CHAPTER 1

The State of Voice Communications

Most telephony books either begin with a history of telecommunications, or have a chapter dedicated to the subject. There is generally a discussion of the United States telephone provider monopoly, and the subsequent government intervention that created the Regional Bell Operating Companies (RBOCs). This is generally followed by a discussion of government regulations in the telephone industry, within the United States and other countries, as well.

In this book, the preceding paragraph is about the most you will see on telecom history and regulations. This is a technology book—not a history or legal book. Leave it to the lawyers to sort out legal issues in the fast-changing telecom regulatory environment. The aim of this book is to help you understand the underlying technologies of two different worlds, and minimize the obstacles you will encounter along the path to integrating your voice and data networks.

After reading this chapter, you should have a good understanding of the voice communications services currently available in traditional telephony environments. It is important that you understand the telephony features you may be required to support across the voice/data network. It is also important that you understand how these features are provisioned—from the perspective of the telephony equipment at your sites and from the perspective of the voice circuits and services provided by circuit vendors. This chapter is organized into the following sections:

- Telephony Systems
- Trunk Circuit
- Calling Features & Services

Telephony Systems

The focus of the following sections is on the systems that implement calling features and services. The sections are segregated into the following types of telephone systems:

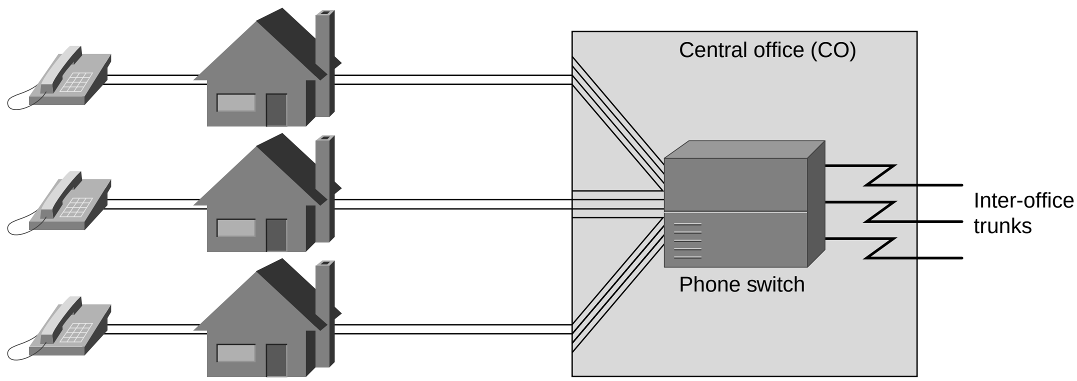
- Residential
- On-premise business
- Off-premise Centrex

Residential Systems

Residential systems are the simplest of the systems described here. The basic elements of the residential telephony system are illustrated in Figure 1-1.

NOTE With the widespread adoption of Signaling System #7 (SS7) in the core public phone networks, residential customers have access to many advanced calling features. These features were not available before SS7 was widely adopted among carriers, because there was no standard way to signal the feature requests between different carrier networks. SS7 is explored in Chapter 3, "Signaling System #7."

Figure 1-1 *Basic Elements of Residential Telephony Systems*



All of the telephone jacks in your house are wired in parallel (similar to the old coaxial Ethernet LANs) with a few pairs of wires. Each phone circuit requires a single wire pair. There is a continuous wired connection from the telephone jacks in your house to the central office (CO). At your house, you use a simple single-line telephone that provides plain old telephone service (POTS). Your telephone only provides a speaker, microphone, ringer, keypad, and a Dual Tone Multi-Frequency (DTMF) generator. The battery power required to power the phone is gleaned from the same wire pair, connected to the CO, that provides the audio path and call signaling. All of the network intelligence comes from the core network switch in the CO. The CO is a secured building, managed by a local exchange carrier (LEC), that terminates physical wires from thousands or tens of thousands of households. To facilitate signal transmission and reduce noise, these wires may have repeaters and/or loading coils (inductors that block high-frequency signals) placed anywhere along the path between your house and the CO.

In addition to the wire terminations, the CO has trunks to other COs and tandem switching nodes in the carrier's network. The residential wires, business wires, and the interoffice trunks (for example, voice circuits between telephone switches in the carrier's network) terminate on large voice switches made by a few vendors. If you have ever configured ISDN on a Cisco router, the **ISDN switch-type** command identifies the type of switch in the CO—examples include 5ess and dms100.

