Voice over IP (VoIP) for Enterprise Networks: Performance Implications & Traffic Models

P.M. Fiorini
BMC Software
pierre_fiorini@bmc.com

ABSTRACT

Voice over Internet Protocol (VoIP) is a rapidly emerging technology. Currently, the Internet is being wired to support voice and fax traffic. The widespread interest in VoIP is not necessarily the ability of IP to carry voice traffic but the ability to carry voice and fax traffic over data networks. The implications of this are far-reaching. That is, the Internet, for the most part, will eventually replace the PSTN (Public Switched Telephone Network). Furthermore, VoIP will play a major role in e-commerce applications such as IP-based call centers. More importantly, by running intra-company inter-site voice/fax over its IP data network, a company can expect to reduce telephony[1] costs by 70%-80%. In this paper, we discuss VoIP technology and its implications. In addition, we present traffic models to explore the performance implications of VoIP upon enterprise networks.

Introduction

Voice over Internet Protocol (VoIP) is the transmission of voice and fax traffic in packets over IP-based networks. There are many terms commonly used to describe this technology such as Internet Telephony, IP Telephony, etc. Interestingly, IP Telephony is viewed by some as a simply a means to place "free" telephone calls using the Internet; however, it is much more than that. In this paper, we discuss the importance of modeling VoIP for enterprise networks. We discuss how VoIP can be deployed and its increasing role in enterprise networks. Performance models that characterize VoIP traffic are presented and discussed.

The Public Switched Telephone Network

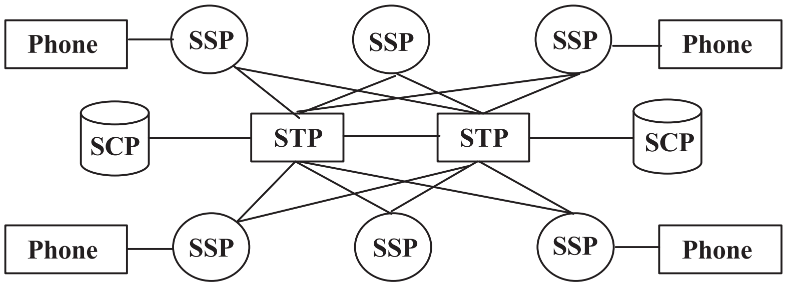
The Public Switched Telephone Network (PSTN) has become an "Intelligent Network" (IN). This means that the network has the capability to use real-time database information to route telephone calls. Using information retrieved from data stores, the following lists some of the telephony services that can be performed:

- *Toll-free calling* - Facilitates toll-free telephone services and proper billing.
- *Wireless roaming* - Enables wireless telephony via a series of networks.
- *Calling Cards* – Enables telephony services and proper billing.
- *Local portability* – Allows telephone users to change local carriers.
- *Call Waiting* – Notifies a calling party that another party is calling when the calling party's line is utilized.
- *Caller ID* – Enables a caller to be identified.
- *Pagers* – Supports the abilities for callers to page subscribers of this service

IN services are controlled by the Signaling Number Seven (SS7) protocol in the PSTN. SS7 is a layered protocol and its functionality is encapsulated in its software layers. The ISDN user part (ISUP) layer is

[1] Telephony, be it IP-based or conventional, can be defined as the science of transmitting voice, data, video, or image signals over physical or wireless communication networks.

Figure 1. An Intelligent Network Architecture



used for setting up and tearing down phone calls while the transaction control application part (TCAP) layer is used for exchanging real-time database information. Consequently, the ability to support IN services in the PSTN necessitates both ISUP and TCAP support.

Figure 1 illustrates the basic architecture of an IN. Signal switching points (SSPs) are telephone switches that initiate and terminate SS7 messages and signal transfer points (STPs) are devices that route SS7 messages within the network. Service control points (SCPs) are database servers that provide real-time data information for IN services. The PSTN is known for its reliability and quality of service (QoS). Voice and data traffic is carried on dedicated connections (switched circuits) and the entire bandwidth is available during a call. As the utilization of the network increases, it is more likely that users will experience busy signals; however, performance will not degrade since bandwidth for a call is always guaranteed. This fundamental aspect of the PSTN is what distinguishes it from IP networks.

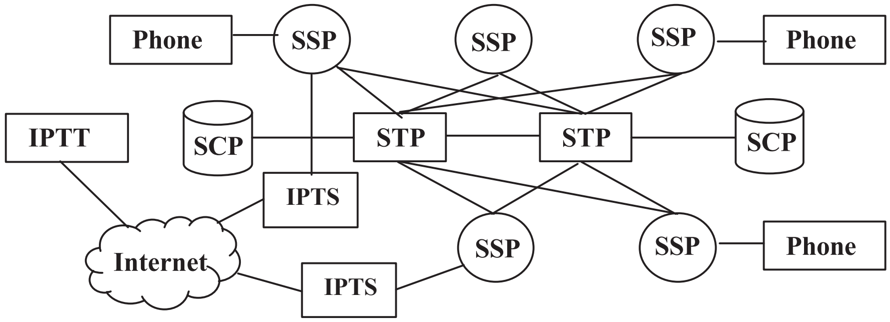
The architecture in the PSTN is very reliable since the SS7 protocol provides functionality in the event of link and node failures. In these cases, SS7 chooses alternate routes and/or facilitates message re-transmissions to make sure voice and data reach their destinations.

