Voice Network Signaling and Control

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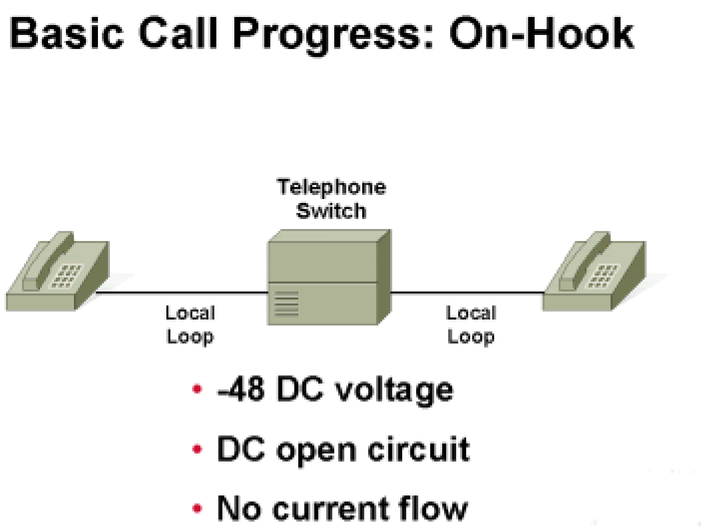
Introduction

This paper discusses the signaling techniques required to control voice transmission. These signaling techniques can be placed into one of three categories; supervision, addressing, or alerting. Supervision involves the detection of changes to the status of a loop or trunk. Once these changes are detected, the supervisory circuit will generate a predetermined response such as closing a circuit (loop) to connect a call. Addressing involves the passing of dialed digits (pulsed or tone) to a private branch exchange (PBX) or central office (CO). These dialed digits provide the switch with a connection path to another phone or customer premises equipment (CPE). Alerting provides audible tones to the user, indicating certain conditions such as an incoming call or a busy phone. A phone call cannot take place without all of these signaling techniques. Before discussing specific signaling types within each category, basic call progress is examined from call origination to termination.

Basic Call Progress

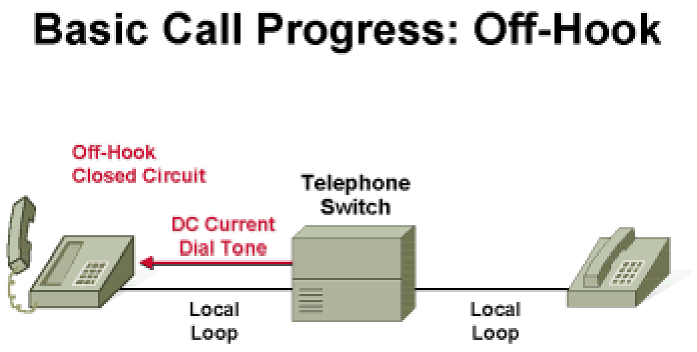
The progress of a telephone call with loop-start signaling in place can be divided into five phases; on hook, off hook, dialing, switching, ringing, and talking. Figure 1 shows the on-hook phase.

Figure 1



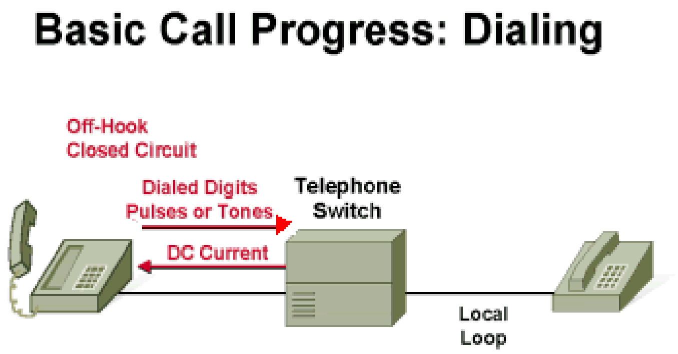
Before a phone call is initiated, the telephone set is in a ready condition waiting for a caller to pick up its handset. This state is called on hook. In this state the 48-VDC circuit from the telephone set to the CO switch is open. The CO switch contains the power supply for this DC circuit. Having the power supply located at the CO switch prevents a loss of telephone service when the power goes out at the location of the telephone set. Figure 2 shows the off-hook phase.

