Chapter 2 - Circuit Signalling

All circuits, analogue and digital, use signalling methods to communicate. Although the signalling methods used will vary depending on the type of circuit, these signalling techniques can be placed into one of three categories:

- Supervisory Signalling
- Address Signalling
- Informational Signalling

Supervisory Signalling

Supervision involves the detection of changes to the status of a circuit, which can be either a loop or trunk. In other words, supervisory signalling is used to indicate the state of a circuit. Once these changes are detected, the supervisory circuit generates a predetermined response. The supervisory signals used by trunks operate in a similar manner to those used by access circuits or loops; however, the supervisory signals used by trunks signal the state between two switches rather than the intent to place or terminate a call. There are three different types of supervisory signals, which are:

1. On-hook
2. Off-hook
3. Ring

When a telephone handset is in the cradle, the circuit is said to be on-hook. The telephone switch, such as a PBX, prevents current from flowing through the telephone handset. When in an on-hook state, the circuit is said to be open, thus preventing the current from flowing through the telephone. In this state, only the ringer is active. The `show voice-port summary` command illustrates the on-hook supervisory state of several analog FXO, FXS, and E&M ports on a Cisco IOS router, indicating that they are presently not being used:

```
R1# show voice port summary

<table>
<thead>
<tr>
<th>PORT</th>
<th>CH</th>
<th>SIG-TYPE</th>
<th>ADMIN</th>
<th>OPER</th>
<th>IN STATUS</th>
<th>OUT STATUS</th>
<th>EC</th>
</tr>
</thead>
<tbody>
<tr>
<td>0/0/0</td>
<td></td>
<td>fxo-ls</td>
<td>up</td>
<td>dorm</td>
<td>idle</td>
<td>on-hook</td>
<td>y</td>
</tr>
<tr>
<td>0/0/1</td>
<td></td>
<td>fxo-ls</td>
<td>up</td>
<td>dorm</td>
<td>idle</td>
<td>on-hook</td>
<td>y</td>
</tr>
<tr>
<td>0/0/2</td>
<td></td>
<td>fxo-ls</td>
<td>up</td>
<td>dorm</td>
<td>idle</td>
<td>on-hook</td>
<td>y</td>
</tr>
<tr>
<td>0/0/3</td>
<td></td>
<td>fxs-ls</td>
<td>up</td>
<td>dorm</td>
<td>idle</td>
<td>on-hook</td>
<td>y</td>
</tr>
<tr>
<td>2/0/0</td>
<td></td>
<td>fxs-ls</td>
<td>up</td>
<td>dorm</td>
<td>on-hook</td>
<td>idle</td>
<td>y</td>
</tr>
<tr>
<td>2/0/1</td>
<td></td>
<td>fxs-ls</td>
<td>up</td>
<td>dorm</td>
<td>on-hook</td>
<td>idle</td>
<td>y</td>
</tr>
<tr>
<td>2/1/0</td>
<td></td>
<td>e&amp;m-wnk</td>
<td>up</td>
<td>dorm</td>
<td>idle</td>
<td>on-hook</td>
<td>y</td>
</tr>
<tr>
<td>2/1/0</td>
<td></td>
<td>e&amp;m-wnk</td>
<td>up</td>
<td>dorm</td>
<td>idle</td>
<td>on-hook</td>
<td>y</td>
</tr>
</tbody>
</table>
```

When the telephone handset is removed from the cradle, the circuit transitions to an off-hook state and the switch hook toggles to a closed state. This results in current flowing through the electrical loop. The flowing current informs the telephone switch that the subscriber is requesting to place a telephone call. When the telephone network senses the off-hook state via the current flow, it provides a signal in the form of dial-tone that it is ready to accept the call. As is the case with the on-hook state, the `show voice-port summary` command can be used to view the off-hook state for analog interfaces on Cisco voice gateways.

When a subscriber makes a call, the telephone sends voltage to the ringer to notify the recipient of an incoming call. The caller also receives a ringback tone from the telephone switch, which alerts the caller that the telephone switch is sending ringing voltage to the called party. It is important to
know that only the ringing that the recipient (i.e. the called party) hears is the supervisory signal; the ringback tone that the caller hears is simply a call-progress indicator and is not a supervisory signal.

**Address Signaling**

Address signaling represents the called party number’s dialed digits, i.e. the digits that are assigned to a subscriber (end user). Address signaling methods transmit this information from the call originator. Address signaling is either conveyed by dial pulse or Dual Tone Multi Frequency (DTMF) -- with the latter being the most commonly used method.

Pulse dialing is considered a legacy signaling method. Pulse dialing is an in-band signaling technique used by telephones that have a rotary-dial switch. The large numeric dial-wheel on a rotary-dial telephone spins to send digits to place a telephone call. These digits must be produced at a specific rate and within a certain level of tolerance. Each pulse consists of a break and a make, which are achieved when the local loop circuit is opened and closed. The break segment is the time during which the circuit is open and the make segment is the time during which the circuit is closed. Each time the dial-wheel on the rotary-dial telephone is turned the bottom of the dial closes and opens the circuit leading to the CO switch or the PBX switch.

A governor inside the dial controls the rate at which the digits are pulsed. When the dial is released, the spring rotates the dial back to its original position, and a cam-driven switch opens and closes the connection to the telephone company. The number of consecutive opens and closes (i.e. breaks and makes) represents the dialed digits.

When the telephone is off-hook, a make occurs and the caller receives a dial tone from the CO switch. Then the caller dials digits, which generate sequences of makes and breaks that occur every 100 milliseconds (ms). The break and make cycle must correspond to a ratio of 60 percent break to 40 percent make, which means that the make cycle occurs every 60 ms and the break cycle occurs every 40 ms. The phone then remains in a make state until another digit is dialed or the phone is put back to an on-hook state, which is equivalent to a break state.

Dial pulse addressing is a very slow process because the number of pulses generated equates to the digit dialed. Therefore, in order to increase the speed of dialing, a new dialing technique called DTMF was developed.

