**Chapter 7**

**Voice Network Design Considerations**

**Analog and Digital Signaling**

The human voice generates sound waves; a telephone converts the sound waves into analog signals. However, analog transmission is not particularly efficient. Analog signals must be amplified (improved) when they become weak from transmission loss as they travel. However, amplification of analog signals also amplifies noise.

The PSTN is a collection of interconnected voice-oriented public telephone networks, both commercial and government-owned. The PSTN today consists almost entirely of digital technology, except for the final link from the central (local) telephone office to the user.

To obtain clear voice connections, the PSTN switches convert analog speech to a digital format and send it over the digital network. At the other end of the connection, the digital signal is converted back to analog and to the normal sound waves that the ear picks up. Digital signals are more safe to noise, and the digital network does not induce (bring) any additional noise when amplifying signals.

Signals in digital networks are transmitted over great distances and are coded, regenerated, and decoded without degradation of quality. Repeaters amplify the signal, restore it to its original condition, and send this clean signal to the next network destination.

#### **Differences between a PBX and a PSTN Switch**

As shown in Table 7-1, PBXs and PSTN switches share many similarities, but they also have many differences.

|  |  |
| --- | --- |
| PBX | PSTN Switch |
| Used in the private sector | Used in the public sector |
| Scales to thousands of phones | Scales to hundreds of thousands of phones |
| Mostly digital | Mostly digital |
| Uses 64-kbps circuits | Uses 64-kbps circuits |
| Uses proprietary(patented or brand ) protocols to control telephones | Uses open-standard protocols between switches and telephones |
| Interconnects remote branch subsystems and telephones | Interconnects with other PSTN switches, PBXs, and telephones |

Table 7-1: PBX and PSTN Switch Comparison

Both the PBX and PSTN switch systems use 64-kbps circuits; however, the scale is very different. A PSTN switch can support hundreds of thousands of telephones, whereas a PBX can support only several thousand.

A PSTN switch’s primary task is to provide residential telephony. A PBX supports user telephones within a company.

#### **PBX Features**

A *PBX* is a business telephone system that provides business features such as call hold, call transfer, call forward, follow-me, Call Park, conference calls, music on hold, call history, and voice mail. Most of these features are not available in traditional PSTN switches.

A PBX switch often connects to the PSTN through one or more T1 or E1 digital circuits. A PBX supports end-to-end digital transmission, employs PCM switching technology, and **supports both analog and digital proprietary telephones.**

A T1 trunk(box or case) can carry **24 fixed 64-kbps** channels for either voice or data, using PCM signals and TDM, plus additional bits for framing, resulting in an aggregate carrying capacity of 1**.544 megabits** per second (Mbps). **T1 lines** originally used copper wire but now also include optical and wireless media.

**In Europe**, the trunk used to carry a digital transmission is an **E1. An E1 trunk** can carry up to **31 fixed 64-kbps** channels for data and signaling, with another 64-kbps channel reserved for framing, giving an aggregate carrying capacity of **2.048 Mbps**.

PBXs support end-to-end digital transmission, use PCM switching technology, and support both analog and digital proprietary telephones. A local ***PBX provides several advantages*** for an enterprise:

* Local calls between telephones within the PBX or group of PBXs are free of charge.
* Most PBX telephone system users **do not call externally**, through the **T1 or E1 circuits, at the same time**. Therefore, companies with a PBX only need the number of external lines to the PSTN to equal the maximum possible number of simultaneous calls, resulting in PSTN cost savings.
* When adding a new user, changing a voice feature, or moving a user to a different location, there is no need to contact the PSTN carrier; the local administrator can reconfigure the PBX.

##### **PSTN Switches**

The PSTN appears to be a single large network with telephone lines connected. In reality, the **PSTN is composed** of circuits, switches, signaling devices, and telephones. Many different companies own and operate different systems within the PSTN.

##### **PSTN Features**

A PSTN switch’s primary role is to connect the calling and called parties. If the two parties are physically connected to the same PSTN switch, the call remains local; otherwise, the PSTN switch forwards the call to the destination switch that owns the called party.

PSTN switches interconnect business PBXs and public and private telephones. Large PSTN switches are located at COs, which provide circuits throughout the telephony network. PSTN switches are deployed in hierarchies to provide resiliency and redundancy to the PSTN network and avoid a single point of failure.

PSTN signaling traditionally supported only basic features such as caller ID and direct inward dialing. Modern PSTN switches now support, on a fee basis, many traditional PBX services, including conferencing, forwarding, call holding, and voice mail.