On-Premise Business Systems

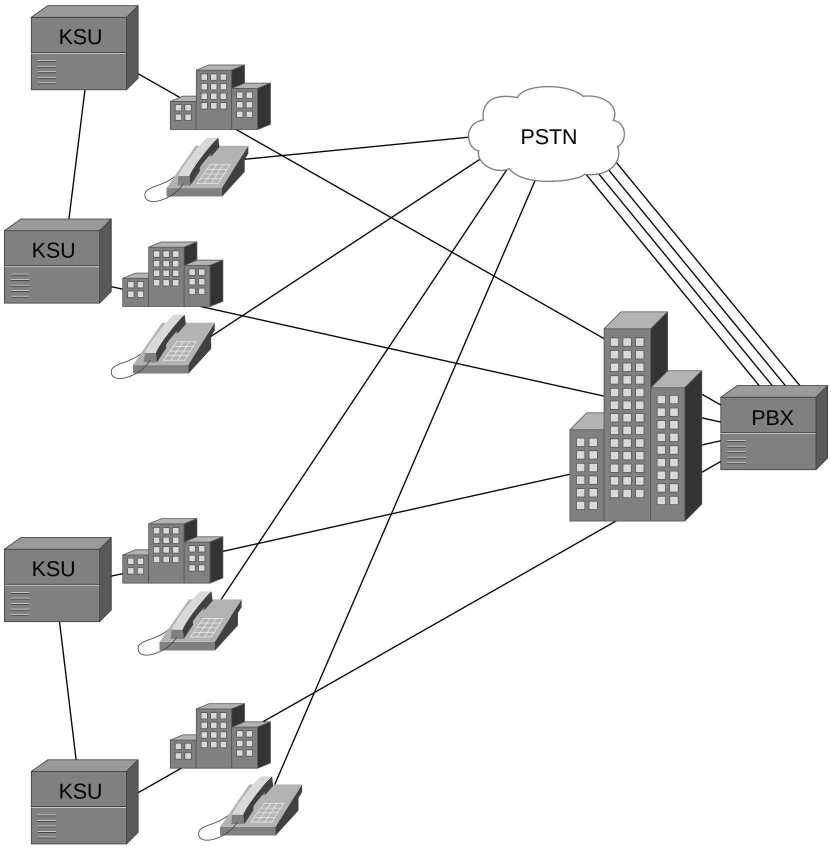
Businesses and other organizations have more complex telecommunications needs than residential customers, because there are more people to consider and because there is a greater focus on personal productivity. Just as businesses can vary greatly in size and communications requirements, the systems to support these businesses vary greatly.

Figure 1-2 provides a voice network overview of a typical organization with multiple sites. The sites are typically connected via tie lines that provide a dedicated physical path for calls between the sites. These dedicated circuits are less expensive to use than paying for public switched calls between sites. At least one of these sites must have trunks to the Public Switched Telephone Network (PSTN). It is a good idea to have at least one analog POTS phone at each site directly attached to the PSTN (that is, not going through a private voice switch) to maintain communications during emergencies like long power outages or voice switch failure.

On-premise business telephony systems may be composed of the following elements:

- Business telephones
- Analog devices
- Key System Units (KSUs) and hybrids
- Private Branch Exchanges (PBXs)
- Automatic Call Distributors (ACDs)
- Interactive Voice Response (IVR) units
- Voice mail and Auto-Attendant systems

Figure 1-2 *Typical Business Voice Network with Multiple Sites*



Business Telephones

A business telephone is typically designed to operate with the specific key system or PBX installed at that location. A key system or PBX manufacturer may offer a variety of telephone models for use with the particular system—from single-line telephones with no display, to multiline telephones with a large display and an attendant console. Business telephones offer more features than standard analog telephones, but the instruments are much more expensive—each telephone can cost hundreds of dollars. The telephones usually have a message-waiting indicator lamp that lights whenever the user receives a voice-mail message. Many business telephones have display screens where information such as caller ID or date/time is displayed. Most telephones accommodate multiple telephone lines, or appearances, so that you can juggle multiple conversations simultaneously. In addition to the keys from an analog keypad, a digital telephone may have a hold button, numerous function keys that provide quick access to advanced calling features, and numerous keys to select lines or outside trunks. The power for the phones comes from the wires that connect to the key system or PBX. The telephones are called digital because the audio signals are transmitted to and from the key system or PBX in digital form. The telephone has a codec to convert between the analog and digital signals.

Analog Devices

Analog devices in a business telephony environment include fax machines, modems, and standard analog phones. These devices require special ports to connect to the key system or PBX because they do not understand the proprietary digital signals from the phone switch. They are designed to plug directly into a POTS line. This is just another name for a regular residential line or a loop-start trunk in a business environment.

Key System Units and Hybrids

Key systems and *hybrids* are devices that enable businesses to share a small number of outside telephone lines among a larger number of employees. Since all employees are not constantly using their phones, businesses can reduce phone circuit expenses by using a fewer number of lines more efficiently. In a traditional key system, users can access any one of the analog telephone lines (for example, 1-MB circuits) that are connected to the system by pressing a button on their telephone. There is no routing logic within a traditional key system. Each button is strictly associated with a port to an outside line.

Key systems are typically scoped for businesses that need from ten to 100 telephones. A starter system might support up to 16 telephones and 6 outside trunks. A larger system might support up to 80 telephones.

In one standard key system implementation, all of the telephones are squared. (The origin of the term, squared, is unclear, but it is associated with the earliest electronic key systems.) When a telephone is squared, it means that every telephone is identically configured with a button for each outside line. In such a scenario, each of ten users might have the same three outside lines accessible from the same three-button locations. An LED or an LCD display indicates to other users when a given line is busy.

One of the defining characteristics of a key system is that there is no call routing or digit manipulation. When you press a button on the telephone for an outside line, you instantly seize the trunk associated with that button and hear a dial tone from the central office. Traditional key systems do not support T1 trunks—they generally use loop-start trunks. The key system may be optionally configured to let you dial 9 to seize the next available trunk instead of pressing a button for a specific outside line.