With DTMF, each button on the keypad of a touch-tone pad or push-button telephone is associated with a pair of high and low frequencies. On the touch-tone telephone keypad, each row of keys is identified by a low-frequency tone and each column is associated with a high-frequency tone, as illustrated in the following diagram:
DTMF uses only two frequency tones per digit as illustrated in the diagram above. For example, if the digit 1 is dialed, only frequency tones 697 and 1209 are generated, instead of the numerous make and break pulses that would be generated by pulse dialing. The combination of both tones (using two different frequencies) notifies the telephone company of the number called, hence the term Dual Tone Multi Frequency. The frequencies illustrated in the diagram above are standard and were selected for DTMF dialing because they are not affected by ordinary background noise.

As is the case with pulse dialing, the timing used in DTMF is still a 60 ms break and 40 ms make for each frequency generated. Additionally, it should be noted that while DTMF uses in-band signaling in the same manner as pulse dialing, it is common practice that DTMF tones are sent out-of-band via the DTMF relay feature. This resolves distortion problems which can occur when the signal is sent in-band.

The address signaling method used on a particular voice port can be viewed by issuing the `show voice port [slot/sub-unit/port]` command as illustrated in the following output:

```
R1#show voice port 0/0/0

Foreign Exchange Office 0/0/0 Slot is 0, Sub-unit is 0, Port is 0

Type of VoicePort is FXO

Operation State is DORMANT

Administrative State is UP

-[Truncated Output]-
```
Voice card specific Info Follows:

**Dial Out Type is dtmf**

- [Truncated Output] -

### Informational Signaling

Informational signaling is used to provide feedback to the calling party or the called party. It is common for informational signaling tones to be referred to as network progress tones or alerting tones. These informational signals are generated by the CO switch during a telephone call and may include the following:

<table>
<thead>
<tr>
<th>Type</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Dial tone</td>
<td>This indicates that the telephone switch, e.g. CO switch or PBX switch, is ready to receive digits from the telephone handset</td>
</tr>
<tr>
<td>Busy</td>
<td>This indicates that the call cannot be completed because the telephone at the remote end, i.e. the called party, is already in use</td>
</tr>
<tr>
<td>Ringback</td>
<td>This indicates that the telephone switch is attempting to complete a call on behalf of the subscriber</td>
</tr>
<tr>
<td>Congestion</td>
<td>This indicates that congestion in the long-distance telephone network is preventing a call from being processed</td>
</tr>
<tr>
<td>Reorder tone</td>
<td>This indicates that the local telephone circuits are busy, which is preventing the call from being processed</td>
</tr>
<tr>
<td>Receiver off-hook</td>
<td>This indicates that the receiver has been off-hook for an extended period of time without placing a call</td>
</tr>
<tr>
<td>No such number</td>
<td>This indicates that the subscriber has placed a call to a number that does not exist</td>
</tr>
</tbody>
</table>

Now that we understand circuit signaling, the following section puts it all into perspective by demonstrating how this signaling is used in basic telephone call.

In the initial state, referred to as the on-hook phase, both telephones are in the on-hook state as neither subscriber is initiating a call. In this state, the 48 VDC circuit from the telephone set to the CO switch is open and the CO switch contains the power supply for the DC circuit. The power supply located at the CO switch prevents a loss of telephone service when the power goes out at the subscriber's location. Only the ringer is active when the telephone is in this position, as illustrated in the diagram below:

![Diagram](image)

Next, subscriber 678-555-1234 decides to place to call to subscriber 770-555-1234 and lifts the handset from the telephone cradle. The switch hook closes the loop between the CO switch and the telephone set and allows current to flow. The CO switch detects this current flow and transmits a dial tone to the telephone set.
The dial tone signals that the customer can begin to dial. The CO switch generates a dial tone only after the switch has reserved registers to store the incoming address, which essentially means that the customer cannot dial until a dial tone is received. If there is no dial tone, then the registers are not available. This is referred to as the off-hook phase and is illustrated in the following diagram:

In the third step, the dialing phase, subscriber 678-555-1234 dials the desired digits using either a rotary phone that generates pulses or a touch-tone phone that generates tones. The subscriber telephone can use two different types of address signaling, i.e. Pulse dialing or DTMF dialing, in order to notify the telephone company where the subscriber calls. The pulses or tones are transmitted to the CO switch across a two-wire twisted-pair cable, referred to as tip and ring lines. This step is illustrated in the following diagram:

In the fourth step, the switching phase, the CO switch translates the pulses or tones received from the subscriber into a port address that connects to the telephone set of the recipient, or called party. This connection could go directly to the requested telephone set, if it is a local call, or go through another switch or even several switches, if it is a long-distance call or international call, before it reaches its final destination. The switching phase is illustrated in the following diagram:

Following the fourth phase, the CO switch attempts to connect to the called line, and if successful, the switch sends a signal to the line, which rings the phone of the called party. While ringing the phone of the called party, the CO switch sends an audible ring-back tone to the caller. This ring-back lets the caller know that ringing occurs at the called party.

The CO switch transmits 440 and 480 tones to the caller phone in order to generate a ring-back. These tones are played for a specific on time and off time. For example, the ringing pattern in the United States is 2 seconds of ringing tone followed by 4 seconds of silence; however, Europe uses a double ring followed by 2 seconds of silence. If the called party phone is busy, the CO switch sends a busy signal to the caller. This phase is illustrated in the following diagram:
In the final phase, the talking phase, subscriber 770-555-1234 hears the phone ring and decides to answer. As soon as the called party lifts the handset, an off-hook phase starts again, this time on the opposite end of the network. The local loop is closed on the called party side, so current starts to flow to the CO switch. This switch detects current flow and completes the voice connection back to the calling party phone. Voice communication can start between both ends of this connection as illustrated in the following diagram:

**Analog Circuits**

Analog refers to the transmission of electromagnetic information achieved by adding signals of varying frequency or amplitude to a carrier wave of a given frequency. The PSTN uses analog technology which is typically represented as a series of varying sine waves. Analog data signals are generated as voltage. All analog signals are identified by four main characteristics. These four core characteristics are amplitude, frequency, wavelength, and pulse.

- The amplitude is a measure of the signals (waves) strength at any given time. The frequency is the number of time the wave’s amplitude cycles from its starting point, through its highest amplitude and its lowest amplitude, back to its starting point over a fixed period of time. Frequency is expressed in cycles per seconds, or hertz (Hz).