##### **PSTN Services**

Modern PSTN service providers offer competitive services to differentiate themselves and generate additional revenue. These PSTN services include the following:

* **Centrex:** *Centrex* is a set of specialized business solutions (primarily, but not exclusively, for voice service) in which the service provider owns and operates the equipment that provides both call control and service logic functions; therefore, the equipment is located on the service provider’s premises.
* **Voice virtual private networks (VPN):** *Voice VPNs* interconnect corporate voice traffic among multiple locations over the PSTN. PBXs are connected to the PSTN instead of directly over tie trunks. The PSTN service provider provides call routing among locations, and all PBX features are carried transparently across the PSTN.
* **Voice mail:** *Voice mail* is an optional service that lets PSTN customers divert their incoming PSTN calls to a voice mailbox when they are unable to answer their telephones, such as when the line is busy or they are unavailable. Alternatively, all calls can be diverted to the voice mailbox.
* **Call center:** A *call center* is a place of doing business by telephone, combined with a centralized database that uses an automatic call distribution (ACD) system. Call centers require live agents to accept and handle calls.
* **Interactive voice response:** *Interactive voice response (IVR) systems* allow callers to exchange information over the telephone without an intermediary live agent. The caller and the IVR system interact using a combination of spoken messages and dual-tone multi-frequency (DTMF) touch-tone telephone pad buttons.

### **Local Loops, Trunks, and Inter switch Communications**

The telephone infrastructure starts with a simple pair of copper wires running to the end user’s home or business. This physical cabling is known as a local loop or telephone line; the local loop physically connects the home telephone to the CO PSTN switch. Similarly, the connection between an enterprise PBX and its telephones is called the station line.

**A *trunk*** is a communication path between two telephony systems. Available trunk types, include the following:

* **Tie trunk:** Connects enterprise PBXs without connecting to the PSTN (in other words, not connecting to a phone company’s CO). Tie trunks are used, for example, to connect PBXs in different cities so that the enterprise can use the PBX rather than the PSTN for intercity calls between offices and, as a result, save on long-distance toll charges. A connection to the PSTN via a CO trunk is still required for off-net calls (to non-office numbers).
* **CO trunk:** Connects CO switches to enterprise PBXs. Enterprises connect their PBXs to the PSTN with PBX-to-CO trunks. The telephone service provider is responsible for running CO-to-PBX trunks between its CO and enterprise PBXs; from a service provider point of view, these are *lines* or *business lines*.
* **PSTN switch trunk:** Interconnects CO switches; also called *interoffice trunks*.

Another type of trunk, **foreign exchange** (FX) trunks, are analog interfaces used to interconnect a PBX to telephones, other PBXs, or to the PSTN. FX trunks save on long-distance toll calls; the dial tone from a different toll region is produced via the FX trunk at a reduced tariff.

**Two** types of FX trunk interfaces exist:

* **Foreign Exchange Office (FXO):** This interface emulates a telephone. It creates an analog connection to a PSTN CO or to a station interface on a PBX. The FXO interface sits on the PSTN or PBX end of the connection and plugs directly into the line side of the PSTN or PBX so that the PSTN or PBX thinks the FXO interface is a telephone. The FXO interface provides either pulse or DTMF digits for outbound dialing. The PBX or PSTN notifies the FXO of an incoming call by sending ringing voltage to the FXO. Likewise, the FXO answers a call by closing the loop to allow current flow. After current is flowing, the FXO interface transports the signal to the **Foreign Exchange Station (FXS).**
* **FXS:** This interface emulates a PBX. It connects directly to a standard telephone, fax machine, or similar device and supplies line power, ring voltage, and dial tone to the end device. An example of where an FXS is used to emulate a PBX is in locations where there are not physical lines for every telephone.

### **Telephony Signaling**

In a telephone system, a signaling mechanism is required for establishing and disconnecting telephone communications.

### **Telephony Signaling Types**

The following forms of signaling are used when a telephone call is placed via a PBX:

* Between the telephone and PBX
* Between the PBX and PSTN switch
* Between the PSTN switches
* Between two PBXs

At a high level, there are two signaling realms

* **Local-loop signaling:** Between a PSTN or PBX switch and a subscriber (telephone)
* **Trunk signaling:** Between PSTN switches, between a PSTN switch and a PBX, or between PBX switches.

Simple signaling examples include the ringing of the telephone, a dial tone, and a ring-back tone. Following are the three basic categories of signals commonly used in telephone networks:

* **Supervision signaling:** Typically characterized as on-hook, off-hook, and ringing, supervision signaling alerts the CO switch to the state of the telephone on each local loop. Supervision signaling is used, for example, to initiate a telephone call request on a line or trunk and to hold or release an established connection.
* **Address signaling:** Used to pass dialed digits (pulse or DTMF) to a PBX or PSTN switch. These dialed digits provide the switch with a connection path to another telephone or customer premises equipment.
* **Informational signaling:** Includes dial tone, busy tone, reorder tone, and tones indicating that a receiver is off-hook or that no such number exists, such as those used with call progress indicators.