Figure 2



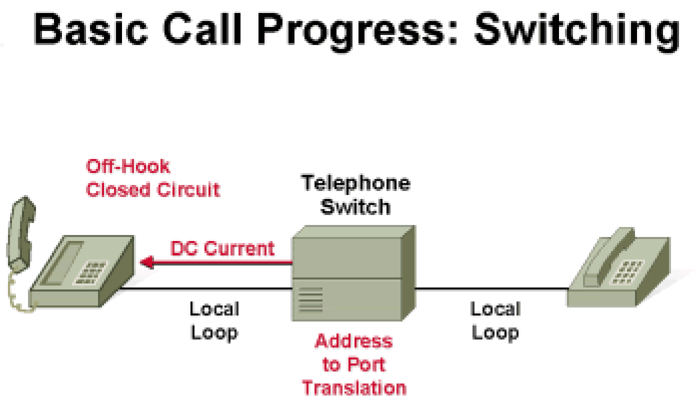
The off-hook phase occurs when the telephone customer decides to make a phone call and lifts the handset off the switch hook of the telephone set. The switch hook closes the loop between the CO switch and the telephone set and allows current to flow. The CO switch detects this current flow and transmits a dial tone (350- and 440-hertz [Hz] tones played continuously) to the telephone set. This dial tone lets the customer know that dialing can start. There is no guarantee that the customer will get dial tone right away. If all the circuits are being used, the customer might have to wait for a dial tone. The access capacity of the CO switch being used determines how soon a dial tone will be sent to the caller's phone. The CO switch generates a dial tone only after it has reserved registers to store the incoming address. Therefore, the customer can't dial until a dial tone is received. If there is no dial tone, then the registers are not available. Figure 3 shows the dialing phase.

Figure 3



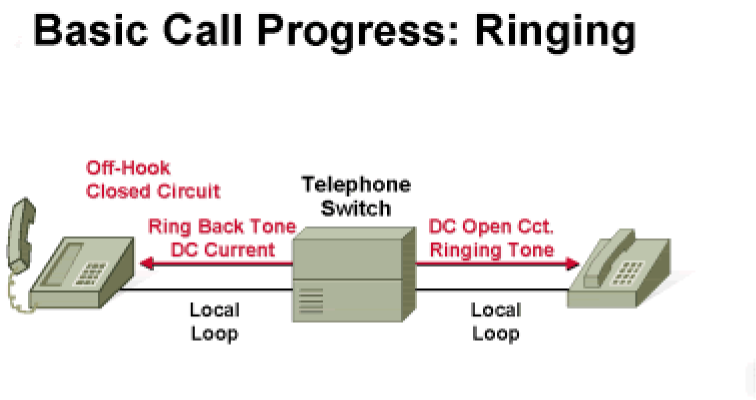
The dialing phase allows the customer to enter a phone number (address) of a telephone at another location. The customer enters this number using either a rotary phone that generates pulses or a touch-tone phone that generates tones. These pulses or tones are transmitted to the CO switch across a two-wire twisted-pair cable (tip and ring lines). Figure 4 shows the switching phase.

Figure 4



In the switching phase, the CO switch translates the pulses or tones into a port address that connects to a telephone set of the called party. This connection could go directly to the requested telephone set (for local calls) or go through another switch or several switches (for long-distance calls) before reaching its final destination. Figure 5 shows the ringing phase.

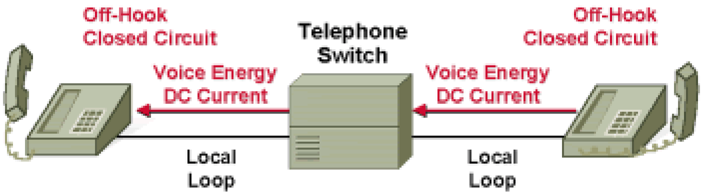
Figure 5



Once the CO switch connects to the called line, it sends a 20-Hz 90V signal to this line. This signal rings the phone of the called party. While ringing the called party's phone, the CO switch sends an audible ring-back tone to the caller. This ringback lets the caller know that ringing is taking place at the called party's phone. The CO switch generates a ringback by transmitting 440 and 480 tones to the caller's phone. These tones are played for a specific on time and off time. If the called party's phone *is* busy, the CO switch sends a busy signal to the caller. This busy signal consists of 480- and 620-Hz tones. Figure 6 shows the talking phase.

Figure 6

Basic Call Progress: Talking



In the talking phase, the called party hears the phone ringing 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's 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's phone. Now voice communication can start between both ends of this connection.

Table 1 shows a summary of alerting tones that could be generated by the CO switch during a phone call.

Table 1

Network Call Progress Tones

Tone	Frequency (Hz)	On Time	Off Time
Dial	350 + 440	Continuous	
Busy	480 + 620	0.5	0.5
Ringback, Normal	440 + 480	2	4
Ringback, PBX	440 + 480	1	3
Congestion (Toll)	480 + 620	0.2	0.3
Reorder (Local)	480 + 620	0.3	0.2
Receiver Off-hook	1400 + 2060 + 2450 +2600	0.1	0.1
No Such Number	200 to 400	Continuous, Freq. Mod 1Hz	

The progress tones in Table 1 are for North American phone systems. International phone systems might have a totally different set of progress tones. Everyone should be familiar with most of these call progress tones, especially dial, busy, and ringback. Congestion progress tone is used between switches. Receiver off hook is the loud ringing that occurs when the phone is left off the hook for an extended period of time. No such number occurs when the number dialed can't be found in the routing table of a switch.

Address Signaling and Tip and Ring

Address Signaling

North American Numbering Plan

The North American Numbering Plan (NANP) uses ten digits to represent a telephone number. These 10 digits are divided into three parts: the area code, office code, and station code.