- Wavelength is the difference between the corresponding points on a wave's cycle, for example, between one peak and the next peak. Wavelengths are expressed in meters or feet. The wavelength is inversely proportional to the frequency, meaning that the higher the frequency, the shorter the wavelength, and vice versa.

- The phase refers to the progress of a wave over time in relationship to a fixed point. If, for example, two waves start at the same time, with both being at their highest amplitude, the two waves would be in phase. However, if both waves started at the same time, with the first wave starting at its lowest amplitude and the second wave starting at its highest amplitude, the waves would be said to be 180 degrees out of phase. These concepts are illustrated in the following diagram showing two different analog waves in red and blue:
A typical analog network is comprised of telephone handsets, drop wires, local loops, a CO, and toll trunks. Telephone models come in all sizes, shapes and colors. These can include rotary-dial, touch-tone and even cordless phones.

The wire that runs from the subscriber telephone to the telephone pole is referred to as drop wire. While the drop wire may be overhead, it is also important to know that it can also be underground. The drop wire terminates to the demarcation point (demarc) which is the point at which the subscriber’s personal responsibility for the telephone responsibility for the telephone, line, and equipment ends and that of the telephone provide begins. Overhead drop wires are more common in rural areas or developing countries.

The drop wire may be combined into several distribution cables, referred to as feeder cables. These feeder cables are then connected to the CO via twisted-pair, coaxial cable or even fiber. If twisted pair cabling is used, it can vary in size from 16 American Wire Gauge (AWG) with a diameter of 0.05082 inches to 26 AWG with a diameter of 0.01594 inches.

Each CO is responsible for managing all telephone calls within its area and either switches the call to the called party in the same exchange or to a trunk and into another CO's area. The side of the CO closest to the local subscriber is referred to as the line-side interface or local loop and the side of the CO connecting to another CO is referred to as the trunk-side interface.

The local loop is primarily responsible for providing battery feed, over-voltage protection, telephone set ringing, call supervision, signal coding, hybrid two-four wire conversion, and circuit testing. The battery feed, which is typically 48 VDC, from the local loop provides power to the customer telephone set, telephone signaling, high AC impedance, as well as low DC resistance. Over-voltage protection protects subscribers and telephone sets from transient voltages due to short circuits, lightning surges, and electrical power lines.

Conventionally, the local CO is a two-wire switch permitting conversation on both directions over the same pair of wires. However, when a signal needs to be transmitted over long distances, the signal must be amplified and transmitted over a four-wire configuration. Hybrid transformers designed to permit full-duplex operation performs the two-to-four wire conversion. Four-wire circuits have two pairs of wires for simultaneous conversation in both directions. If the call needs to be switched between two CO locations, then the call would be sent across a toll trunk line.

The most common components described in this section are illustrated at a very high level in the following diagram:
FXO and FXS Ports

Given that FXO and FXS ports have a lot of similarities, they are going to be described together. In addition to their similarities, the most common differences between FXO and FXS ports will also be highlighted and described in this section.

Foreign Exchange Station (FXS) ports are the ports that you plug a telephone, fax machine, or modem into. These ports provide the telephony service for these analog devices. The FXS port has the ability to provide ring voltage, dial tone, and other basic signaling to the end station. The FXS port connects with a standard RJ-11 connection. The standard RJ-11 pinout and pinout signals are illustrated in the following diagram:

<table>
<thead>
<tr>
<th>PIN</th>
<th>Signal</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>-----</td>
</tr>
<tr>
<td>2</td>
<td>-----</td>
</tr>
<tr>
<td>3</td>
<td>Ring</td>
</tr>
<tr>
<td>4</td>
<td>Tip</td>
</tr>
</tbody>
</table>

Like FXS ports, FXO ports also connect with a standard RJ-11 connection. However, rather than supplying the signaling and voltage needed for basic telephony equipment, FXO ports are used to connect subscriber devices to the CO or customer PBX to receive subscriber services.

When using FXO ports, it is important to use only an FXO port that is approved for the specific country or region to connect to the PSTN. If this is not possible, then the FXO port should be connected to a PBX, which would then be connected to the PSTN instead. This is because FXO ports emulate the operation of a telephone handset waiting for ring voltage from the CO switch.

When deciding on FXO and FXS ports, it is important to keep in mind that FXS ports should never be connected to the PSTN because the FXS interface emulates the CO switch to the endpoint device by providing dial tone and ring voltage. Do not connect endpoints to an FXO port. If an FXO was inadvertently used in place of the FXS, the endpoint may receive signaling; however, it will be unable to ring the phone to alert the user of the incoming connection. Additionally, if an FXS port is used in place of the FXO, it will not anticipate the ring voltage on the line, thus it will be unable to close the loop to complete the connection.

Cisco FXO ports support a feature referred to as FXO Power Failover. The FXO Power Failure feature is a hardware feature built into the FXO cards that allows connectivity to an analog phone patch into the right pair of wires to be activated by a relay if power to the Cisco router containing the High-Density Analog Network Module (NM-HDA) module fails. The High-Density Analog Network Module supports an RJ-21 connector and both FXS and FXO traffic.

This allows PSTN calls to be made via the FXO line normally connected to the router from a designated emergency phone in the office while power is out. For an FXO Expansion Module (EM) in slot 0 of the NM-HDA, the analog phone must be connected to pair 14 on the RJ-21 connector to take advantage of the FXO Power Failover feature. For an FXO EM in slot 1 of the NM-HDA, the analog phone must be connected to pair 24 on the RJ-21 connector to take advantage of the FXO Power Failover feature. If both EMs are populated with FXO modules, two emergency phones can be used.
FXO and FXS ports share the same signaling types. These signaling types are used to 'seize' the line in preparation for a call. This line seizure is also referred to as 'starting' since it completes the circuit, enabling voltage to flow, thus 'starting' the call. The two signaling types used by FXO and FXS ports are ground-start and loop-start signaling. Both of these signaling types will be described in detail shortly.