## Integrating Voice Architectures

## Integrating data, voice, and video in a network enables vendors to introduce new features. The unified communications network model enables distributed call routing, control, and application functions based on industry standards. Enterprises can mix and match equipment from multiple vendors and geographically deploy these systems wherever they are needed.

One means of creating an integrated network is to replace the PBXs’ voice tie trunks with IP connections by connecting the PBXs to voice-enabled routers. The voice-enabled routers convert voice traffic to IP packets and direct them over IP data networks. This implementation is called *VoIP*.

IP telephony, a superset of VoIP, is another implementation. IP phones are used, and the phones themselves convert the voice into IP packets. A dedicated network server that runs specialized call processing software replaces the PBX; in Cisco networks, this is the Cisco Unified Communications Manager. IP phones are not connected with telephone cabling. Instead, they send all signals over standard Ethernet.

#### **Drivers for Integrating Voice and Data Networks**

Although a PSTN is effective for carrying voice signals, many business drivers are forcing the need for a new type of network for the following reasons:

* Data has overtaken voice as the primary traffic on many voice networks.
* Companies want to reduce WAN costs by migrating to integrated networks that can efficiently carry any type of data.
* The PSTN architecture was designed and built for voice and is not flexible enough to optimally carry data.
* The PSTN cannot create and deploy features quickly enough.
* Data, voice, and video cannot be integrated on the current PSTN structure.

IP telephony is cost-effective because of the reduced number of tie trunks and higher link efficiency, and because both voice and data networks use the same WAN infrastructure. It is much easier to manage a single network than two separate networks, because doing so requires fewer administrators, a simplified management infrastructure, and lower administrator training costs.

Whether or not either caller is talking, circuit-switched (classical voice) calls require a dedicated duplex 64-kbps dedicated circuit between the two telephones. During the call, no other party can use the 64-kbps connection, and the company cannot use it for any other purpose.

Packet-switched networking uses bandwidth only when it is required. This difference is an important benefit of packet-based voice networking.

On an IP network, voice servers and application servers can be located virtually anywhere. The rationale for enterprises to maintain voice servers, as with data application servers, is diminishing over time. As voice moves to IP networks (using the public Internet for inter-enterprise traffic and private intranets for intra-enterprise traffic), service providers might host voice and application servers.

### **Introduction to IP Telephony**

IP telephony refers to cost-effective communication services, including voice, fax, and voice-messaging applications, transported via the packet-switched IP network rather than the circuit-switched PSTN.

VoIP uses voice-enabled routers to convert voice into IP packets and route those packets between corresponding locations. Users do not often notice the implementation of VoIP in the network; they use their traditional phones, connected to a PBX. However, the PBX is not connected to the PSTN or to another PBX, but to a voice-enabled router that is an entry point to VoIP.

IP telephony replaces traditional phones with IP phones and uses the Cisco Unified Communications Manager, a server for call control and signaling, in place of PBXs. The IP phone itself performs voice-to-IP conversion, and voice-enabled routers are not required within the enterprise network. If connection to the PSTN is required, a voice-enabled router or other gateway must be added where calls are forwarded to the PSTN.

The basic steps for placing an IP telephone call include converting the analog voice signal into a digital format, and compressing and translating the digital signal into IP packets for transmission across the IP network. The process is reversed at the receiving end.

The IP telephony architecture, illustrated in Figure 8-18, includes four distinct components: infrastructure, call processing, applications, and client devices. These components are described as follows:

* **Infrastructure:** The infrastructure is based on data link layer and multilayer switches and voice-enabled routers that interconnect endpoints with the IP and PSTN network. Endpoints attach to the network using switched 10/100 Ethernet ports. Switches may include Power over Ethernet (PoE) ports that sense the presence of IP devices that require inline power, such as Cisco IP phones and wireless access points, and provide that power. Voice-enabled routers perform conversions between the circuit-switched PSTN and IP networks.
* **Call processing:** Cisco Unified Communications Manager is the software-based call-processing component of the Cisco enterprise IP telephony solution. Cisco Unified Communications Manager provides a scalable, distributable, and highly available enterprise IP telephony call processing solution and performs much like the PBX in a traditional telephone network, including providing call setup and processing functions.

The Cisco Unified Communications Manager can be installed on Cisco MCS 7800 Series server platforms and selected third-party servers.

* **Applications:** Applications provide additional features to the IP telephony infrastructure. Cisco Unity unified messaging (integrating e-mail and voice mail), Cisco Unified Meeting Place (multimedia conferencing), Cisco Unified IP IVR, and Cisco Unified Contact Center products (including intelligent contact routing, call treatment, network-to-desktop computer telephony integration, and multichannel automatic call distribution) are among the Cisco applications available for IP telephony. The open-source application layer allows third-party companies to develop software that interoperates with Cisco Unified Communications Manager.
* **Client devices:** Client devices are IP telephones and software applications that allow communication across the IP network. Cisco Unified Communications Manager centrally manages the IP telephones through Ethernet connections in the Building Access Layer switches.

