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Technology Backgrounder on Telephony & VoIP Basics

By Larry Hettick and Steven Taylor

Managers of small and medium businesses (SMBs) have, by necessity, learned about data networking and desktop computing because few businesses can operate without computers and access to data networks like the Internet. Medium-sized businesses have probably even hired or outsourced an information technology (IT) staff to help set up their computers and data networks.

While many of these small and medium businesses (SMBs) have tackled the management of their own basic data communications systems, most rely on "the phone company" to take care of their voice communications. However, as the worlds of voice and data communications converge, bringing the two networks together dictates that SMB managers now understand some basics about both networks and how they operate. And if a SMB wants to enjoy the business benefits that a Voice over IP (VoIP) solution brings to the workplace, understanding both phone systems and VoIP is especially important. This backgrounder will provide some basics about telephony and VoIP.

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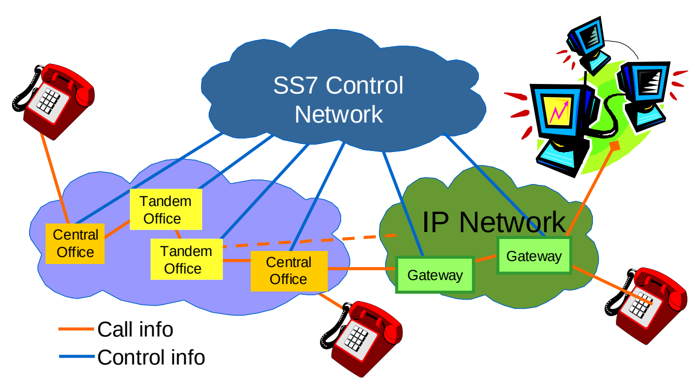
The Public Switched Telephone Network (PSTN)

The Public Switched Telephone Network (PSTN) consists of a series of central offices (COs) on the edge of the network, tandem offices in the network core, a distribution network connecting the COs and tandem offices, and a signaling system to control these network elements. This architecture is shown in the left half of Figure 1.

Figure 1: A Converged PSTN and VoIP Network



A combination of analog and digital transmission technologies are used to transmit conversations within this network. Analog transmissions use continuously variable electrical impulses representing the conversation. Digital transmission systems convert analog signals into discrete electrical impulses represented by 0s and 1s for transmitting information.

While either analog or digital transmission techniques may be used in the "local loop" from the premise to the local CO, essentially all transmissions between COs are digital to assure better transmission quality and efficiency. Digitizing allows for another important efficiency in the PSTN: when analog sound waves are sampled and converted into digital signals, voice and data may be sent over the same digital transmission facilities.

The most common modulation techniques are pulse code modulation (PCM) and adaptive differential pulse code modulation (ADPCM). When industry-standard PCM is used, each analog conversation is converted into a digital bit stream of 64,000 bits per second (64 kbps). ADPCM is a more technically advanced method for digitizing voice. It reduces PCM's 64 kbps to 40, 32, 24, or even 16 kbps. For technologies like VoIP, additional digital voice encoding schemes permit even lower bandwidth to be used per call while maintaining excellent call quality.

64 kbps PCM has been used in the PSTN for decades. Because of this use, 64 kbps increments (sometimes referred to as DS-0s) are the fundamental building blocks of digital transmission systems. In North America, the transmission system, called "T-carrier," bundles 24 DS-0s per T-1 (also called DS-1), for a total capacity of 1,544,000 bits per second (1.544 Mbps). 28 T-1s are bundled into a T-3 (also called DS-3) with a total capacity of 45 Mbps. Multiple T-3 channels can be bundled into an optical channel (OC) at even higher speeds. Some countries use a similar system that combines 30 DS-0s into a single bundle called an E-1, and multiple E-1s are combined into E-3s. Each T-1, E-1, T-3, or E-3 transmission link can be channelized into 64 kbps streams (or multiples of 64 kbps streams). Alternatively, the entire transmission bandwidth may be used as a single entity, sometimes called "unchannelized" or "clear channel."