Both ground-start and loop-start have three signaling states, which are the idle, seizure and ring states. In the idle state, the handset is on-hook and no voltage is being applied to indicate an incoming call. The seizure state occurs when the handset goes off-hook. This state completes the loop and allows current to flow. In the ring state, voltage is passed to the ring generator to indicate an incoming call. The same wires that carry voice are used by loop-start and ground-start to supply signaling, i.e. the signaling is performed in-band. This technique enables telephone companies to reduce the number of wires terminated at the subscriber location.

Loop-start signaling is used on local loops to initiate a call. It involves the breaking and connecting of the 48V circuit loop originating from the CO. In the on-hook state, there is a break in the loop so no voltage is passing. When the telephone set goes off-hook, the loop is closed and current flows.

On the remote end, i.e. the called party, when the incoming call is detected, the CO or PBX equipment supplies AC current to the ring line, causing the ring generator to active, thus ringing the telephone. When the call is answered, the loop completes and voltage ceases to flow to the ring generator and the call is connected. The diagram illustrated below shows how loop-start signaling operates on a telephone connected to a Cisco router FXS port:

Loop-start signaling works quite well for simple, single-line connections such as homes. It also has the advantage of not requiring a common ground between the CO or FXS port and the subscriber connection. However, it does have some disadvantages as well.
The first disadvantage is that loop-start signaling is prone to glare, which is a condition that occurs when two parties want to speak to each other and call each other simultaneously, thus getting a busy signal and getting a busy tone. The second disadvantage is that loop-start signaling does not provide remote-end disconnect monitoring. In other words, there is no mechanism to detect that the remote end has returned to the on-hook state. It is for these reasons that ground-start signaling was developed.

Ground-start signaling works in an almost similar manner to loop-start signaling. However, ground-start signaling can detect when loops have been seized at both ends and does not have the inherent glare weaknesses of loop-start signaling.

Ground-start signaling requires that the 48V loop be grounded on both sides of the connection. When the line is idle, the subscriber has a break in the ring and the CO or FXS port has a break in the tip and no voltage is supplied. When the subscriber equipment goes off-hook, the ring is grounded, allowing voltage to pass through and when the CO detects this, it grounds the tip. When the subscriber equipment detects the tip ground, it closes the loop and removes the ring ground, which closes the circuit.

As is the case with loop-start signaling, when a call comes in, AC voltage flows over the ring wire, causing the ring generator to produce a ring tone. The CO or FXS senses the current flowing from the tip and ring loop, and then removes the ringing tone. The PBX or FXO must sense the tip ground and ringing within 100 ms or the circuit times out and the caller has to reorder the call. This 100-ms timeout helps prevent glare. The figure that follows illustrates the basic operation of ground-start signaling:

![Ground-start signaling diagram](image)

FXO and FXS ports use analog address signaling, which can either be pulse dialing or DTMF tones. This address signaling is transmitted only from the FXO port to the FXS port and is used to indicate the final destination of the call. This is often confusing a confusing concept; however, as you may recall, earlier in this chapter we learned that FXO ports are the ports on subscriber devices, such as analog telephones, fax machines and modems that connect to the PSTN or PBX by way of an FXS port. This means that when you dial digits on a telephone set, for example, they leave the FXO port on the telephone set and are sent to the FXS port connected to the PBX, PSTN, or Cisco voice gateway with FXS modules.

This means that informational signaling is provided by only an FXS port, which emulates the Central Office. FXS ports use call progress (CP) tones, i.e. network progress or alerting tones, to indicate the status of calls. Functions are determined by the frequency of the tone that is sent, as well as by the cadence, i.e. the tone-on and tone-off durations, of that tone.
For example, dial-tone is comprised of two interfering tones between 350 Hz and 440 Hz in the U.S. with the busy signal being 480 Hz and 620 Hz, whereas it comprises a constant single tone of 425 Hz in most of Europe. The default cadence in the US is 2 seconds of ringing tone followed by 4 seconds of silence and ½ a second on and off for a busy signal; however, the European standard uses a double ring followed by 2 seconds of silence.

It is important to remember that CP tones are country specific. Keep this in mind because Cisco voice gateways default to U.S. CP tones, which are not used worldwide. CP tones can be adjusted by using the `cptone [locale]` command under the voice port. The following output illustrates the options available with this configuration command:

```
R1(config)#voice-port 0/0/1
R1(config-voiceport)#cptone ?
locale 2 letter ISO-3166 country code
AR Argentina IS Iceland PE Peru
AU Australia IN India PH Philippines
AT Austria ID Indonesia PL Poland
BE Belgium IE Ireland PT Portugal
BR Brazil IL Israel RU Russian Federation
CA Canada IT Italy SA Saudi Arabia
CN China JP Japan SG Singapore
CO Colombia JO Jordan SK Slovakia
C1 Custom1 KE Kenya SI Slovenia
C2 Custom2 KR Korea Republic ZA South Africa
CY Cyprus LB Lebanon ES Spain
CZ Czech Republic LU Luxembourg SE Sweden
DK Denmark MY Malaysia CH Switzerland
EG Egypt MX Mexico TW Taiwan
FI Finland NP Nepal TH Thailand
FR France NL Netherlands TR Turkey
DE Germany NZ New Zealand GB United Kingdom
GH Ghana NG Nigeria US United States
GR Greece NO Norway VE Venezuela
HK Hong Kong PK Pakistan ZW Zimbabwe
HU Hungary PA Panama
```

The selected tone can then be viewed in the gateway running configuration as follows:

```
R1#show running-config | begin voice-port 0/0/1

voice-port 0/0/1

cptone ZW
```

The `show voice port` command can also be used to view the selected tone as illustrated in the Cisco voice gateway output printed below:

```
R1#show voice port 0/0/1 | include Tone

Region Tone is set for ZW
```
Ear and Mouth (E&M) signaling is not as common in carrier networks as it was in the networks of yesteryear, however, it is still the most prevalent method of analog trunking. E&M signaling was developed to interconnect PBX devices using dedicated circuits from the PSTN. In modern telephony networks, E&M signaling can also be found on back-to-back inter-PBX tie trunks.

E&M is a type of supervisory line signaling that uses Direct Current (DC) signals on separate leads, called the E lead and M lead. Depending on who you are talking to, E&M may mean ear and mouth or even earth and magnet. The ear portion of this moniker represents the electrical ground, and the magnet portion represents the electromagnet used to generate tone in the telephone handset. Another easy way to remember E&M is by using the phrase recEive and transMit, because the E lead is used to receive signals and the M lead is used to transmit signals.