Key systems have evolved into hybrid key systems, which offer services more like PBXs, and in some cases offer a number of integrated voice applications. These systems may provide advanced calling features, support for T1, PRI, and direct inward dialing (DID) trunks, and may have self-contained voice mail, auto-attendant, ACD, and IVR systems.

These systems really have more functionality than a PBX, but the name hybrid key system is still used because the system may support fewer lines or trunks than a traditional PBX.

Private Branch Exchanges

A *private branch exchange (PBX)*, which is sometimes called a *private automatic branch exchange (PABX)*, is a true call-routing switch. Unlike a traditional key system that has buttons for specific outside lines, a PBX maintains a call-routing table. This may be a static table, or it may be a complex set of rules that define a *Least Cost Routing (LCR)* scheme. Do not confuse LCR with any concepts from dynamic data routing protocols. LCR simply refers to routing calls based on the least financial cost. The LCR scheme generally considers the destination number and the time of day to decide which trunk will provide the lowest per-minute cost for the call. The set of rules is statically defined in the PBX configuration.

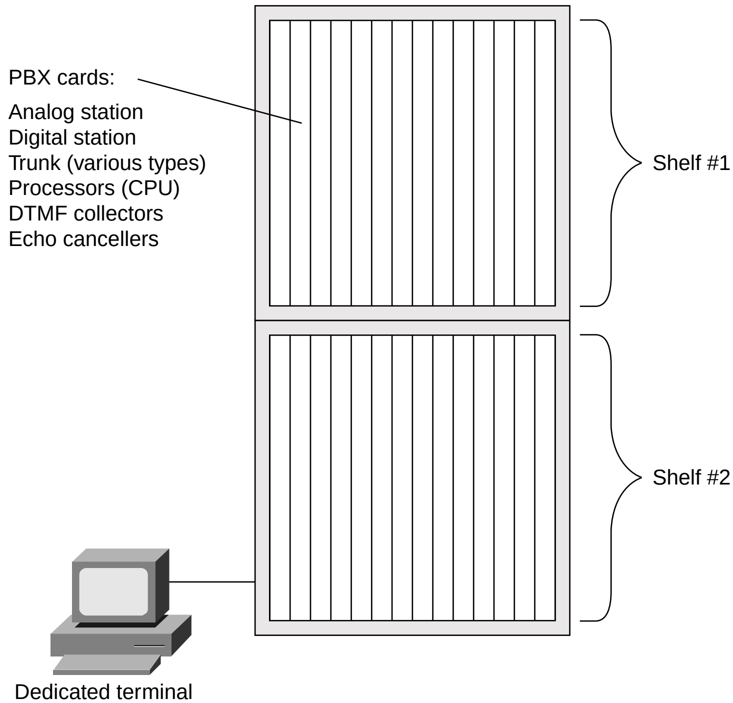
From a physical perspective, a PBX is one or more shelves (or carriers according to AT&T/Lucent terminology) with a backplane that accepts cards. Types of cards that you are likely to encounter include digital station cards, analog station cards, and trunk cards. There may be a number of other card types, including CPU cards, echo cancellers, and so on. Figure 1-3 illustrates the functional components of a PBX. Most PBX systems enable cards to be hot-swapped, with only the corresponding section of the PBX configuration needing to be modified. Some PBX cards are reconfigured by setting DIP switches on the sides of the cards. Often you have to remove a card to see the DIP switches.

WARNING Be careful to follow the procedures defined by a PBX vendor before removing cards in a PBX. You may need to unplug the card to set DIP switches, or simply to reset the card while troubleshooting problems. In some cases you must toggle a switch on the card before removing it, or you risk damaging the card and the entire phone system. In some PBXs, if you do not remove system power from a card (via a toggle switch, for example), the telephone switch can literally catch on fire when you unplug the card.

Automatic Call Distributors

An *automatic call distributor (ACD)* is just a specialized PBX. ACDs are typically associated with call centers, such as sales or support organizations. Inbound calls are distributed to call agents according to a set of rules. A call may be sent to the longest idle agent, or to the agent with the most experience with a particular product or service, for example. Agents must log on to the ACD system, so make sure that the logon process is supported as part of the voice/data integration design.

Figure 1-3 *Functional Components of a PBX*



How does the ACD get data to make call-distributing decisions? The dialed number identification service (DNIS) value from the inbound call is a start. The automatic number identification (ANI) can often provide unique customer identification. The caller may also interact with an interactive voice response (IVR) unit that collects information, such as account numbers or product serial numbers, prior to the caller reaching a live agent. This information is used as a key to search a database, and the appropriate record may be presented to the call agent via a computer application. The term *screen pop* is commonly used to describe the information that appears on the call agent's computer screen.

One interesting feature of an ACD system is that there may be more trunks than there are call agents to service the trunks! Whenever you call a business and hear, "We are experiencing unusually high call volumes—please wait for…," you are one of the callers in queue. ACD systems offer a simple priority queuing mechanism to move favored customers to the front of the line. Conversely, customers calling to discontinue a service may be pushed to the back of the line. So be wary the next time an automated system gives you the option to discontinue a service. Good ACD systems can tell you how long to expect to be on hold, or how many callers are ahead of you. Bad ACD systems, or poorly implemented ACD systems, can provide dead air that makes you think you have been disconnected.