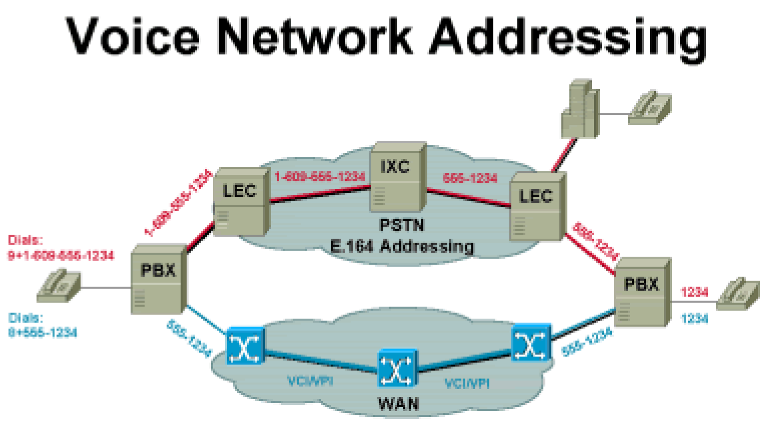
In the original NANP, the area code consisted of the first three digits of the telephone number; it represented a region in North America (including Canada). The first digit was any number from 2 to 9, the second digit was 1 or 0, and the third digit was any number from 0 to 9. The office code consisted of the second three digits of the telephone number; it uniquely identified a switch in the telephone network. The first digit was any number from 2 to 9, the second digit was any number from 2 to 9, and the third digit was any number from 0 to 9. The area code and office code could never be the same because the second digit of each code was always different. With this numbering system, the switch was able to determine whether this was a local call or long-distance call by looking at the second digit of the area code. The station code consisted of the last four digits in the telephone number. This number uniquely identified a port within the switch that was connected to the telephone being called. Based on this 10-digit numbering system, an office code could have up to 10,000 different station codes. In order for a switch to have more than 10,000 connections, it would have to have more office codes assigned to it.

An increase in the number of phone lines installed in homes, Internet access, and fax machine usage dramatically reduced the number of phone numbers available. This scenario prompted a change in the NANP. The present plan is basically the same as the old plan except for the area code and office code sections of the telephone number. The three digits for the area code and office code are now selected in the same fashion. The first digit can be any number from 2 to 9, and the second and third digits can be any number from 0 to 9. This scenario dramatically increases the number of area codes available, one in turn increasing the number of station codes that can be assigned. If the call is a long-distance number, a one must be dialed before the 10-digit number.

International Numbering Plan

The International Numbering Plan is based on ITU-T specification E.164, an international standard that all countries must follow. This plan states that the telephone number in every country cannot be greater than 15 digits. The first three digits represent the country code, and the remaining 12 digits represent the national specific number. For example, the country code for North America is 011. Therefore, when calling North America from another country, 1 must be dialed first to access the NANP. Then the 10 digits required by the NANP are dialed. The 12 digits of the national specific number can be organized in any manner deemed appropriate by the specific country. Figure 7 illustrates network addressing in North America.

Figure 7



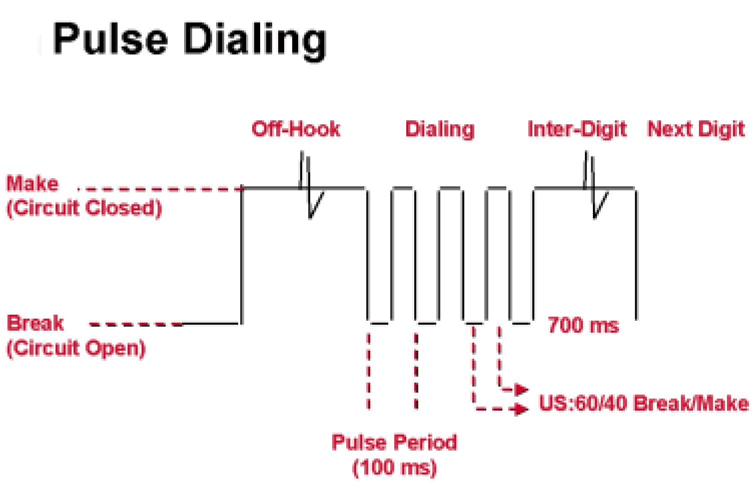
In this figure, the caller is generating a call from within a customer premise that uses a PBX to access the Public

Switched Telephone Network (PSTN). To get past the PBX, the caller must dial 9 first (this is how most PBXs are set up). Then, the caller must dial 1 for long distance and the 10-digit number of the telephone the caller is trying to reach. The area code takes the caller through two switches, first a local switch and then an inter-exchange carrier (IXC) switch, taking the call long distance. The office code (second three digits) takes the caller through a local switch again, and then to another PBX. Finally, the station code (last four digits) takes the caller to the telephone being called.