E&M signaling provides states that indicate on-hook and off-hook conditions, which decreases the likelihood that glare will occur. Central Offices use reverse-battery DC signaling among local and remote CO exchanges to indicate switched circuit status. The area CO closest to the party requesting the call selects an idle toll trunk circuit. A polarity change on the trunk indicates to the local CO originating the connection that the called phone is on-hook and ringing, or is off-hook and busy. The distant-end CO completes the operation by reversing the voltage polarity to indicate that the called party has answered.

E&M signaling defines a signaling side and a trunking side for each connection. The PBX is the signaling side and the trunking side may be the CO or voice gateway. The signaling side sends its on-hook and off-hook indicators over the M lead and the trunking side sends its on-hook and off-hook indicators over the E lead. This system effectively provides each individual side of the link with a dedicated signaling path.

Unlike 2-wire FXO and FXS circuits, E&M circuits are 8-wire and use an RJ-48 connection. The voice path uses either 2 or 4 wires and the remaining 4 wires are used for signaling. The table that follows lists and describes the wires, or leads, that are used in E&M circuits:

<table>
<thead>
<tr>
<th>Wire / Lead Description</th>
<th>Pin #</th>
</tr>
</thead>
<tbody>
<tr>
<td>SB -48V Signaling Battery</td>
<td>1</td>
</tr>
<tr>
<td>M Signaling Input</td>
<td>2</td>
</tr>
<tr>
<td>R Ring, Audio Input</td>
<td>3</td>
</tr>
<tr>
<td>R1 Ring, Audio Input/Output</td>
<td>4</td>
</tr>
<tr>
<td>T1 Tip, Audio Input/Output</td>
<td>5</td>
</tr>
<tr>
<td>T Tip, Audio Input</td>
<td>6</td>
</tr>
<tr>
<td>E Signaling Output</td>
<td>7</td>
</tr>
<tr>
<td>SG Signaling Ground</td>
<td>8</td>
</tr>
</tbody>
</table>

Like FXO and FXS, E&M does support both pulse dialing and DTMF for address translation. However, E&M interfaces typically do not use dial-tone but instead use immediate-start, wink-start, or delay-start signaling to indicate an off-hook state or to indicate incoming calls.

Wink-start is the most popular signaling protocol for E&M trunks. The wink-start process is defined as the originating switch going off-hook and then waiting for an off-hook pulse, or wink, from the terminating, the terminating switch sending the wink, and the originating switch sending the digits. After the called party answers the call, the remote switch indicates call answer by transmitting off-hook.

With delay-start, the originating switch waits for a configurable time period before inspecting the incoming signal from the remote switch. If the remote switch indicates an on-hook state, the originating switch sends digits to the remote switch. However, if the received signal is off-hook, the originating switch will wait until the signal returns to on-hook before sending the digits. The remote switch then indicates call answer by transmitting off-hook.
With immediate-start (the most basic of the signaling methods), the originating switch goes off-hook for a certain amount of time (usually less than 200 ms) and then starts sending digits. This method of signaling is seldom implemented. Wink-start signaling is the default signaling method used on Cisco E&M ports. This can be viewed via the `show voice port` command on an E&M port as illustrated in the following output:

R1# `show voice port 1/0/0`

E&M Slot is 1, Sub-unit is 0, Port is 0

**Type of VoicePort is E&M**

-[Truncated Output]-

Voice card specific Info Follows:

**Signal Type is wink-start**

-[Truncated Output]-

There are six different types of E&M signaling: Type I through Type V and addition E&M 5. Cisco platforms do not support type IV; however, all other types are supported.

E&M Type I signaling is the most common signaling type used in the United States and Japan. Battery is provided on both the E lead and the M lead. During inactivity, the E lead is open and the M lead is ground. A PBX indicates an off-hook condition by connecting the M lead to battery and the gateway or CO side indicates an off-hook condition by connecting the E lead to ground. E&M Type I is a six-wire signaling method that uses six leads: E, M, T, R, T1, and R1. Because E&M Type I does not provide ground isolation, it is not suitable for back-to-back configurations; i.e. connecting two Cisco voice gateways back-to-back using E&M Type I.

E&M Type II is also used in the United States, though not nearly in the same capacity as Type I. It operates in the same manner as Type I signaling, with the addition of a ground isolation feature. During inactivity, both leads are open. The PBX indicates an off-hook state by connecting the M lead to the SB lead. The gateway or CO indicates an off-hook state by connecting the E lead to SG. Type II signaling is symmetrical and allows for signaling nodes to be connected back-to-back using a simple crossover cable. E&M Type II uses four leads: E, M, SB, and SG. E&M Type II signaling is commonly used for geographically dispersed equipment where a common ground between the PBX and the CO or gateway is not possible.

E&M Type III signaling is not commonly used in modern systems. During the on-hook state, the E lead is open and the M lead is set to ground by connecting it to the SG lead from the CO. A PBX indicates an off-hook state by disconnecting the M lead from the SG lead and connecting it to the SB lead from the CO. The voice gateway or CO side indicates an off-hook condition by connecting the E lead to ground. E&M Type III uses four leads: E, M, SB, and SG.

E&M Type IV signaling is not supported on Cisco platforms. This signaling method is similar to Type II signaling. During the on-hook state, both the E and M leads are open. The PBX indicates the off-hook state by connecting the M lead to the SB lead connected to ground. The voice
gateway or CO side indicates off-hook by connecting the E lead to the SG connected to the ground. E&M Type IV uses four leads: E, M, SB, and SG.

While E&M Type V signaling may be used in the United States, it is the most common interface type used outside of United States, e.g. in Europe and South America. During inactivity, both the E lead and M lead are in an open state. A PBX indicates an off-hook condition by connecting the M lead to ground. The voice gateway of CO side indicates an off-hook condition by connecting the E lead to ground. Type V signaling is symmetrical and allows for back-to-back connections using a crossover cable. As is the case with Type I E&M signaling, Type V is a six-wire E&M signaling type that uses six leads: E, M, T, R, T1, and R1.