Interactive Voice Response

Aside from collecting information for call center agents, an Interactive Voice Response (IVR) system can also offload some traffic from the call center. Simple queries like store hours or directions can be automatically processed, so call agents have more time to manage complex calls. IVR systems, when linked with back-end databases, can provide real-time and customer-specific data that further offload call centers. Processing requests such as account balance inquiries and appointment scheduling are ideally suited to IVR systems. Call center agents are happier when they do not have to perform such repetitive tasks.

Voice Messaging and Auto-Attendant

A voice-messaging (or voice-mail) system stores spoken messages destined for users that have a voice-mail account on the system. The system is normally a standalone device that connects to a PBX, though it may be bundled into a hybrid key system. Standalone–voice-mail systems normally connect to a PBX via standard phone lines or trunks for the audio path and call signaling (such as loop start, E&M), and via an RS-232 serial connection (such as Simplified Message Desk Interface [SMDI]) for signaling about which voice mailbox to access.

When a PBX forwards a call to a voice-mail system (for example, when the destination is busy or not answering), the voice-mail system needs to know the destination mailbox. This information enables the voice-mail system to play a personalized greeting, and to record a message from the caller so that the intended recipient can retrieve it at a later time. The PBX may be programmed to provide the destination mailbox to the voice-mail system via the serial SMDI connection. If the PBX is not programmed for this functionality, then it cannot forward calls to a personalized voice mailbox for each user; it can only forward calls to the main voice-mail number, where an auto-attendant processes the call.

The voice-mail system auto-attendant provides a recorded greeting such as, "Welcome to company X; please enter a mailbox number or press 1 to spell by name." Callers can then interact with the voice-mail system by pressing keys on their phone in response to voice prompts. The auto-attendant processes digit input from the caller, recalls the personalized voice-mail greeting of the intended destination, and records a message from the caller. The auto-attendant may then accept additional input from the caller to leave a message for a different recipient.

Off-Premise Centrex Systems

Carriers and service providers offer Centrex service to their customers as an outsourced telecommunications solution. Traditional Centrex service provides the functionality of a customer's PBXs from the phone switches in a CO. Centrex service can support direct inward dial (DID), abbreviated dialing between customer sites, call transfer, forward, conference, hold, and call waiting—all of the normal PBX functions. Centrex also supports voice mail, though most phones used with Centrex service are not equipped to take advantage of the message-waiting lamp. A stutter dial tone is normally used to indicate a waiting voice-mail message.

The telephones are the only equipment located at the customer sites when Centrex service is used. Customers may use normal telephones, or they may connect a PBX to the Centrex service to enable employees to use telephones with more features (for example, number display, multiple lines, message-waiting lamp, buttons dedicated to features, and so on).

Centrex is a very compelling service for some businesses, but it does have its limitations. The month-to-month service lease is a small financial commitment compared to purchasing an expensive PBX. The service is provided by highly reliable CO telephone switches that are monitored and maintained 24 hours per day, with no dependence on electricity from the customer's location. For widely distributed campuses, Centrex may be more cost effective than installing phone switches in multiple buildings or ordering off-premise extension (OPX) circuits. Centrex is conceptually more scalable than a small phone system for rapidly growing companies, in that the CO phone switch can accommodate many Centrex lines.

On the less appealing side of Centrex, you are at the whim of a phone company whenever you add an employee who needs a phone. The carrier may have a provisioning backlog or run out of Centrex lines in a busy exchange, which leaves you few options when you have new users that need a telephone. You must have a separate copper wire pair for each phone connected to the Centrex service, and the explosion of wiring becomes impractical as an organization grows.

Trunk Circuits

To provide useful communication, the telephony equipment at your site needs to interface with the equipment at other sites and with the PSTN. This is analogous to having WAN data circuits at each site and a connection to the public Internet. The types of circuits used for voice communications bear resemblance to data circuits at the most basic level—that is framing, line coding, and channelization—but the similarities end there.

The physical facility for a voice circuit is either a DS0 that delivers a single analog voice channel, or a time division multiplexing (TDM) digital facility that delivers many voice channels. In the United States, a T1 delivers 24 voice channels with in-band signaling. Most of the world uses an E1 to provide 30 voice channels, a framing channel, and a signaling channel. A digital facility can be directly connected to a T1/E1 card in the PBX, but an external channel service unit (CSU) commonly serves as a point of demarcation between the carrier-owned circuit and the customer-owned wiring and equipment. If the PBX or key system does not have a T1/E1 card, then a Channel Bank can split the T1 into separate DS0s and provide a wire pair to each analog port in the voice switch.

In the United States, you can order the following types of voice circuits from circuit vendors:

- Dial-tone lines
- Both-way trunks
- DID trunks
- ISDN PRI trunks
- Point-to-point tie lines
- Off-premise extensions
- Foreign-exchange trunks

Dial-tone lines are circuits that provide analog connectivity to the CO via a single pair of wires. Each dial-tone line is uniquely associated with a phone number that is assigned by the circuit vendor. This type of connection is used in residential phone lines. In a business environment, this circuit may also be called a *loop-start trunk* if it is connected to a key system or PBX. These types of circuits are more commonly associated with key systems than with PBXs.

