WHITE PAPER

IP Telephony

Introduction

This paper will address three areas of Internet protocol (IP) telephony. The first is how IP telephony evolved and how it is being integrated into the public switched telephone network (PSTN). The second is a discussion of Internet Call Diversion (ICD). The third and final part examines how converged PSTN and IP telephony networks can be integrated at the services layer.

The Evolution of IP Telephony

The six-stage evolution of IP telephony started in 1995. In stage one, the IP Telephony industry introduced personal computer (PC) software that allowed users to make telephone calls over the Internet. Both ends of the connection needed the same software. It was an economical way to make long-distance international calls. The cost was several hundred dollars up front but allowed unlimited free telephone calls over the Internet.

Stage two allowed phone calls over the local enterprise companies' intranet network. Stage three connected intranets to the Internet for low-cost voice-over IP (VoIP) between enterprise intranet networks. In 1998, we also began to see common nomenclature as well as interoperable standard products. The media gateway (MG) and media gateway controller (MGC) were introduced as IP elements that we considered separate from the underlying PSTN.

Stage four began the integration at the transport level between the underlying private IP network and the PSTN. Because there was no Signaling System 7 (SS7) connectivity between the two networks, this required a two-step dialing process where users had to dial an 800 number to access the IP Telephony MGC. Then after they dialed the 800 number, they would dial the called party phone number. The rates offered were significantly lower than those offered in the traditional interexchange carrier (IXC) market.

Stage five commenced when companies began deploying softswitch-based architectures as part of the public network. The softswitch-based architectures were not quite yet part of the PSTN. By contrast, a competitive local-exchange carrier (CLEC) that puts in a traditional Class 5 switch is considered part of the PSTN. The difference is merely conceptual. Stage five had PSTN interoperability but not integration. Stage five is in its infancy at the time of this publication.

Stage six will finally introduce PSTN/IP network integration, with full SS7 connectivity and complete interoperability between the IP telephony network and the traditional PSTN service switching point (SSP)-based network. These two networks will coexist and inter-operate at the transport and services levels for a long time. The industry is entering stage six today. Carriers are beginning to deploy softswitch-based architectures and taking advantage of the economics of packet networks.

Internet Congestion Relief

The Internet has had a large effect on the performance of the underlying PSTN. The PSTN was designed for three-minute calls, three times per day, on average. With the Internet, calls are 10 times longer (or more). To support 10,000 voice sessions on the PSTN during peak time, 1,000 trunk groups are required for 99.9 percent availability. In contrast, to support 10,000 Internet sessions on the PSTN during the same time interval would require 5,000 trunks. 5,000 trunks require 4,000 more ports and transport facilities to maintain the same level of service performance, and the additional cost to the PSTN is significant.

Internet Call Diversion

The volume of Internet traffic and traffic characteristics leads to several points of congestion on the PSTN. A typical Internet configuration involves an Internet Service Provider (ISP) who connects at a Point Of Presence (POP) which is a carrier end-office or access tandem. As a result, Internet traffic generated throughout the Metropolitan Service Area (MSA) for this ISP is routed to this POP. Due to the mismatch between Internet traffic characteristics and the engineering of the PSTN serious congestion can occur on the PSTN due to Internet calling. First, blocking/congestion can occur on the primary rate interface (PRI) link between the terminating SSP and the ISP. This also can creates serious congestion on the link between the tandem switch network and the terminating SSP. This results in deployment of additional transport facilities and switch ports.

Fortunately, there are services to make the converged Internet and PSTN operate more efficiently. Networks can install an Internet Call Diversion (ICD) device. Internet Call Diversion off loads the Internet traffic off the PSTN allowing Internet calls to be directly connected to an a enhanced router, off loading the transport facilities and SSP. The device has a SS7 interface that sends Internet traffic directly to the data network. This process frees the trunk groups that were congested between the SSPs and tandem switch, and the trunks between the tandem switch and end SSP. ICD service is an example how the new IP-data world can be merged with the PSTN in a smooth way.

Integrating with Legacy Systems

IP telephony networks must be able to integrate with the legacy PSTN for seamless interoperability. In a PSTN-IP telephony integrated scenario, the existing 160 million wireline telephones and 75 million wireless telephones. The IP equivalent SSP called a softswitch is being introduced to carrier networks. A softswitch is a highly distributed SSP based upon open systems standards. As such, interoperability with the PSTN requires several softswitch interfaces to the PSTN. One of these softswitch interfaces is to the media gateway, which is a router that converts voice signals on the DS-0 trunk from the SSP into IP packets to the IP network. It compresses the 64K voice information into 16K blocks, for example, attaches an IP header with address information, and sends the packets over the IP network. The traffic terminates at another media gateway, which performs the reverse process on the blocks. It converts the packets back into PSTN DS-0 equivalent voice messages. This all happens in the transport layer.

The Control Layer

PSTN and IP networks need to interoperate at the control layer as well. The Media Gateway Controller (MGC) is the call-control entity associated with IP Telephony and softswitches. The MGC and media gateway perform the job that the SSP does in the PSTN. The gatekeeper does authentication, authorization, and address (AAA) mapping between the PSTN and IP network. On the PSTN side, people still are dialing PSTN telephone numbers. However, the IP network is routing based upon IP addresses, so there must be an address conversion between the two spaces. The gatekeeper does that as well.

A signaling gateway (SG) does the conversion between SS7-IP packets and the SS7 network. Interoperation between the two requires ISDN User Part (ISUP) trunk signaling information. When a call is made on the PSTN, the SSP (service switching point) sends an initial address message to the signal transfer point (STP). That message is communicated to the MGC via the SG. The MGC signals back to the SSP with an acknowledgement message, just as the SSP does today. In this way, companies can deploy networks based upon IP transport technology, but everything still will look like a normal SSP switch to the PSTN.

Database Access

IP network providers, whether local or interexchange, will need access to database services. Through the SS7 network, they can access LIDB (Line Information DataBase), Customer NAMe (CNAM), Tollfree 800, LNP (Local Number Portability) and other SS7 database services. For example, when the MGC requires access to calling-name information, it will send a message through the SG and SS7 network to the CNM database to get access to the name information. The CNAM information will be sent back to the SG via TCAP (Transaction CAPability) over MTP (message Transfer Protocol), and then to the MGC via TCAP over IP. The MGC will then signal to the MG with the name of the calling party who will then transmit the name information over the IP network and finally to the voice set. Other SS7 database services work likewise.

Intelligent Networks and IP Networks

VoIP's most significant impact is the true separation of call control from transport. This evolution began with the first introduction of SCP technologies in the 1980s and continues in to the new millennium at a record pace based upon the highly distributed open architecture associated with IP. Many argue that with IP (especially the Internet), intelligence will migrate to the edge, or even out of the network. Nothing could be farther from the truth. While it is clear that intelligence in the customer premises environment has increased significantly over the last two decades, intelligence in the network has increased in proportion. This trend will continue with intelligence growing in all areas including the network as well as the customer premises. Things will only get smarter.