Pulse Dialing

Pulse Dialing is an in-band signaling technique. It is used in analog telephones that have a rotary dialing switch. Each time the dial is turned, the bottom of the dial closes and opens the circuit leading to the CO switch or PBX switch. This dial closes and opens the switch in the same way the switch hook does when the phone is off hook or on hook. The number of digits dialed corresponds to the number of times the switch is closed and opened. Therefore, if the digit 3 is dialed, the switch will be closed and opened three times. Figure 8 represents the sequence of pulses that occur when dialing a digit 3 using pulse dialing.

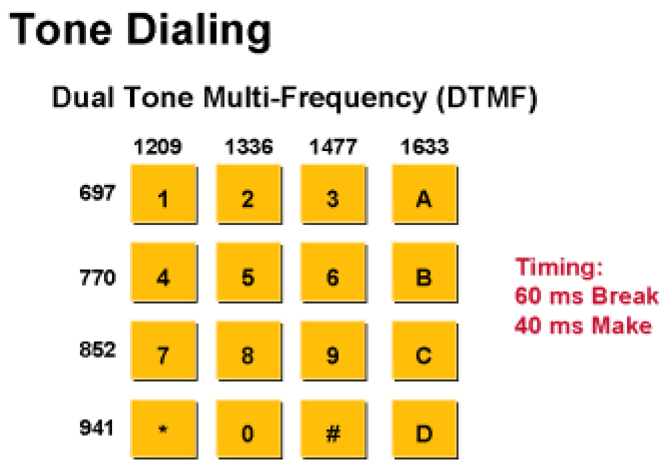
Figure 8



This illustration displays two new terms, make and break. Make is when a circuit is closed and break is when a circuit is opened. 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, generating sequences of makes and breaks that occur every 100 milliseconds (ms). Typically a break will last 60 ms and a make will last 40 ms. Then the phone stays in a make state until another digit is dialed or the phone is put back to an on-hook (equivalent to a break) state. Dial pulse addressing is a very slow process because the number of pulses generated equates to the digit being dialed. So, dialing digit 9 generates nine make and break pulses. Dialing digit 0 generates ten make and break pulses. To increase the speed of dialing, a new dialing techique called dual tone multifrequency (DTMF) was developed. Figure 9 shows the frequency tones generated by DTMF dialing (also called touch-tone dialing).

DTMF Dialing

Figure 9



DTMF dialing is an in-band signaling technique just like pulse dialing. This technique is used in analog telephone sets that have a touch-tone pad. This dialing technique uses only two frequency tones per digit, as shown in Figure 9. Therefore, when dialing digit 0, only frequency tones 941 and 1336 are generated instead of the ten make and break pulses generated by pulse dialing. The timing is still 60-ms break and 40-ms make for each frequency generated. These frequencies were selected for DTMF dialing based on their insusceptibility to normal background noise.

Single-Frequency and Multifrequency Signaling

R1 and R2 signaling standards are used to transmit supervisory and address signaling information between voice network switches. They both use single-frequency signaling for transmission of supervisory information and multifrequency signaling for addressing information.

R2 Signaling

R2 signaling specifications are contained in ITU-T Recommendations Q.400 through Q.490. The physical connection layer for R2 is usually an E1 (2.048 megabits per second [Mbps]) interface that conforms to ITU-T standard G.704. This interface uses time slot 0 for synchronization and framing (same as for Primary Rate Interface [PRI]) and uses time slot 16 for ABCD signaling. There is a 16-frame multiframe structure that allows a single 8-bit time slot to handle the line signaling for all 30 data channels.

R2 Call Control and Signaling

Two types of signaling are involved: line signaling and register signaling. Line signaling involves supervisory information (on hook/off hook) and register signaling deals with addressing. These are described in more detail below.

R2 Line Signaling

R2 uses channel-associated signaling (CAS), meaning that, in the case of E1, one of the time slots (channels) is dedicated to signaling, as opposed to the signaling used for T1, which uses the top bit of every time slot in every sixth frame.

This signaling is out-of-band signaling and uses ABCD bits in a similar manner to T1 robbed-bit signaling to indicate on-hook/off-hook status. These ABCD bits appear in time slot 16 in each of the 16 frames that make up a multiframe. Of these four bits, sometimes known as signaling channels, only two (A and B) are actually used in R2 signaling; the other two are spare.

In contrast to robbed-bit signaling types such as wink start, these two bits have different meanings in the forward and backward directions. However, there are no variants on the basic signaling protocol.

R2 Register Signaling

The transfer of call information (called/calling numbers, and so on) is done using tones in the time slot being used for the call—called in-band signaling.

R2 uses six signaling frequencies in the forward direction (from the party initiating the call) and a different six frequencies in the backward direction (from the party answering the call). These inter-register signals are of the multifrequency type using a two-out-of-six in-band code. Variations on R2 signaling that use only five of the six frequencies are known as decadic CAS systems.

Inter-register signaling is generally performed end to end by a compelled procedure, meaning 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.