Both-way trunks, also called *combination trunks*, must be connected to a key system or PBX. Ground-start signaling is normally used to prevent inbound and outbound calls from seizing the trunk at the same time. Chapter 2, "Enterprise Telephony Signaling," examines loop-start and ground-start signaling. A common way that companies use both-way trunks is for outbound calling only. This is because another circuit provides direct inward dial, so the both-way trunk effectively becomes a direct outward dial (DOD) trunk.

A *direct inward dial (DID) trunk* enables callers to directly reach employees without being transferred by an attendant. Consider a business without DID trunks that is assigned 555-1000 as the main number, and uses three-digit extensions for its employees. If you want to reach someone in this company, you must first dial the main number, and then be transferred to the extension—say, x132. If the company ordered DID trunks and reserved the number block of 555-1xxx, then you could call the person directly by dialing 555-1132.

So why is this so special? What makes it different from residential lines where you can directly reach someone? DID trunks enable more telephone numbers to be assigned than the number of trunks. You can order 24 DID trunks (a full T1) with a block of 100 phone numbers for the employees in your office, and give each of them a unique phone number for the outside world. With normal dial-tone trunks, you only get one phone number for each trunk, and the phone number is always associated with a specific physical line. DID trunk lines accept calls for any of the dialed numbers in the DID block, and the PBX forwards the call to the appropriate extension. This enables inbound calls to use any trunk for any destination in the DID block.

One disadvantage of standard DID trunks is that they only allow inbound calling. This means that you must pay for additional trunks for outbound calling. If your voice services are provisioned on a T1 local loop, then you can use some channels for a DID trunk group, and some channels for a both-way trunk group. The both-way trunks would be used for outbound calls only.

An ISDN *Primary Rate Interface (PRI)* circuit provides a 23B+D service in the United States. This means that you get 23 bearer channels that carry voice traffic, and a data channel that carries signaling traffic. ISDN can provide sophisticated calling features and services because there is a full DS0 for signaling traffic. One of the fancier calling features supported by voice PRI circuits is two-way DID. The same trunks can provide DID and enable outbound calls. On inbound calls, automatic number identification (ANI) is available before ringing begins, because it is delivered as part of the call-setup message. ISDN PRI circuits also pass the private switch automatic location identification (PS/ALI) information required for emergency services. In general, the ISDN PRI connection is the conduit that will enable integration of advanced calling features in private voice networks and the PSTN. PBX vendors may continue to use proprietary signaling schemes with their digital telephones. The public voice network has embraced SS7 to support advanced calling features. An ISDN PRI enables these features to function through both networks.

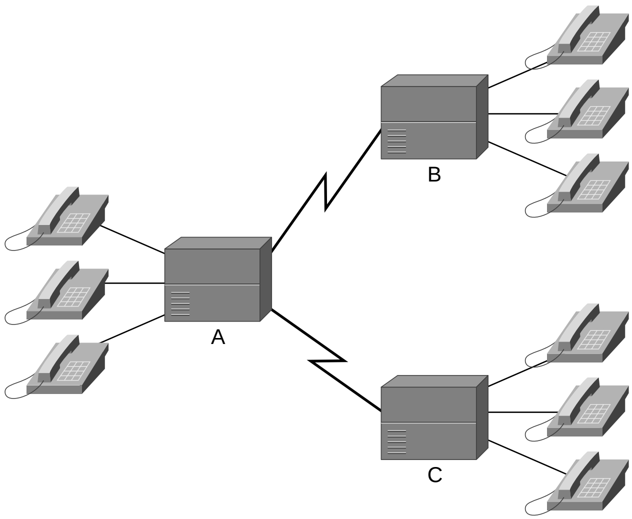
NOTE DID, ANI, and PS/ALI are described later in this chapter in the section entitled “Calling Features and Services.”

Point-to-point circuits, or tie lines, are clear channel circuits that can be used for voice or data communication. You may be accustomed to T1 data circuits that use extended super-framing (ESF) and binary-8 zero substitution (B8ZS) line coding. It is more common in the voice world to use d4 super-framing and alternate mark inversion (AMI) line coding. Use ESF/B8ZS if your PBXs support it. Just make sure that you use the same framing and line coding that the circuit vendor is using.

Across the tie line, PBXs peer directly with each other—not with the local CO. The PBXs use E&M signaling, discussed in Chapter 2, for call setup and teardown. In a typical implementation, you dial a short-trunk access code to seize a specific tie line. When the PBX at the other end of the tie line provides dial tone, you then dial extensions at that site, or dial 9 to reach an outside line from that office. This is called *tail-end hop-off,* or *toll bypass,* and it can save you a lot of money on toll calls.

In a larger tie-line network, you might have to traverse multiple tie lines to reach the intended destination. The process for doing this is analogous to source routing— every hop of the voice network must be specified as part of the dialed number. Consider Figure 1-4. In this scenario, you are a caller at site B trying to reach someone at site C. You first dial the trunk access code (TAC) for site A. After you hear a dial tone from the PBX at site A, you dial the TAC for site C. When you hear a dial tone from the PBX at site C, you can call extensions of users at the site.

Figure 1-4 *Tie-Line Network with a TAC Numbering Plan*



The process just described illustrates one of the key differences between a traditional key system and PBX. The key system does not have call-routing capabilities, so you must source route all of your calls through the network. As a workaround, a button on your telephone might be configured to dial a sequence of TACs interspersed with suitable pauses. Then, you would press a single button to get dial tone from office X. However, you still need to know which button to push to dial which office, so there is still a routing choice you must manually make with each call.