E&M SSDC5 signaling is similar to Type V signaling. However, SSDC5A differs in that on-hook and off-hook states are backward to allow for fail-safe operation. If the line breaks, the interface defaults to off-hook (busy). SSDC5 is most often found in England.

The following table illustrates the operation of the different E&M signaling types:

<table>
<thead>
<tr>
<th>Type</th>
<th>M-Lead Off-hook</th>
<th>M-Lead on-hook</th>
<th>E-Lead Off-hook</th>
<th>E-Lead on-hook</th>
</tr>
</thead>
<tbody>
<tr>
<td>I</td>
<td>Battery</td>
<td>Ground</td>
<td>Ground</td>
<td>Open</td>
</tr>
<tr>
<td>II</td>
<td>Battery</td>
<td>Open</td>
<td>Ground</td>
<td>Open</td>
</tr>
<tr>
<td>III</td>
<td>Loop Current</td>
<td>Ground</td>
<td>Ground</td>
<td>Open</td>
</tr>
<tr>
<td>IV</td>
<td>Ground</td>
<td>Open</td>
<td>Ground</td>
<td>Open</td>
</tr>
<tr>
<td>V</td>
<td>Ground</td>
<td>Open</td>
<td>Ground</td>
<td>Open</td>
</tr>
<tr>
<td>SSDC5</td>
<td>Earth On</td>
<td>Earth Off</td>
<td>Earth On</td>
<td>Earth Off</td>
</tr>
</tbody>
</table>

Digital Circuits

Digital signals are generated and interpreted as electric current, which is measured in volts. The stronger the electrical signal, the higher the voltage. After the signal has been generated, it travels over copper cabling as electrical current; over fiber optic cabling as light pulses (waves); or through the atmosphere as electromagnetic (radio) waves. The following diagram shows a digital signal and illustrates the 1s and 0s in digital communication:
Digital transmission is more robust and of better quality than analog transmission. In order to maximum digital circuit capabilities providers use multiplexing to allow multiple voice, data, and video signals to be carried over the same communications channel. To allow the physical media to carry multiple signals, the media is logically separated into several smaller channels, also commonly referred to as sub-channels.

In order to combine and transmit multiple signals over a single medium a multiplexer (mux) is required at the transmitting end of the channel. At the receiving end, a demultiplexer (demux) is required to separate the combined signals and regenerate them in their original form. Multiplexing allows networks to increase the amount of data that can be transmitted in a given amount of time over a given bandwidth.

There are many different types of multiplexing available and the type that is used depends on the media, transmission and reception that the equipment can handle. However, the two most common forms of multiplexing that are employed are Frequency Division Multiplexing and Time Division Multiplexing.

Frequency Division Multiplexing (FDM) assigns a unique frequency band to each individual communications sub-channel. Signals are modulated with different carrier frequencies and are then multiplexed to simultaneously travel over a single channel. Each signal is then demultiplexed at the receiving end. FDM was first used by telephone companies when they discovered that it allowed them to send multiple voice signals over a single cable. That meant that rather than running separate lines for each residence they could send as many as 24 multiplexed signals over a single neighborhood line. Each signal was then demultiplexed before being brought into the home.

With recent technological advances, telephone companies can use FDM to multiplex signals on the phone line that enters a home. Voice communications use the frequency band of 300 Hz -- 3300 Hz, although the most common representation of this range is 300 Hz -- 3 KHz. Because everything above the 3 KHz range was simply unused space, telephone companies used FDM to allow them to send data signals in this space without interrupting voice communications, allowing for DSL service over existing telephone lines.

In a similar manner to telephone companies, cable operators also use FDM to multiplex signals over a single channel and provide television, voice and data services over cable networks. However, voice calls transmitted over FDM equipment were often noisy as a result of atmospheric noise picked up in the transmission medium, as well as a result of amplifier noise picked up from analog electronics. To solve the problem of noise introduced to voice calls transmitted over analog FDM links, digital TDM was introduced in the mid-1960s. Because TDM transmits and regenerates voice digitally, it is immune to the noise that is picked up in the transmission medium.

TDM works by converting all signals into a digital format. With voice traffic, for example, an analog signal in the range of 300 Hz to 3 KHz bandwidth is converted to a 64 Kbps digital channel. In the United States, twenty-four of these channels are then multiplexed into a single T1 at great speed, which allows for the transmission of all voice and data traffic in a string. T and E carriers are the most widely used digital transmission formats. These carrier types will be described in detail later in this section.

In order to convert analog data into a digital format, the data must be modulated. Modulation is used to modify analog signals to make them more suitable for carrying data over a communication path. In modulation, a simple wave called a carrier wave is combined with the information or data wave to produce a unique signal that gets transmitted from one node to another. The carrier wave contains preset properties, such as the frequency, amplitude, and phase and when combined with the information wave, any one of the carrier wave properties is then modified resulting in a new blended signal that contains the properties of both the carrier wave and the data wave. When the signal reaches its destination, the receiver separates the data (information) from the carrier wave via demodulation.

In 1928, Henry Nyquist developed a method for digitizing analog signals. The Nyquist theorem states that an analog signal waveform can be converted to digital format and be reconstructed without error from samples taken at equal time intervals if the sampling rate is equal to, or greater
than, twice the highest frequency component in the analog signal. Given the fact that maximum frequency is set at 4000 Hz, or 4 KHz, this results in a sampling rate of 8000 times per second (4000 x 2). The Nyquist theorem forms the basis for Pulse Code Modulation (PCM), the fundamental method for converting analog voice to digital format. PCM will be described in detail later in this chapter.

T-Carrier Circuits

Although there are numerous T-carrier circuits, only the T1 will be described in detail in this section and in this guide. Originally, T1s used FDM to carry calls across a single copper loop using a 96 KHz spectrum; however, the T1s of today use TDM and transmit digital signals instead of analog signals.

In voice communications, a single voice channel in digital form requires 64 Kbps of bandwidth, which is calculated by multiplying 8000 samples by 8 bits per sample, resulting in 64000 bps. This 64 Kbps channel is referred to as a DS0 (Digital Signal Zero). T1s are comprised of a total of 24 of these channels, resulting in a bandwidth of 1536 Kbps or 1.536 Mbps. However, because an additional 8 Kbps was added for framing information, the actual speed of a T1 is 1544 Kbps or 1.544 Mbps.