Features that may be signaled include:

- Called/calling party number
- Call type (transit, maintenance, and so on)
- Echo-suppressor signals
- Calling party category
- Status

R1 Signaling

R1 signaling specifications are contained in ITU-T Recommendations Q.310 through Q.331. A summary of the main points follows. The physical connection layer for R1 is usually a T1 (1.544-Mbps) interface that conforms to ITU-T standard G.704. This standard uses the 193rd bit of the frame for synchronization and framing (same as T1).

R1 Call Control and Signaling

Again two types of signaling are involved: line signaling and register signaling. Line signaling involves supervisory information (on hook/off hook) and register signaling deals with addressing. They are described in more detail below.

R1 Line Signaling

R1 uses in-slot CAS by bit robbing the eighth bit of each channel every sixth frame. This type of signaling uses ABCD bits in an identical manner to T1 robbed-bit signaling to indicate on-hook/off-hook status.

R1 Register Signaling

The transfer of call information (called/calling numbers, and so on) is done using tones in the time slot being used for the call. This type of signaling is also called in-band signaling.

R1 uses six signaling frequencies that are 700 to 1700 Hz in 200-Hz steps. These inter-register signals are of the multifrequency type using a two-out-of-six in-band code. The address information contained in the register signaling is preceded by a KP tone (start-of-pulsing signal) and terminated by a ST tone (end-of-pulsing signal).

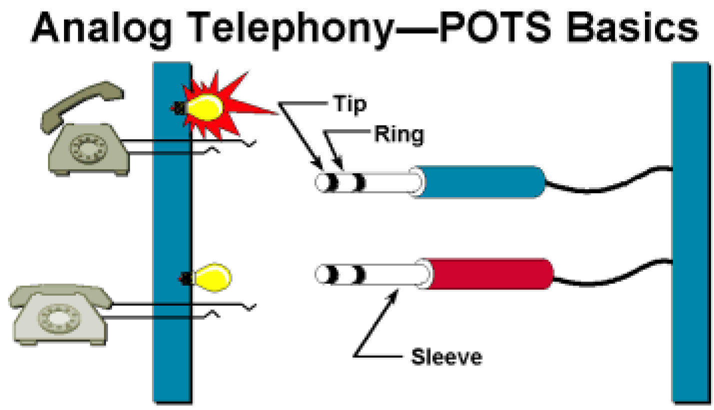
Features that may be signaled include:

- Called-party number
- Call status

Tip and Ring Lines

Figure 10 illustrates tip and ring lines in a plane old telephone service (POTS) network.

Figure 10



The standard way to transport voice between two telephone sets is to use tip and ring lines. Tip and ring lines are the twisted pair of wires that connect to your phone via an RJ-11 connector. The sleeve is the ground lead for this RJ-11 connector.

Loop Start Signaling

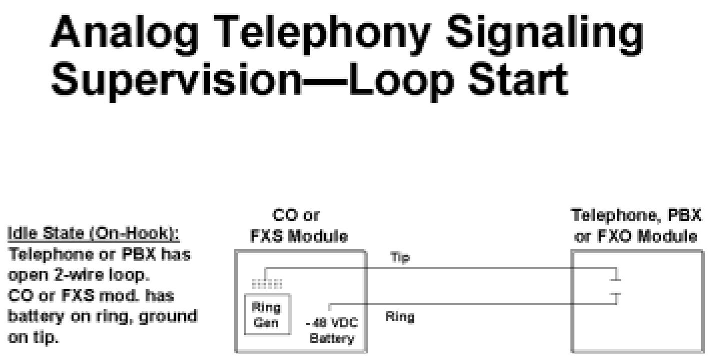
Loop-start signaling is a supervisory signaling technique that provides a way to indicate on-hook and off-hook conditions in a voice network. Loop-start signaling is used primarily when connecting from the telephone set to a switch. This signaling technique can be used in any of the following connections:

- Telephone set to CO switch
- Telephone set to PBX switch
- Telephone set to foreign exchange station (FXS) module (interface)
- PBX switch to CO switch
- PBX switch to FXS module (interface)
- PBX switch to foreign exchange office (FXO) module (interface)
- FXS module to FXO module

Analog Loop-Start Signaling

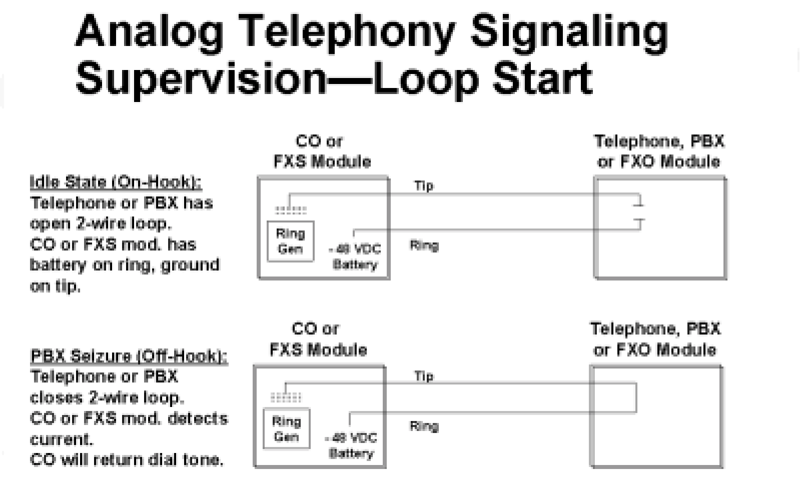
Figures 11 through 13 illustrate loop-start signaling from a telephone set, PBX switch, or FXO module to a CO switch or FXS module. Figure 11 shows the idle state for loop-start signaling.