When you have a PBX network, you no longer need to know where your call is routed. You just need to know the extension of the user you want to reach. The PBX network is preconfigured to route the call out the appropriate trunk at each hop. You dial x231 to reach Chris in the San Jose office, but you do not know what path the call takes (unless you configured the PBX). The call could traverse a tie line, or the PBX may append 1-408-555-1 and send the full 11 digits to the PSTN.

You might have a network with both key systems and PBXs. In one scenario, the central site has a PBX that maintains call-routing information for the entire hub-and-spoke network. The remote sites have key systems, dial-tone trunks to the PSTN, and tie trunks to the central site. The telephones at the remote sites have buttons for dial tone from the PSTN, and dial tone from the tie lines. From a remote site, you can seize a tie trunk and dial an extension in any office directly. The PBX at the central site receives the digits that you dial, and routes the call to the proper destination. This is analogous to a hub-and-spoke data network where the central site has routes for the entire network, but the remote sites only have a default route to the central site.

Off-premise extensions, or *OPX lines*, are useful when you want to integrate remote users into your voice network. At the main site, you terminate the OPX line into your PBX. You assign an extension to that line as if a telephone is locally attached. The other end of the OPX line is directly connected to a telephone at an employee's residence, or a satellite office with a few users. Centrex systems are essentially OPX lines at business locations connected to a CO phone switch.

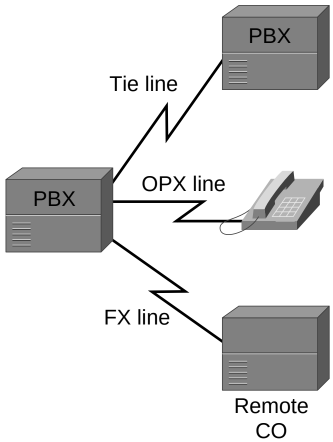
Remote telephones connected via OPX lines can place calls and receive calls as if they were directly connected to the phone switch at the main site. They do not have all of the features of digital telephones, but they can use hook-flash or key sequences, like *70, to take advantage of PBX features. A voice-mail–alert lamp will not work, but a stutter dial tone can be used to indicate voice mail.

Foreign exchange lines, or *FX lines*, give an organization a local telephone presence in areas where it may not have a local physical presence. Imagine that you have a business with offices in San Francisco but many clients in Los Angeles. You may not want your clients in Los Angeles to make long-distance calls to speak with you. In this case, you can order an FX line from your business office to the circuit vendor's CO in Los Angeles. You then have a local number for Los Angeles callers, but it rings directly to you in San Francisco. This is basically a back-hauled voice line. This application for FX lines is not commonly used now, because 800-number service provides the same function and more. With 800 service, callers have toll-free access to your business, and you have the same 800 number for many different geographic regions.

FX lines are still useful if your business makes a large number of calls to a specific remote destination. If you have a tie line to a business office in that location, you can use tail-end hop-off. If you do not have a business office in the area, then you can order an FX line and use that line for local calls to the area.

Figure 1-5 summarizes the relationship between tie lines, OPX lines, and FX lines. Remote connections between key systems or PBXs are provided with tie lines. Remote connections between a voice switch and a telephone are provided with OPX lines. Remote connections between a voice switch and a distant CO are provided with FX lines.

Figure 1-5 *Relationship Between Tie, OPX, and FX Lines*



Calling Features & Services

In addition to basic call connection and disconnection, a variety of additional calling features and services are available for residential and business telephone users. The most common features and services are grouped here according to the following categories:

- Call Session Management
- User Identification
- Convenience
- Security
- Emergency Response

Call Session Management

These features help you control how new or existing call connections are handled, and how additional call connections are handled when other connections already exist. You have real-time control over where calls are terminated—whether to a different phone set, multiple phone sets, or to a voice mailbox. These features are provided by the phone switch (for example, hybrid key system or PBX) at each site.

Call hold places an active connection in an idle state without disconnecting the call. The endpoint of the call is temporarily suspended, but it is not moved. At this point, you can receive or place a call on another line, or perform some action while the other party waits.

Call hold is often implemented with music on hold, to reassure the holding party that the connection is still active.

Call park is used to transfer an active call connection to a temporary holding point, from which you may retrieve the call. An extension identifies the temporary holding point, or where the call is parked. Call park is often used in conjunction with an intercom system that notifies users of calls that are waiting. For example, a loudspeaker might announce, “Pat, you have a call parked on 101.” Pat would then retrieve the call by pressing a button on the phone (for example, Intercom) and entering the extension 101. This feature is similar to call transfer, except that you manually accept the call from any telephone instead of passively receiving the call when your phone rings. This is beneficial in environments where you need to receive calls but may not be close to your own phone. Instead of trying to track your present location, the attendant can notify you over the loudspeaker.

Call transfer enables you to transfer your end of an active call connection to another user without disconnecting the remote party. Attendants (or receptionists) use this feature extensively to direct calls to various extensions. This feature saves the remote party from having to redial a different number in your organization. If you only have an extension and no direct phone number, then a party from outside of your network cannot call you without being transferred.

