delay, for jitter and for dropped packets.

The IP network used to support IP telephony can be a standard LAN, a network of leased facilities or even the Internet. Although very appealing due to cost considerations, the Internet is constantly plagued by congestion problems and uncontrollable packet delays and losses. Therefore, dedicated networks provide a more reliable method of VoIP communications with guaranteed bandwidth available and manageable Quality of Service.

VoIP Gateways

VoIP technology rides over a pure IP network but individual calls can be made or received using standard analog, digital and IP phones. VoIP gateways, usually a dedicated server, serve as a bridge between the PSTN and the IP network. A call can be placed over the local PSTN network to the nearest gateway server, which digitizes the analog voice signal, compresses it into IP packets and moves it onto the Internet for transport to a gateway at the receiving end. With the use of VoIP gateways, computer-to-telephone calls, telephone-to-computer calls and telephone-to-telephone calls can be made with ease.

Access to a local VoIP gateway for originating calls can also be supported in a variety of ways. For example, a corporate PBX (Private Branch Exchange) can be configured so that all international direct dialed calls are transparently routed to the nearest gateway. High-cost calls are automatically supported by VoIP to obtain the lowest cost.

To ensure interoperability between different VoIP gateways manufacturers, gateways must contain a standard common call setup and control protocol. H.323 is a family of software-based standards that define various options for compression and call control. VoIP equipment should comply with the

H.323 standard that the ITU defines terminals, equipment and services for multimedia communications over networks, such as the Internet, that cannot guarantee a high Quality of Service. Other call setup and control protocols being utilized, discussed and or standardized include MGCP, SIP and Megaco.

Why use VoIP ?
Originally regarded as a novelty, IP telephony is attracting more and more users.
VoIP offers:

- Tremendous cost savings relative to the PSTN - Remote offices and users can bypass long-distance carriers and their per-minute usage rates and run their voice traffic over the Internet for a flat monthly Internet-access fee.
- Integrated infrastructure - Small businesses are able to deploy one network for voice and data-communications, further reducing costs.
- Scalability - These systems are modular and can be scaled according to the needs of users.

What is the future of VoIP ?
The market for VoIP products is established and is in a rapid growth phase. End user demand is expected to grow rapidly over the next 5 years and according to recent research, it is expected that VoIP will be deployed by 70% of fortune 1000 companies. The immediate goal for VoIP designers, manufacturers and service providers is to reproduce existing telephone capabilities at a significantly lower "total cost of operation" and to offer a technically competitive alternative to the PSTN. However, telephony over the Internet cannot make compromises in voice quality, reliability, scalability and manageability. It must also work seamlessly with telephone systems worldwide. Future extensions will include innovative new solutions including conference bridging, voice/data synchronization, combined real-time and message-based services, text-to-speech conversion and voice response systems.

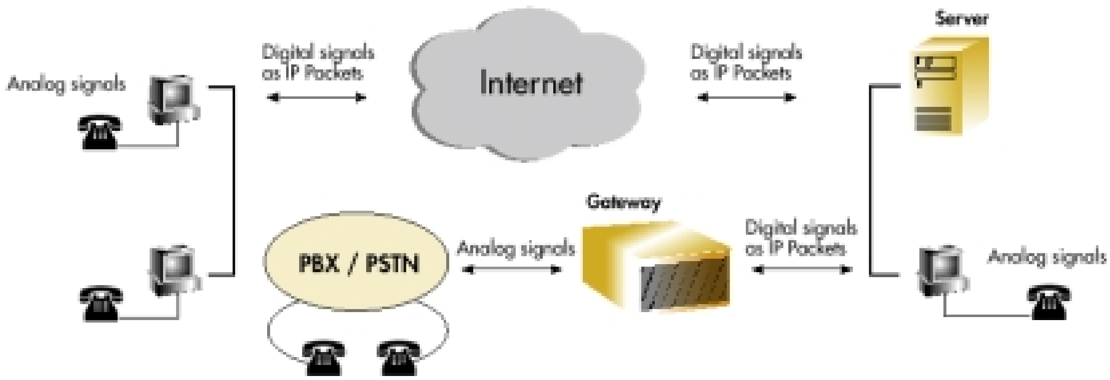
Summary
In a relatively short period of time, Internet telephony has advanced rapidly. This appealing technology, which allows users to perform calls with a fraction of the cost of regular telephone calls by using IP networks to carry the voice signals, is growing ever more popular. Many developers and manufacturers now offer PC telephony hardware and software but also gateway servers are emerging to act as an interface between the Internet and the PSTN making VoIP calls almost an obvious choice.

VoIP Myths and Reality
Myth: The voice quality of a VoIP telephone call isn't comparable to a PSTN.

Fact: In a dedicated network, voice quality of a VoIP network is extremely clear and can be compared to a call placed over the PSTN. However, Internet-based VoIP calls are not yet up to par with the PSTN quality.

Myth: Anyone can intercept my call since it is transmitted over the Internet.

Fact: Many software and hardware VoIP solutions offer an easy and secure way of encrypting calls, eliminating chances of interception.



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