Figure 11



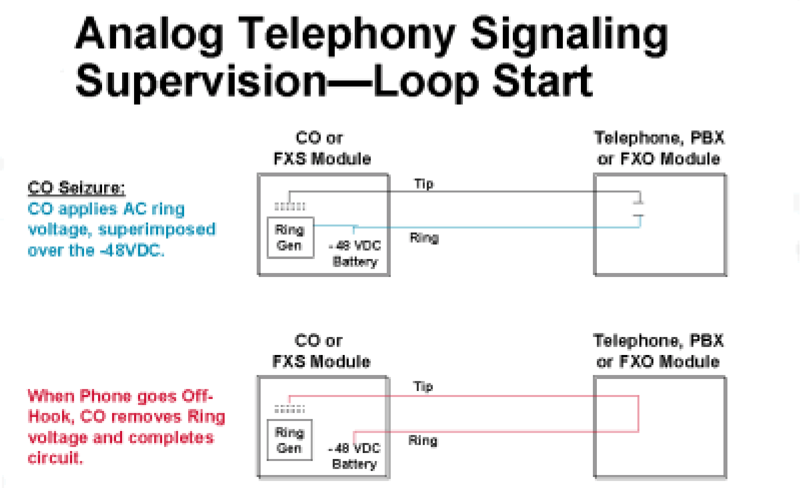
In this idle state, the telephone, PBX, or FXO module has an open two-wire loop (tip and ring lines open). It could be a telephone set with the handset on hook, or a PBX or FXO module generating an open between the tip and ring lines. The CO or FXS are waiting for a closed loop that would generate a current flow. The CO or FXS have a ring generator connected to the tip line and –48VDC on the ring line. Figure 12 shows an off-hook state for a telephone set or a line seizure for a PBX or FXO module.

Figure 12



In this illustration, a telephone set, PBX, or FXO module closes the loop between the tip and ring lines. The telephone takes its handset off hook or the PBX or FXO module closes a circuit connection. The CO or FXS module detects current flow and then generates a dial tone, which is sent to the telephone set, PBX, or FXO module, to indicate that dialing can commence. What happens when there is an incoming call from the CO switch or FXS module? Figure 13 shows this situation.

Figure 13



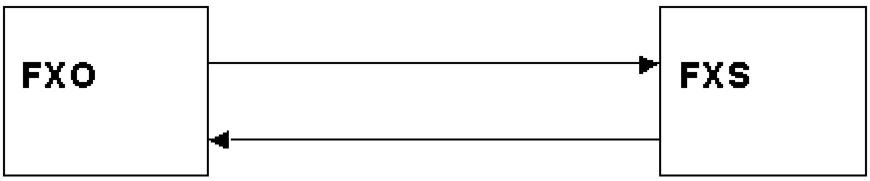
In the illustration, the CO or FXS module seizes the ring line of the telephone, PBX, or FXO module being called by superimposing a 20-Hz, 90-VAC signal over the –48VDC ring line. This procedure rings the called party's telephone set or signals the PBX or FXS module that there is an incoming call. The CO or FXS module removes this ring once the telephone set, PBX, or FXO module closes the circuit between the tip and ring lines. The telephone set closes the circuit when the called party picks up the handset. The PBX or FXS module closes the circuit when it has an available resource to connect to the called party. The 20-Hz ringing signal generated by the CO switch is independent of the user lines and is the only way to let a user know that there is an incoming call. The user lines do not have a dedicated ring generator. Therefore, the CO switch must cycle through all the lines it must ring. This cycle takes about four seconds. This delay in ringing a phone causes a problem, known as glare, when the CO switch and the telephone set, PBX, or FXO module seize a line simultaneously. When this happens, the person calling is connected to the called party almost instantaneously, with no ring-back tone. Glare is not a major problem from the telephone set to the CO switch because an occasional glare situation can be tolerated by the user. Glare becomes a major problem, however, when using loop start from the PBX or FXO module to the CO switch or FXS module because more call traffic is involved; therefore, the chances of glare occurring increases. This scenario explains why loop-start signaling is used primarily when connecting from the telephone set to a switch. The best way to prevent glare is to use ground-start signaling, which is covered in a later section.

Digital Loop-Start Signaling

This mode of operation is most commonly used in off-premise extension (OPX) applications. This is a two-state signaling scheme, using the "B bit" for signaling.