Call conference enables three or more people to simultaneously communicate. There are two types of audio conferencing: spontaneous and planned. The spontaneous type (that is, ad hoc conference) enables an active call connection to be converted to a conference call by the addition of other parties at any time. A PBX or hybrid key system typically supports this feature. A planned conference (or meet-me bridge) normally has a “conference bridge” phone number to which all parties must connect, and a passcode that identifies the specific conference. Planned conferences are usually provisioned on specialized equipment that accommodates more participants than spontaneous conferences. Spontaneous conferences have the advantage that participants need not be notified prior to the conference.

Call waiting notifies you of inbound calls when you are already engaged in an active call connection. This feature is useful when you do not want to miss calls. In business environments, this feature is typically implemented with multiline phones. With Centrex or residential phone service, you can be notified via an audible tone during your active conversation.

Call forwarding enables calls destined for your phone to be directed elsewhere in response to different conditions. Calls can be forwarded to your voice mailbox when your line is busy or you do not answer after a preset number of rings. This feature can function with call waiting. You can also specify a number to which all calls should be forwarded, such as another call agent, or your mobile phone. Some systems offer additional options for conditionally forwarding calls.

Do not disturb (DND) creates the appearance that your phone is busy. If you have voice mail provisioned, all calls will be forwarded to your voice mailbox. This feature is useful when you are woefully behind at work and need to get out of interrupt mode to get some work done.

Voice messaging provides the basic functions of a phone answering machine and more. Callers may have the option of marking messages as urgent or private, reaching a live person by pressing a key sequence (normally 0), or transferring to a different extension. You may be notified when you have voice messages by a lamp indicator on your desk telephone or a stutter dial tone when you pick up the phone. These notifications are called Message Waiting Indicators, or MWI. In addition, the voice-mail system may send an alert to your pager based on conditions that you define (for example, all messages, or only urgent messages). When you connect to your password-protected–voice-mail account and listen to your messages, the urgent messages are played before the normal messages. You can sometimes control how messages are played back—faster or slower, rewind, fast forward, repeat, and so on. You have the option of replying to a message or forwarding it to others, unless the message has been marked private. You may be able to send messages to people in your voice-mail network by spelling their names as opposed to entering their extension numbers. This voice-mail network may extend between all of the sites in your organization, or each site may have an independent voice-mail system.

Identification

Several technologies provide identity information about the party that originates a call and the party that receives a call. This information can be used by the calling and called party in a number of ways.

Caller ID is provided on loop-start trunks by telephone circuit vendors (Chapter 2 describes loop-start circuits in detail). This service is available to residential phone subscribers, as well. The actual telephone number at the source of the call is passed to the receiver as an analog signal between the first and second ring. This means that calls answered during the first ring may not be identified. The party that originates the call has the option of blocking the source telephone number, except when the call is destined for 800 or 900 numbers. The exceptions are intended to allow billing reconciliation.

Automatic Number Identification (ANI) is a service available on ground-start, T1, or PRI trunks (each of these circuit types is discussed in detail in Chapter 2). For calls from a residence, the ANI and caller ID may be the same, but for calls from a business, the ANI delivers the bill-to number (BTN) of the calling party to the destination. ANI does not deliver the exact phone number that placed the call unless this is the same as the BTN. Parties that originate calls to toll-free (that is, 800 numbers in North America) or premium rate services (that is, 900 or 976 numbers in North America) cannot block their ANI, because this information is required by the receiving side of the connection for billing purposes.

Dialed Number Identification Service (DNIS) is generally used when multiple 800 or 900 numbers are routed by the telephone circuit vendor to the same trunk group. An organization might advertise separate phone numbers for different types of calls, but the same call center might receive all of the calls. The DNIS value enables the call center to track which number the caller dialed, and to route the call to an appropriate call agent.

Custom ringing uses the calling party information to generate special ringing tones when the call is from a predefined party. This helps you identify who is calling you even if you do not have a display to see the phone number of the calling party. A common feature of key systems and PBXs is to provide a separate ringing tone for calls from within the private network and calls from the PSTN. This enables businesses to prioritize customer calls ahead of internal calls. In a residential context, custom ringing is also used to identify which number is called when multiple phone numbers are associated with the same line. This is a popular service to provide a separate "teenager phone number" without physically ordering two separate phone circuits.

Call return is the service commonly invoked by dialing *69 on a POTS phone in the United States. When you dial *69, the number that last attempted to call you is dialed. Even if the last call attempt to your number was not established (for example, the call is abandoned after two rings), the number is still considered the last number to call you. If that number uses caller ID blocking, then the phone network is unable to complete the call.

A variety of quite useful call logging features are now available. Intelligent phone sets or computer-based telephony terminals can track a list of recent missed calls, received calls, and placed calls. Calls can be returned to numbers from any of these lists. These logging features depend on the availability of caller ID or ANI.

Caller ID blocking allows you to maintain a degree of privacy by not revealing your number to destinations that use caller ID. This privacy cannot be maintained when you call 800 numbers, because the destinations have a right to know who is making the calls for which they pay. Similarly, you cannot maintain your privacy while dialing 900 numbers because they must know whom to bill.

NOTE The Advanced Intelligent Network (AIN) features that are accessed by dialing *<digit><digit> in North America, such as *69 and *70, are referred to as Custom Local Access Signaling Services, or CLASS.