Voice over IP

A result of the tremendous growth of the World Wide Web (WWW) the Internet Protocol (IP) has become the de facto standard for data networking [BLAC00]. Unlike the circuit-switched technology of the PSTN, IP networks are packet switched and network bandwidth is shared by all users. A consequence of this is that when the network gets more utilized, it is more likely to experience performance degradation. Although PSTN and IP networks are fundamentally different in terms of their architectures, it is possible for the networks to be integrated; that is, exchanging voice and data traffic. Figure 2 depicts one topology to achieve this integration.

To see how this works suppose a subscriber connected to an SSP places a long-distance call though the PSTN. Voice traffic is routed though an intermediary toll SSP and there can be many hops involved when routing a call. The SS7 protocol is utilized to reserve voice trunks from one hop to the next and to set up the phone call. VoIP offers an alternative approach. A call is placed to an IP Telephony switch that digitizes and compresses the voice traffic into data packets and then sends this information across the IP network. SSPs on both ends of the call communicate with their respective IP Telephony switches (IPTS) using ISDN connections that include both voice and signaling information. Calls can be initiated or terminated on either the PSTN or the IP network. On IP networks, users send and receive voice using an IP Telephony

Figure 2. An Example of Integrating an IP network with the PSTN.



terminal (IPTT), which is a computer (i.e. usually a PC) that runs telephony software.

On an IP network, VoIP devices communicate using either proprietary protocols or one of the emerging standards such as H.323 or MGCP (Media Gateway Control Protocol). Because of the lack of fully defined standards, initial VoIP products mandated gateways from the same vendor exist on both sides of the IP network. Recently, however, examples of multivendor interoperability (based on H.323) have appeared. Presently, VoIP products are limited in their functionality since, for the most part, basic call setup and call tear down is possible; however, IN services are only starting to appear (e.g. call waiting). In any event, it is only a matter of time until most or all IN services are available via VoIP products.

4. Applications of VoIP Technology

Currently, the primary application of VoIP has been to transport voice and fax calls over an IP network to save on long-distance charges. Other applications are beginning to emerge and incorporate IP voice and fax into telephony applications for enhanced network services (e.g. INs). These two objectives are the primary focus in main VoIP market applications. Some applications of VoIP are listed as follows [MICO99]:

1. *Toll-Free Corporate Telephony Services*– Toll-free intra-company voice and fax between corporate locations.
2. *Fax over IP* – Toll-free or reduced rate fax-machine fax between any two locations.
3. *PC-phone to PC-phone* – Toll-free voice between any two PC's on the Internet.
4. *IP-Based Public Phone Service* – New public phone services, at reduced rates (especially international calls), where voice is sent over the Internet or over new public IP networks.
5. *IP-Based Call-Centers* – These applications allow PC users at home with their browsers to click on a telephone icon in a catalog at a customer service Web page that enables he/she to an agent via the PC as a phone.
6. *IP Line Doubler* – A PC user at home or in a hotel with just one connection to the Internet could subscribe to a new service that facilitates a single phone line to carry one or more phone calls in addition to data.

Note that applications 1, 4, and 6 rely on an emerging communications technology known as a Voice/Fax IP gateway. A VoIP gateway converts analog voice and fax into IP packets. Application 3 does not use an IP gateway since PCs perform the gateway functionality. In other words, the PC has a sound card, speakers and microphones, or a phone card with a telephone. Thus, the PC performs voice packetizing. PC-fax is just data so only a fax-machine needs a gateway. For the most part, apart from gateways, most telephony technology is software driven [MICO99].

Since toll-free corporate telephone services has been the fastest to mature and be adapated, we will discuss this area further. The popularity of the IP protocol in corporate data networks has increased tremendously recently. Networking managers have adapted IP as the protocol foundation for their networks since most any type of corporate network can be built upon IP. The reason is that it scales to millions of nodes and users. More importantly, by running intra-company, inter-site voice/fax over its IP data network, a company can expect to reduce telephony costs by 70%-80% [MICO99].