Framing, or frame synchronization, is the process by which incoming frame alignment signals, i.e., distinctive bit sequences, are distinguished from data bits, permitting the data bits within the frame to be extracted for decoding or retransmission. A T1 time slot is an 8-bit segment for each DS0. A frame consists of 24 time slots (one for each DS0 in the T1) and an additional framing bit, for a total of 193 bits.

The framing bit does not contain any user data; however, it is used by the transmitting and receiving equipment for the actual synchronization of the incoming data with the receiver. Within a bit stream, framing bits are used to determine the beginning or end of a frame. They occur at specified positions in the frame, do not carry information, and are commonly repetitive. If the transmission is temporarily interrupted, or a bit slip event occurs, the receiver must re-synchronize. T1s use two types of framing: Super Frame (SF) and Extended Super Frame (ESF), with the latter being the most prevalent method implemented.

Super Frame (SF) is an older framing standard for T1s that is also referred to as D4 or even D3/D4 framing. In order to determine where each channel is located in the stream of data being received, each set of 24 channels is aligned in a frame. The frame is 192 bits long and is terminated with the framing bit (the 193rd bit), which is used to find the end of the frame. To ensure that the framing bit will be located by receiving equipment, a specific pattern is sent on this bit. Equipment will search for a bit which has the correct pattern, and will align its framing based on that bit. The pattern sent is 12 bits long, so every group of 12 frames is called a Super Frame. The pattern used in the 193rd bit is 1000 1101 1100.

For the most part, Super Frame has all been replaced by Extended Super Frame, or ESF. Extended Super Frame, sometimes called D5 framing, doubles the number of the frames in the SF from 12 to 24. This framing method includes a Cyclic Redundancy Check, or CRC, and a 4 Kbps channel capacity for a data link channel, which is used to pass out-of-band information between equipment. It requires less frequent synchronization than the earlier SF format, and provides online, real-time testing of circuit capability and operating condition.

E-Carrier Circuits

Outside of the United States, E-carrier circuits are used to carry digital transmissions. The most common E-carrier circuit is the E1, which has its electrical interference and framing defined in ITU-T G.703 and G.704. E1 time-slots are numbered TS0 to TS31, where TS1 through TS15 and TS17 through TS31 are used to carry voice, which is encoded with PCM, or to carry 64 kbps data. Of these 32 timeslots, 1 is used for framing, 1 is used for telephony signaling, and 30 are used for voice and data transport for an aggregate capacity of 2048 Kbps or 2.048 Mbps.

Timeslot 0 is used for framing, alarming, and international bits; while timeslot 16 is used to carry signaling information, which may be CAS, ISDN, SS7, or proprietary signaling. CAS, ISDN, and
SS7 signaling will be described later in this chapter. There is a 16-frame multi-frame structure that allows a single 8-bit time slot to handle the line signaling for all 30 data channels.

The E1 standard also allows for optional CRC-4 error-checking and is most commonly used in Europe, parts of Asia, South and Central America and Africa.

Call Control Signaling

There are several forms of communication signaling that you should be aware of. This section describes the following call control signaling methods:

- Channel Associated Signaling
- Common Channel Signaling
- E&M Signaling
- R2 Signaling
- Signaling System 7

Channel Associated Signaling (CAS) was the only signaling system used until the mid-1970s.

The biggest disadvantage of CAS signaling is its use of user bandwidth to perform signaling functions (i.e. in-band signaling). Therefore, CAS signaling is often referred to as robbed-bit-signaling because user bandwidth is being 'robbed' by the network for other purposes. Robbed bit signaling allows the Least Significant Bit (LSB) in some timeslots to transmit signaling information -- which is the 6th and 12th frame for Super Frame (SF) and the 6th, 12th, 18th, and 24th frame for Extended Super Frame (ESF).

The most common forms of signaling on CAS circuits are the same as those used on analog circuits; i.e. loop-start, ground-start, and E&M signaling. Although CAS is still used today, primarily in underdeveloped countries, it is being replaced by Common Channel Signaling (CCS). The two main characteristics of CAS that differentiate it from CCS are:

1. CAS signaling is mainly analog
2. CAS signals are carried in the same actual circuit that is used to carry voice (in-band)

However, it is important to remember that in addition to receiving and placing calls, CAS signaling also processes the receipt of DNIS and ANI information, which is used to support authentication and other functions. This is one of the primary reasons it is still in use.

Common Channel Signaling (CCS) was developed as an alternative form of call control signaling for trunks. In CCS, the individual trunks do not carry signaling information and the channel used for CCS signaling does not carry user data. This concept is referred to as out-of-band signaling. In out-of-band (OOB) signaling, signaling bits are sent in special order in a dedicated signaling frame. OOB signaling has the advantage in that signaling and control information does not directly compete with payload (data) for bandwidth. The most common CCS signaling methods in use today are ISDN and SS7. ISDN signaling is used primarily on trunks connecting end-user PBX systems to a central office. SS7 is primarily used within the PSTN. Both methods will be described in this chapter.

E&M signaling is the preferred signaling method when using T1 or E1 channels for compressed voice calls and for defining the CAS method by which the voice gateway connects to the PBX or PSTN. Both network and user transmit all 0s for on-hook and all 1s for off-hook. Cisco voice gateways support the same E&M signaling supported by analog circuits, which includes delay-start, wink-start, and immediate-start. Cisco voice gateways support Feature Group signaling in conjunction with other E&M signaling types.
A Feature Group is United States telephone industry jargon for four types of access to long distance service. They defined switching arrangements between Local Exchange Carriers (LECs) Central Offices (COs) to Interexchange Carrier (IXC). Feature Groups are also referred to as Access Feature Groups. Currently, there are four common feature categories: FGA, FGB, FGC, FGD, and the most commonly implemented standards on Cisco voice gateways are FGB and FGD. FGA offers access to the Local Exchange carrier network through a subscriber line rather than a trunk line, while FGC is used almost exclusively by AT&T.