Convenience

Though several of the features that make use of calling- and called-party identification can be considered features of convenience, they are discussed in the preceding section because of their dependence on identification functions. The features of this section are convenience features that do not require special information from the network. Generally, these features save you from the tedium of dialing long digit sequences when alternatives exist.

The *intercom/paging* functions of a key system enable users of the system to communicate without using outside lines. The phone connection may be switched directly between the user phones or to an audio broadcast system in the building. Predefined text messages may also be sent to users with display phones. Each local user is accessed via a short extension, typically three or four digits.

Systemwide and personal speed dialing plans use short key sequences to specify calling destinations. The key system or PBX recognizes the short key sequence and substitutes the correct expanded destination number.

Redial enables you to call the last number that you dialed again, even if the connection was not completed. This is useful to retry a busy number, to resume a conversation that was accidentally disconnected, or whenever you want to speak to the same party again.

Repeat dialing is useful when the intended destination of a call is busy. A key system, PBX, or a service from the circuit vendor will attempt to place the call at predefined intervals, and ring your phone set when the destination line is no longer busy. When a telephone circuit vendor provides this service, you may have instant knowledge of when the destination line is free. With a computer-based telephone terminal, key system, or PBX, you will not know that the destination is free until the predefined interval arrives for the call-retry mechanism.

Voice activated dialing is an emerging feature that frees your hands from the keypad. A digital signal processor (DSP) recognizes a characteristic audio sample, and dials the number associated with the audio sample. The audio sample can be the name of a person to call, or it can be an open-ended system where you speak the actual numbers to be dialed. Do not confuse this feature with voice activity detection (VAD), which is a bandwidth-saving technique used in digital voice transmission.

Security

A number of telephony features are designed to reduce your exposure to threatening or harassing calls, or to prevent unauthorized use of your phone system. These features enable you to stop receiving unwanted calls, or to identify threatening and harassing callers. They can also discourage dishonest employees and phreakers (telecom hackers) from exploiting your phone system for toll fraud.

Anonymous call rejection (ACR) stops all calls to your number that originate from numbers that use caller ID blocking. The caller hears a system message indicating that such calls are not accepted, followed by instructions to remove caller ID blocking for the specific call.

Call screening prevents calls from numbers that you have identified to be threatening or harassing. The caller hears a system message indicating that the desired destination will not accept the call.

Call trace is a service that must be specially ordered from the phone company. With this feature, call detail reports are immediately sent to law enforcement officials for criminal investigation. Some organizations, like public school facilities, often have requirements for this service.

Number restrictions are often used to prevent calls from your facility to 900 numbers, international numbers, or long-distance numbers. Most key systems and PBXs have a mechanism to create multiple restriction profiles, and assign those profiles specific users. For example, your company executives may need to place international calls to arrange business meetings, but members in the sales department only need to make domestic calls.

Emergency Response

This is an important service that should not be neglected in the process of integrating your voice and data networks. Table 1-1 lists the emergency response telephone numbers for various countries.

Table 1-1 *Emergency Response Telephone Numbers in Different Countries*

Country	Ambulance	Fire	Police
Armenia	03	01	02
Australia	000	000	000
Brazil	193	193	190
Canada	911	911	911
China	120	119	110
Czech Republic	155	150	158
Denmark	112	112	112
Egypt	123	180	122
Germany	115	112	110
Hong Kong	999	999	999
Iceland	112	112	112
India	102	101	100
Israel	101	102	100
Italy	118	115	112
Japan	119	119	110
Korea	119	119	112
Malaysia	999	994	999

continues

Table 1-1 *Emergency Response Telephone Numbers in Different Countries (Continued)*

Country	Ambulance	Fire	Police
Mexico City	080	080	080
New Zealand	111	111	111
Portugal	112	112	112
Singapore	995	995	999
South Africa	112	112	112
Sweden	112	112	112
United Kingdom	999	999	999
USA	911	911	911

You can find a more comprehensive list at the following URL:

www.ambulance.ie.eu.org/Numbers/Index.htm

In North America, it is not enough that you provide call routing for 911; you must also ensure that the Public Safety Answering Point (PSAP), or equivalent agency, receives information from your site to properly dispatch emergency services.

Consider the simple case of a business operating in a small building. When someone at this business site places a call to 911, the ANI information can be used by the PSAP to uniquely identify the location from which the call is placed. But for larger organizations, the ANI information is not sufficient. Consider a campus with multiple buildings, multiple floors in each building, and multiple rooms on each floor. The ANI information could be the same for a 911 call sourced from any room in any of these buildings! How would an ambulance know where to go?

Telephone circuit vendors offer a service that enables PBXs to provide more information about the location of the call. PBXs can send a Private Switch Automatic Location Identifier (PS/ALI) message as part of the call to 911, so emergency dispatchers can identify the building, room, and location from which the emergency call was placed. In some networking environments, there are big liability issues associated with maintaining the customer records of which extension is in which location. In these environments, extra care must be exercised when designing an integrated voice/data network.

Summary

This chapter has reviewed a variety of telephone features and services, the telecom equipment that provides the features, and the types of circuits that connect the equipment between sites. Chapter 3 explores the signaling details for the trunk types presented here.