In the pre-VoIP world, all digital phone calls were sent over a time division multiplexed (TDM) link, and TDM is still the default multiplexing used technique today. TDM transmissions enjoy a built-in quality of service because each transmission receives a dedicated time slot for the user's data or voice. That time slot cannot be over-run by another user's voice or data. Unfortunately, if the dedicated time slot is not used by the voice or data, the slot is wasted since it cannot be shared with another user.

To signal calls between users, the PSTN employs a packet-based network called signaling system 7 (SS7) or common channel signaling system 7 (CCSS7) to determine the best call route, connect the callers, and provide call control. The left half of Figure 1 shows the traditional PSTN architecture. The right half of the diagram shows that SS7 must be integrated for call control when introducing voice over IP (VoIP); VoIP will be discussed in a later section.

Central offices provide basic features like dial tone, dialing (processing call set-up requests), and incoming call notification (electrical "ringing" current). COs also provide more sophisticated feature sets like Centrex. Working together, the SS7 network and the CO provide more advanced features like caller ID and call forwarding.

Private voice network systems like private branch exchanges (PBXes) and key systems also work with the PSTN to provide a hybrid public/private network. A PBX allows multiple users at a site to share incoming and outgoing PSTN trunk lines, so a site with a PBX does not have to dedicate individual PSTN lines to each user or phone station. The PBX also provides abbreviated number dialing to PBX extensions so internal callers can bypass the PSTN and save additional expense. In addition, the PBX offers features like call transfer, call pickup, and auto attendant. A key system is, in simplest terms, a smaller version of the PBX. Centrex, offered by local service providers, has features similar to a PBX except that each phone station is linked back to the local Central Office so users can't share incoming and outgoing trunk lines.

PBX and key systems also integrate with other value added applications and features like voice mail, unified messaging, and interactive voice response (IVR) systems. Unified messaging systems allow common telephony user interface (TUI) and graphical user interfaces (GUI) to voice mail, email, and fax. IVR systems interact with the caller to help direct calls based on the caller's selection of audio prompts. Additionally, dedicated circuits, often called "tie lines," can be used to connect a company's PBXes at various locations, bypassing the PSTN.

Voice over IP

Voice over IP (VoIP) Technology is becoming increasingly important as businesses large and small look at how they will communicate. Using VoIP, the business can save money on long distance[1] and on moves/adds/changes to their voice network, while offering cost-effective integrated communications tools like unified messaging and IVR. While VoIP does add a layer of complexity to the communications network, the cost-benefit will provide a near term and long term return on investment.

Using IP to signal and transport voice brings several fundamental shifts to voice communications. In a TDM environment, unused bandwidth cannot be shared. Using packetized transmission for voice allows for greater efficiency by sharing bandwidth, thereby reducing cost. IP is the packet protocol of choice for voice because the overall volume of users' wide area network traffic is dominated by IP. Service providers and large businesses were quick to take advantage of this efficiency and became early VoIP adopters. For example, many service providers are now replacing their TDM tandem switches (shown in Figure 1) with VoIP.

In the legacy PSTN, voice network features are delivered to a user on a static pair of copper wires to a static local CO switch or PBX. VoIP allows the traditionally switched services to be delivered to a user anywhere the user is connected. Because of this, central office and PBX functions are provided in a distributed environment, and user proximity to the CO or PBX becomes irrelevant. Users can remotely connect to their company's PBX and enjoy the full range of PBX features without sitting at the same location as their PBX. IP-PBX user moves, adds, and changes can be automatically tracked by the IP-PBX at user log-in.

[1] *Note that long distance savings are realized when the company has multiple sites. Carriers have also taken advantage of IP to reduce long distance costs, contributing to the price reductions in long distance over the last several years*