Idle condition:

To FXS: A bit = 0, B bit = 1

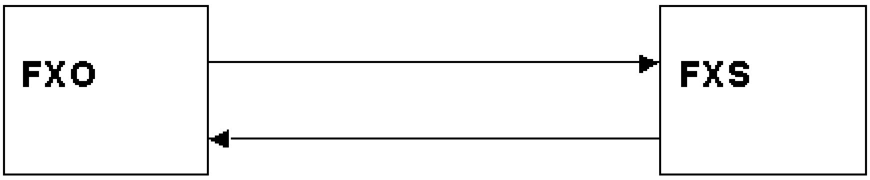


From FXS: A bit = 0, B bit = 1

FXS originates:

Step 1 : FXS changes A bit to 1, signaling the FXO to close the loop.

To FXS: A bit = 0, B bit = 1



From FXS: A bit = 1, B bit = 1

FXO originates:

Step 1 : FXO sets the B bit to 0. The B bit toggles with the ring generation:

To FXS: A bit = 0, B bit = 1

From FXS: A bit = 0, B bit = 1

Step 2 : FXS responds by changing the A bit to 1. The FXO trips the ring generator by setting the B bit to 1:

To FXS: A bit = 0, B bit = 1

From FXS: A bit = 1, B bit = 1

Loop-Start Testing

Testing the signaling states of a loop-start trunk is discussed with reference to two viewpoints: from the demarc looking toward the CO and from the demarc looking toward the PBX.

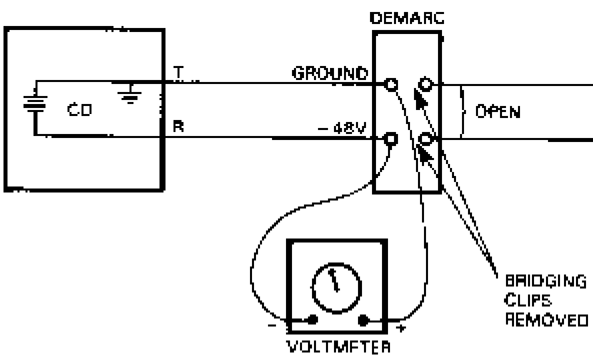
Idle Condition (on hook, initial state)

The idle condition is represented in Figure 14. The bridging clips have been removed to isolate the CO from the PBX.

Looking toward the PBX, an open condition is observed between the T-R leads at the demarc.

Looking toward the CO from the demarc, ground is observed on the T lead and –48V is observed on the R lead. A voltmeter connected between T and R on the CO side of the demarc should read close to –48V.

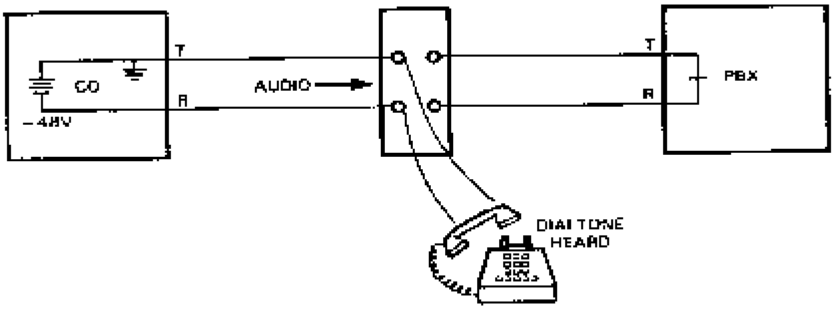
Figure 14: Idle Condition



Outgoing (off hook)

Testing the operation toward the CO is done by removing the bridging clips and attaching a test telephone set across the T-R leads toward the CO. The test set provides loop closure. The CO detects the loop closure, attaches a digit receiver to the circuit, establishes an audio path, and transmits dial tone toward the PBX. (See Figure 15.)

Figure 15



Once dial tone is received by the test telephone, dialing may proceed using either DTMF or dial-pulse signaling as allowed by the CO. Some COs are equipped to receive only dial-pulse addressing. Those equipped to receive DTMF can also receive dial pulse. Upon receiving the first dialed digit, the CO removes dial tone.

After all digits have been dialed, the digit receiver is removed at the CO and the call is routed to the distant station or switch. The audio path is extended over the outgoing facility, and audible call-progress tones are returned to the test telephone. Once the call is answered, voice signals may be heard over the audio path.

Incoming (ringing at destination)

A test telephone at the demarc can also be used for testing loop-start trunks for incoming call operation. The test setup is the same as for outgoing calls. Typically the PBX technician calls a CO technician on another line and asks the CO technician to call the PBX on the trunk under test. The CO applies ringing voltage to the trunk. The test phone at the demarc should ring. The PBX technician answers the call on the test phone. If the technicians can talk to each other over the trunk under test, the trunk is functioning normally.

Testing between the PBX and the demarc with bridging clips removed is difficult. The loop-start interface circuits in most PBXs require battery voltage from the CO for their operation. If the voltage is not present, the trunk cannot be selected for outgoing calls. The usual procedure is to test the trunk from the demarc to the CO, first with the bridging clips removed as described above, and then after installing the bridging clips. If the trunk fails to function properly when connected to the PBX, the problem can be assumed to be in the PBX or in the wiring between the PBX and the demarc.