#### **IP Telephony Design Goals**

Typical design goals of an IP telephony network are as follows:

* **End-to-end IP telephony:** Using end-to-end IP telephony between sites where IP connectivity is already established. IP telephony can be deployed as an overlaid service that runs on the existing infrastructure.
* **Widely usable IP telephony:**To make IP telephony widely usable, voice quality should be at the same level as in traditional telephony; this is known as *toll quality voice*.
* **Reduced long-distance costs:** Long-distance costs should be lower than with traditional telephony. This can be accomplished by using private IP networks, or possibly the public Internet, to route telephone calls.
* **Cost-effective:** Making IP telephony cost effective depends on using the existing WAN capacity more efficiently and the cost-of upgrading the existing IP network infrastructure to support IP telephony. In some cases, this goal can be accomplished by using the public Internet or private IP networks to route telephone calls.
* **High availability:** To provide high availability, redundant network components can be used and backup power can be provided to all network infrastructure components, including routers, switches, and IP phones.
* **Lower total cost of ownership:** IP telephony should offer lower total cost of ownership and greater flexibility than traditional telephony. Installation costs and operational costs for unified systems are lower than the costs to implement and operate two infrastructures.
* **Enable new applications on top of IP telephony via third-party software:** For example, an intelligent phone used for database information access as an alternative to a PC is likely to be easier to use and less costly to own, operate, and maintain.
* **Improved productivity:** IP telephony should improve the productivity of remote workers, agents, and stay-at-home staff by extending the productivity-enhancing enterprise telephony features such as voice mail and voice conferencing to the remote teleworker.
* **Facilitate data and telephony network consolidation:** Such consolidation can contribute to operational and equipment savings.

### **Call Control and Transport Protocols**

Voice communication over IP is a mix of call control signals and voice conversations coded and possibly compressed into IP packets. Both reliable (connection-oriented) and so-called “unreliable” (connectionless) transmissions are required for voice communication.

Reliable transmission guarantees sequenced, error-free, flow-controlled transmission of packets. However, because reliable transport is connection-oriented, it can delay transmission and reduce throughput. TCP provides reliable transport in the IP stack, and all voice call control functions make use of it.

The User Datagram Protocol (UDP), which provides best-effort delivery, supplies connectionless transmission in the IP stack and is used for voice conversation transport between two endpoints.

## 7.3. Voice Issues and Requirements

### **Voice Quality Issues**

Overall voice quality is a function of many factors, including delay, jitter, packet loss, and echo. This section discusses these factors and ways to minimize them.

### **Packet Delays**

Packet delay can cause voice quality degradation. When designing networks that transport voice, you must understand and account for the network’s delay components. Correctly accounting for all potential delays ensures that overall network performance is acceptable.

The generally accepted limit for good quality voice connection delay is 150 milliseconds (ms) one-way. As delays increase, the communication between two people falls out of synch (for example, they speak at the same time or both wait for the other to speak); this condition is called *talker overlap*.

### **Voice Coding and Compression**

Voice communication over IP relies on voice that is coded and encapsulated into IP packets.

The term *codec* can have the following two meanings:

* **A coder-decoder:** An integrated circuit device that typically uses PCM to transform analog signals into a digital bit stream and digital signals back into analog signals.
* **A software algorithm:** Used to compress and decompress speech or audio signals in VoIP, Frame Relay, and ATM.

A *codec* is a device or software that encodes (and decodes) a signal into digital data stream.

### **Bandwidth Considerations**

Bandwidth availability is a key issue to consider when designing voice on IP networks. The amount of bandwidth per call varies greatly, depending on which codec is used and how many voice samples are required per packet. However, the best coding mechanism does not necessarily result in the best voice quality; for example, the better the compression, the worse the voice quality. The designer must decide which is more important: better voice quality or more efficient bandwidth consumption.

#### **Voice Bandwidth Requirements**

When building voice networks, one of the most important factors to consider is bandwidth capacity planning. One of the most critical concepts to understand within capacity planning is how much bandwidth is used for each VoIP call.

### **QoS for Voice**

IP telephony places strict requirements on IP packet loss, packet delay, and delay variation (jitter). Therefore, QoS mechanisms on Cisco switches and routers are important throughout the network if voice traffic is sharing network resources with data traffic. Redundant devices and network links that provide quick convergence after network failures or topology changes are also important to ensure a highly available infrastructure.

#### **Bandwidth Provisioning**

Bandwidth provisioning involves accurately calculating the required bandwidth for all applications, plus the required overhead. CAC should be used to avoid using more bandwidth than has been provisioned.