Implementing VoIP

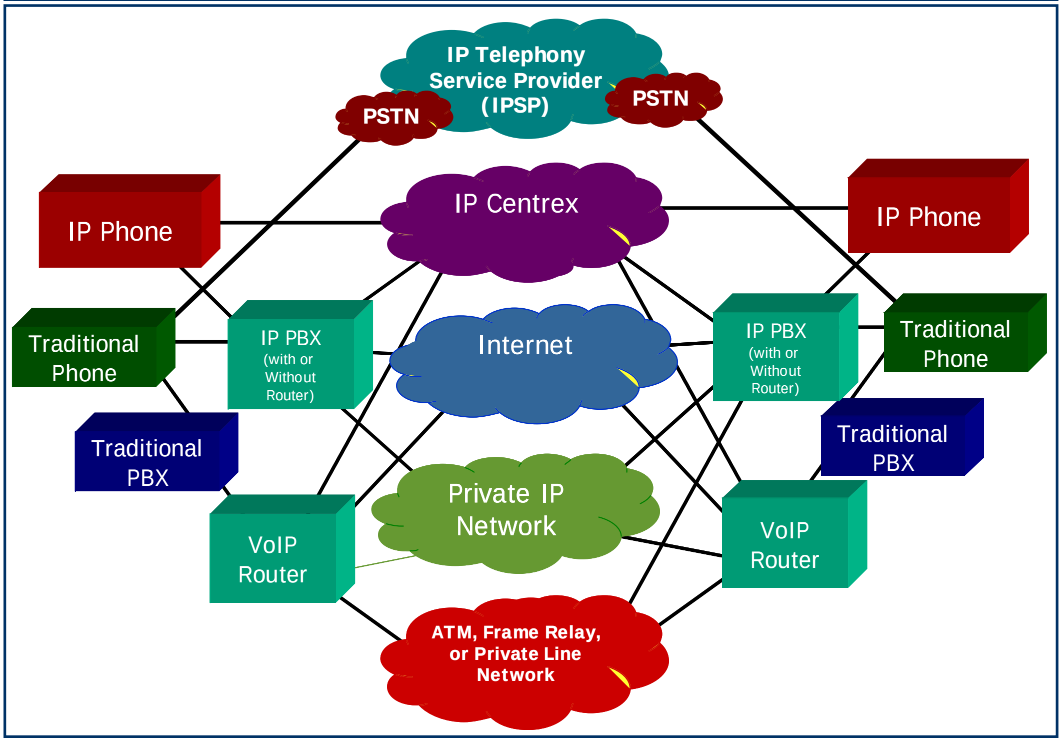
When deploying VoIP (as the name implies), voice will be carried on an Internet Protocol (IP) infrastructure. Therefore, IP devices like routers and IP protocols are an inherent part of any VoIP deployment. To deploy VoIP successfully, the business manager must understand and plan for four factors: voice quality of service, network security, network manageability, and graceful migration.

While voice quality of service (QoS) can meet or exceed the voice quality of the PSTN or PBX, QoS doesn't come automatically. Pre-deployment strategies **must** include a network assessment to make sure the existing IP network can accommodate the added traffic that voice will put onto the IP network. Further, ongoing QoS management techniques and protocols must be supported. Managers should work with their suppliers or service providers to complete the pre-deployment assessment and to implement ongoing QoS.

Voice conversations over an IP network can be as secure as or more secure than voice over the PSTN. However, network security also doesn't come automatically. Standard data networking security techniques like the use of firewalls and user authentication must be included in a VoIP deployment. Voice encryption is optional but sometimes recommended.

Network manageability should be matched to the skills available inside the company. While many features on a VoIP network are "plug and play" after initial setup, businesses must find a comfortable approach to managing the network. Small and medium businesses can rarely afford their own network operations center, so they should select a supplier or service provider who will provide simple, intuitive network management tools that meet business needs.

Figure 2: VoIP Network Architecture



Finally, pre-deployment planning should include a migration strategy. Any VoIP system must interwork with the PSTN; it must also be complementary to private phone system components like PBXes and traditional phones. A phased migration will provide the best possible solution for interconnection and maximize existing capital asset investments.

Implementing a VoIP infrastructure presents the business manager with many choices. As shown in Figure 2, a variety of configurations are possible with VoIP. The most common deployments are based on a VoIP-enabled router or use an IP-PBX or IP-enabled PBX. Some deployments use a combination of approaches. Also shown is that connectivity for VoIP is available using a variety of public network services.

The **VoIP gateway** function transforms existing voice calls and signaling to control those calls into Internet Protocol (IP) based transmission and signaling. By installing equipment that acts as a gateway, a business can connect simultaneously to an IP network using data network connections and to the PSTN using traditional telephony connections. On-net calls are made using VoIP, and off-net calls can be directed to the PSTN.