Feature Group B (FGB) is the arrangement that is associated with 950-xxxx calling; the user enters 950 and 4 additional digits, followed by the long-distance number. E&M wink-start (Feature Group B) notifies the remote side that it can send DNIS information.

Feature Group D (FGD) is a local exchange carrier network service that allows public-safety dispatch offices to receive a 10-digit data stream, including the full call-back number, alongside wireless 911 calls. FGD is offered as a way by which wireless carriers can meet FCC enhanced 911 rules and dispatch offices can overcome their current bandwidth limits. FGD is associated with equal access arrangements, which allows the end user to have the same dialing plan, i.e. 1 plus the telephone number, to reach any long distance phone companies of their choice.

Wink-start with wink-acknowledge or double-wink (Feature Group D) uses a second wink that is sent to acknowledge the receipt of the DNIS information. FGD can accept ANI and DNIS for inbound calls, i.e. from the telephone network to Customer Premise Equipment (CPE), which may be a Cisco voice gateway, for example, but can only provide DNIS for outward calls from the voice gateway or PBX to the telephone network.

NOTE: In order to provide and accept both ANI and DNIS at the same time using CAS, the T1 must be split into two different DS0 groups. One group is for inbound calls and one group is for outbound calls. T1 DS0 group configuration will be described in detail later in this guide.

Cisco voice gateways also support FGD-EANA, which stands for Feature Group-D (FGD) of type Exchange Access North American (EANA). EANA is a type signaling protocol that provides certain call services, such as emergency (USA-911) calls. FGD-EANA can also provide ANI and DNIS for outbound calls via the calling-number outbound dial-peer configuration command. This is not used with E&M signaling and this configuration is beyond the scope of the CVOICE. Although FGD-EANA can provide ANI for outbound calls, it is important to know that it cannot accept ANI for inbound calls.

R2 signaling specifications are contained in ITU-T Recommendations Q.400 through Q.490. The physical connection layer for R2 is usually an E1 interface that conforms to ITU-T standard G.704. E1 R2 uses two types of signaling: line signaling for supervisory signals, and inter-register signaling for call setup control signals.

E1 R2 line signaling involves supervisory information, such as on-hook and off-hook, and is used for call setup and teardown. R2 uses CAS carried in timeslot 0 of the E1 frame. The supported methods of E1 R2 line signaling are:

1. R2-Analog--R2 line signaling type ITU-U Q.411, typically used for carrier systems
2. R2-Digital--R2 line signaling type ITU-U Q.421, typically used for PCM systems
3. R2-Pulse--R2 line signaling type ITU-U Supplement 7, typically used for satellite links

E1 R2 inter-register signaling is used between switches for informational and address messages. Inter-register signaling is generally performed end-to-end by a compelled procedure. This means that tones in one direction are acknowledged by a tone in the other direction. This type of signaling is known as Multifrequency Compelled (MFC) signaling. There are three types of inter-register signaling:

1. R2-Compelled--When a tone-pair is sent from the switch, the tones stay on until the remote end responds with a pair of tones that signals the switch to turn off the tones. The tones are compelled to stay on until turned off.
2. R2-Non-Compelled--The tone-pairs are sent as pulses, so they stay on for a short duration.
Responses to the switch are sent as pulses.
3. R2-Semi-Compelled--Forward tone-pairs are sent as compelled. Responses to the switch are sent as pulses. This scenario is the same as compelled, except that the backward signals are pulsed instead of continuous.

Signing System 7 (SS7) is an out-of-band signaling protocol that uses separate data links to support packet signaling between switches and databases for network services, such as the toll-free 800 services for example. This also provides SS7 with the ability to deliver call-related information, such as Calling Party Identification (CPID), which is telephone number of the calling party. SS7 allows calls to be set up and torn down faster than with CAS, thus utilizing the network more efficiently. In conclusion, the SS7 standard defines the protocol by which network elements in the PSTN exchange information over a digital signaling network to affect wireless and wire-line call setup, routing, and control.

**Integrated Services Digital Network (ISDN)**

Integrated Services Digital Network (ISDN) technology is standardized according to the International Telecommunications Union (ITU). These recommendations describe the protocols and architecture to implement a worldwide digital network. ISDN uses the same telephone numbering plan that is used on the traditional PSTN. The telephone network numbering plan is administered by the ITU, and the addressing plan is referred to as ITU Recommendation E.164.

E.164 is an international numbering plan for public telephone systems. In the E.164 addressing scheme, each assigned number contains a Country Code (CC) and a National (Significant) Number (N(S)N). The N(S)N is comprised of the National Destination Code (NDC), and a Subscriber Number (SN). This format is illustrated in the following diagram:

<table>
<thead>
<tr>
<th>Country Code</th>
<th>National (Significant) Number</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td>National Destination Code</td>
</tr>
<tr>
<td></td>
<td>Subscriber Number</td>
</tr>
</tbody>
</table>

The Country Code is a one, two, or three digit number that is used to represent the specific country or region. The Country Code of the United States is 1, and the Country Code of the United Kingdom is 44, for example.

The National (Significant) Number is the number used to select the destination subscriber. In essence, this is the subscriber code or number for an end user. As illustrated in the diagram above, there can be up to 15 digits in an E.164 number and the N(S)N is comprised of the National Destination Code and the Subscriber Number.

The National Destination Code is a variable length number that contains the Destination Network (DN) or Trunk Codes (TC) which indicate how the call or calls should be routed. Hence, the National Destination Code is commonly referred to as the Area Code. The Subscriber Number is a variable length number that is actually assigned to the subscriber.

The two basic types of ISDN service are Basic Rate Interface and Primary Rate Interface. Both of these ISDN types use out-of-band signaling, as is the case with CCS. With BRI, voice and data is carried in two 64 Kbps bearer (B) channels, and signaling information is carried in a separate data (D) channel. BRI is commonly therefore referred to as 2B+D, indicating the two bearer channels and the single data channel. BRI provides a data throughput of 128 Kbps, and a total throughput of 144 Kbps: 128 Kbps for bearer channels and 16 Kbps for the data channel.

E1 PRI service is commonly referred to as 30B+D. While T1 PRI will allow for 23 voice calls on a single circuit, E1 PRI allows for 30 voice calls on a single circuit.