Internet Telephony - Services, Technical Challenges and Solutions

Executive Summary

The rapid acceptance of the Internet as a universal communications network has increased overall interest in Internet based technologies for transport of real-time voice traffic. IP telephony is the major packet based technology for voice transport over IP based networks. As such it opens up new opportunities to telecommunications carriers, enterprises, small business, and single users. Extremely flexible by design, IP telephony can easily be bundled with a number of other services for delivering complete communications solutions to users. To take full advantage of that technology, however, companies have to address a number of technical problems in order to ensure consistent delivery of high quality voice. This paper discusses various new services based on IP telephony, technical challenges in implementing that technology, and available solutions.

Introduction

There are two major technologies for building telecommunications networks - circuit and packet switching. The traditional PSTN telephone system is based on establishing a circuit switched connection between end points for the duration of the call. Because of the physical connection between the end points, service providers can easily guarantee quality of service (QoS). Packet switching, on the other side, is a technology developed to transport data packets between end points. Therefore most of today's packet switched networks, like the Internet are not designed to natively handle real-time data transmission.

In recent years, however, the mass adoption of Internet based communications has given rise to an increasing interest in the possibility of transporting real-time voice and video traffic over the Internet. Because the Internet was originally designed as a data transport network, real-time communications applications face a number of technical issues that need to be resolved to ensure successful service offerings. The major problem with IP telephony comes from the difficulty in implementing guaranteed Quality of Service (QoS) solutions.

Applications and Services

Although implementing IP telephony solutions could be challenging it could be also very rewarding as the technology offers a number of advantages over traditional telephone networks. For example, packet based telephony allows the offering of integrated voice, fax and data services over the same network. Below we describe some of the major IP telephony based emerging applications.

Unified Messaging

In large businesses, effective communication among employees and with external customers is critical for the organizational success. Today employees use a number of communications channels to keep in touch with colleagues and customers. E-mail, telephone, mobile phone, fax, and voicemail are some of the most widely used communications methods. As communications channels continue to proliferate, their integration becomes very important.

In a typical business environment, employees rely on multiple communications channels which sometimes confuses customers. The development of IP telephony creates possibilities for implementing unified messaging systems within enterprises and beyond. In such systems, all user specific communications are sent to a single location which s/he can access at his convenience. For example, voice mail can be forwarded as an attachment to emails, the system can "hunt" users switching between their wireline, wireless and other communication devices until it the user receives the call.

Low-cost Voice Calls

IP telephony uses packet switched network for voice transport which it shares with other computers and network appliances. As VoIP relies on existing networks, the incremental cost of transporting voice is very low. There are several scenarios for conducting a VoIP calls: (1) PSTN-VoIP-PSTN, (2) VoIP-PSTN, and (3) VoIP-VoIP. Scenarios 1 and 2 require conversion between PSTN signal and VoIP data packets which is typically done through a gateway[1] owned by the service provider. Costs of voice calls are substantially lower than calls routed through the traditional PSTN network as call transportation is almost free (through the Internet). The only costs to the service provider are those for PSTN-VoIP conversion and termination. Scenario 3 allows for "free" calls which are conducted directly between two VoIP enabled devices (e.g. IP phones, PDAs, computers, etc.).

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Forward Error Correction

Under forward error correction, redundant packet information is included in subsequent packets. If the original packet is lost, it can be reconstructed from the information in subsequent packets. There are two possible scenarios - 1) The redundancy is independent of the data stream, and (2) The redundancy uses the stream characteristics to enhance the repair process. Scenario 2) requires a mechanism within the existing protocols for media transport (Real-Time Transport Protocol (RPT) in IP telephony) to carry redundant voice information.

NETWORK JITTER

Jitter refers to the variability in the frame arrival times at the receiving point. It occurs due to variability of queuing delays in the network and propagation delays in connections utilizing LEO satellites. If a voice frame is not received on time it is considered lost by the receiver. In general the greater the variability in the frame arrival time, the worse the quality of the voice gets.

To partially correct for network jitter, receivers typically hold the first packet in a jitter buffer for a given period of time before playing it out. The amount of the hold time is the measure of the size of the jitter buffer. Proper selection for jitter buffer size is critical for IP telephony systems. An optimal buffer size balances between reducing the jitter and increasing the overall voice delay within tolerable levels. If the buffer size is set too low, some packets are still lost, but if it is set too high, that results in higher delays. Typical buffer sizes range from 50 to 100 ms. While older VoIP systems rely on constant jitter, newer implementation allow dynamic change of the buffer size which improves voice quality.

CONCLUSION

IP telephony enables service providers to offer an entirely new communication services. In order to make IP telephony quality comparable to that of the traditional PSTN, however, there are a number of technical problems that need to be addressed. Solutions to those problems do exist and as new ones are developed that will further accelerate the rate of IP telephony penetration.

About SysMaster Corporation

SysMaster is a leading vendor of voice, video, and data equipment, serving emerging and traditional telecoms and service providers. The company offers extensive line of Voice-over-IP, IPTV, and Wireless products and solutions which enable service providers to build robust and scalable networks for delivery of next generation services to subscribers. SysMaster's innovative solutions are successfully deployed in over 60 countries worldwide.

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