5. VoIP Voice Quality

There are several technologies to ensure good voice quality. They will be described in turn by the following:

- *Echo Cancellation* – When a telephone cable connects to, say, a Private Branch Exchange (PBX) interface or a telecommunications company central office (CO), a special electrical circuit called a hybrid is used for the signal conversion. Although hybrid circuits are efficient in their conversion ability, a small percentage of telephony energy is not converted but instead reflected back to the caller. This is referred to as echo. If the caller is near the PBX or CO switch, the echo it is not discernible. However, this may not always be the case. To prevent this gateway manufacturers include special code in Digital Signal Processors (DSPs) that try to cancel the echo.
- *Network Delay & Jitter* – IP packet delay and network jitter is responsible for reduced voice quality on VoIP networks. Network delay is the average length of time for a packet to travel in a network. Network jitter describes the variability in arrival time of a packet. When a voice packet is delayed and does not arrive at the destination time to fit into the voice stream going out of the destination gateway, it is discarded, and the previous packet is replayed. If this happens too often, the listener will perceive reduced voice quality.
- *VoIP Packet Prioritization* – VoIP works well over a corporate IP network because the network can be optimized for low VoIP packet jitter and low delay. This results from corporate routers prioritizing voice packets. The router is instructed to look for voice IP packets and put them ahead of any data packets waiting in the router's transmit queue. This way, a stream of outgoing packets will not add to the variability of the arrival time of voice packets. The router is instructed to prioritize voice/fax IP packets either by the network administrator explicitly programming the router to look for the gateway's UDP port number, or by using a prioritization protocol called RSVP. When the gateway determines it needs to receive a voice/fax call, it establishes an RSVP session with the router using the LAN to pass information. The gateway instructs the router to prioritize such packets for the duration of the call.
- *IP Packet Segmentation* – This method is used to ensure that large data packets do not delay voice packets from exiting the router in a timely manner. This is achieved by programming the router to segment all outbound data packets according the speed of the WAN access link.

6. VoIP Standards

H.323 is a set of standards defining real-time multimedia communications and conferencing over packet-based networks. It is a standard choice for VoIP. Version 1 of the H.323 protocol was first accepted by the ITU-T in 1996; however, the IETF is still debating a few alternative standards. For the most part, the market place has already adapted the H.323 standard. Version 2 of H.323 was adapted in 1998 and basically comprises four components and are discussed below:

- *Terminals* – Terminals provide real-time communications. They must support voice communications. The most common H.323 terminals are telephony applications (e.g. Microsoft's NetMeeting? that runs on a PC).
- *Gateways* – H.323 gateways provide services to H.323 clients so that they can communicate with non-H.323 terminals and telephones on the circuit-switched network. The gateway must provide translation between different transmission formats, communication procedures, and audio codecs.
- *Gatekeepers* – Gatekeepers provide call control services for H.323 end-points such as address translation and bandwidth management. Gatekeepers are optional. If they are present in a network, however, endpoints must use their services. The H.323 standards define mandatory services that a gatekeeper must provide.
- *Multipoint Control Units* (MCUs) – They provide support for conferences of three or more endpoints. An MCU manages conference resources and negotiations between endpoints for the purposes of determining the audio or video codec to use. Sometimes, it also handles the media stream.

Unfortunately, at the present time, most VoIP products are not Version 2 compliant and most implementations are missing important aspects of the H.323 standard [WILL99]. For instance, security is available; that is, authentication, encryption, and integrity. However, products do not need to offer security to be H.323 compliant. Without security, it is easy via packet analyzers to eavesdrop on conversations wherever packets pass. Some protocol analyzers can be configured to detect VoIP streams and play them back as audio files.

As mentioned previously, the gatekeeper function is optional. Essentially, the gatekeeper provides mechanisms to prevent the system being overloaded with voice (and video) calls. It also provides call management, signaling, and overall system control. The VoIP gatekeeper is significant for real system management. Without the gateway, instead of receiving a busy signal, endpoints submit packets to the network without guarantees of anything. As a result, audio and video packets can overflow network devices. In addition to the H.323 set of standards, the Internet Engineering Task Force (IETF) is developing specifications that will facilitate IP networks to handle voice calls as reliably as the PSTN in the future. The IETF is considering whether to preserve the PSTN's native signaling protocols, including SS7, or create new IP control protocols. One issue is how to send SS7 signals over IP networks. The IETF's Media Gateway Control Working Group is operating under the assumption that SS7 networks will eventually evolve into IP networks. MGCP defines how media devices controls packets and determines how calls are manipulated and forwarded and gives the network device the ability to determine if a call should be sent over a company's intranet, over the Internet, or over the the PSTN. Service providers are not only depending on the IETF for VoIP standards; they are looking to the ITU and vendor driven forums to address tariffing and voice traffic exchange specifications that promise interoperability.

7. Advantages of VoIP

The advantages of VoIP technology can be divided into the following categories [TECH99]:

- *Cost Reduction* – Fixed rate pricing is available with the Internet and can result in savings for both voice and fax transactions. Lower prices are based on avoiding telephony charges rather than being a reduction in resource costs. Also, sharing equipment and operations

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