VoIP enabled routers add a gateway function to the router. Physical connections compatible with existing PSTN equipment are accepted by the VoIP enabled router, as are Ethernet connections using standard data networking technology. VoIP enabled routers allow companies to upgrade their router to include the gateway and voice-specific features; however, the VoIP enabled router usually does not include all the features and applications support found in a PBX or key system. These devices are typically used to emulate tie lines in order to save on long-distance toll charges.

IP-PBX and IP-enabled PBX deployments are similar in that they start with PBX features and include a gateway function. A pure IP-PBX is purpose-built to include IP, whereas an IP-enabled PBX adds IP functionality to existing PBX equipment and features. Many IP-PBXes use industry-standard hardware and operating systems, with IP-PBX-specific software. IP-enabling an existing PBX will require additional software and may require additional hardware.

VoIP Protocols

The options discussed above use one of two IP-based protocols for VoIP signaling and call control; these are H.323 or Session Initiation Protocol (SIP). International Telecommunication Union-Telecommunication Standardization Sector (ITU-T) standard H.323 is actually an "umbrella" of standards for packet-based multimedia communications systems. This family of standards defines the different devices that make up a multimedia system, including techniques to signal, control, and transmit voice, video, and data onto a converged network. Older than SIP, H.323 has been widely deployed in early implementation of VoIP and, as an ITU-T standard, provides for a rich set of telephony features.

SIP is a newer standard that differs slightly from H.323 in that it offers presence information and can accommodate multiple, concurrent multimedia sessions. Presence information indicates that a user is (or is not) available to accept a call or data session, similar to the buddy list icon seen on an instant message screen. Concurrence allows a user to switch between or maintain simultaneous phone calls, web chat sessions, and document collaboration sessions. SIP is enjoying broader acceptance and growing support because of these two added features.

Next Step: Unified Communications, Applications Convergence

From a network efficiency perspective, VoIP brings multiple advantages over a traditional TDM voice infrastructure. However, an even greater benefit can be had from a VoIP system. For example, unified messaging brings together user control of voice mail, email, and fax to a single graphical screen on the user's desktop. Unified messaging is supported by both TDM and IP-based communications systems. However, IP-based systems can also easily add instant messaging (with presence), click to call, click to conference, web chat, and document collaboration to the same user interface. Using SIP, these real time communications sessions can be multimedia and concurrent. Adding real time communications to unified messaging converts a unified messaging system into a unified communications system.

More than just cool technology, each of these features can add to user productivity and improve communications between employees and customers. As business becomes more dependent on data-centric communications with the customer, these features become increasingly important to business operations.

Moving beyond person-to-person communications, VoIP and web-based protocols can be used to add data integration services. For example, a customer's profile or account can be added as a "screen pop" to the user interface when the customer calls for an update. VoIP and web-based protocols provide the baseline to add a screen pop in a matter of hours; a similar integration with a legacy phone system would take a skilled programmer days, weeks, or months to accomplish.

Summary and Conclusions

Telephone networks and voice communications are more than "just another application" that can ride on a data network. Bringing together all the PSTN features and providing the needed interworking between voice and data networks is no simple task. However, the rewards to businesses that properly migrate to a VoIP-based telephone system are many. By deploying VoIP, the company will reduce operational and network costs, bring added productivity tools to the employees, and lay the foundation for improved customer service.

Additional Resources

This backgrounder is one of a series targeted to help the small and medium business improve their skills in telecommunications and data communications. Additional backgrounders, papers, presentations, and detailed technical and business information are available from the backgrounder's sponsor and from www.webtorials.com.

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Steven Taylor, consultant and broadband packet evangelist, is President of Distributed Networking Associates and Publisher of Webtorials. An independent consultant, planner, author, and teacher since 1984, Mr. Taylor is frequently quoted in the trade press and is one of the industry's most published authors and lecturers on high bandwidth networking techniques. He has served as a Contributing Editor for Data Communications magazine, publishes articles in both Business Communications Review and Network World, and co-authors two newsletters - Convergence and Wide Area Networking - distributed by Network World Fusion.

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A word from the sponsor
Nortel Networks

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