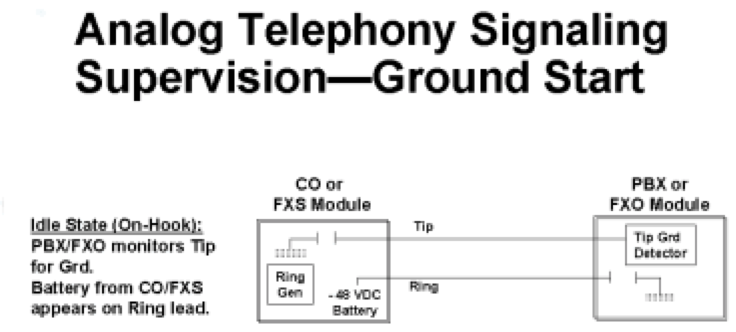
Ground Start Signaling

Ground-start signaling is another supervisory signaling technique, like loop-start, that provides a way to indicate on-hook and off-hook conditions in a voice network. Ground-start signaling is used primarily in switch-to-switch connections. The main difference between ground-start and loop-start signaling is that ground-start requires ground detection to occur in both ends of a connection before the tip and ring loop can be closed.

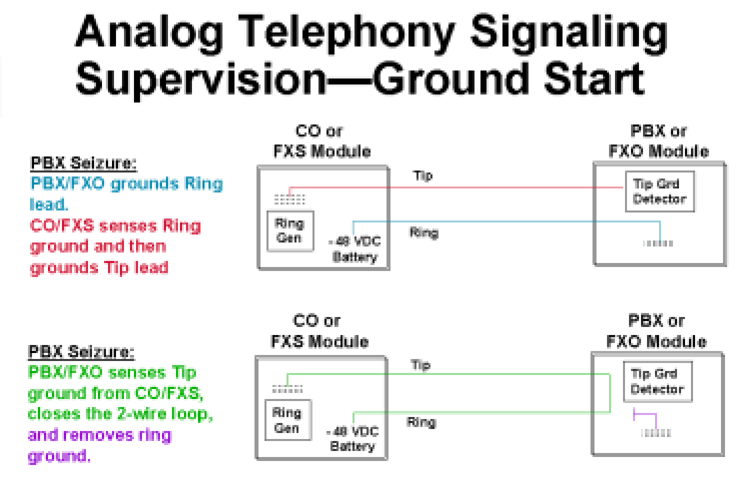
Analog Ground-Start Signaling

Figures 16 through 19 cover ground-start signaling only from a CO switch or FXS module to a PBX or FXO module. Figure 16 shows the idle (on-hook) condition of ground-start signaling.

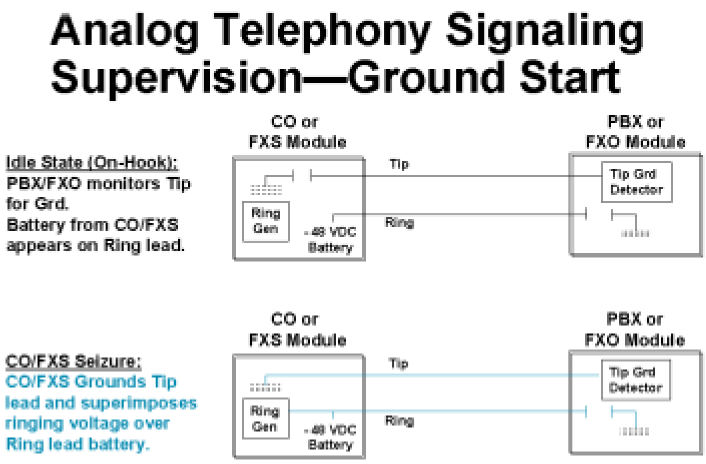
Figure 16



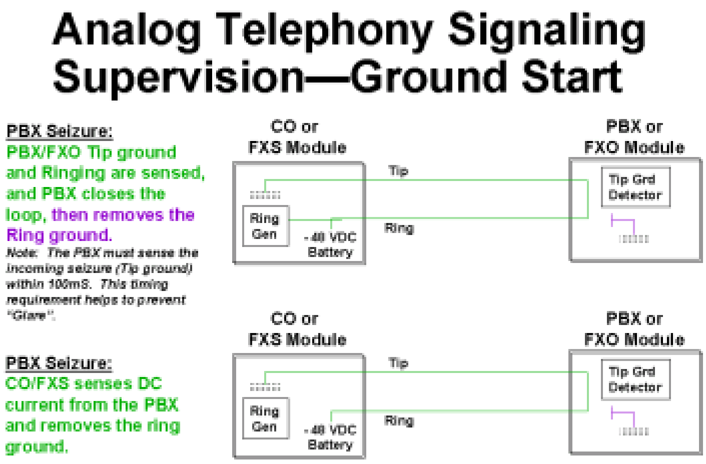
In the illustration, both the tip and ring linse are disconnected from ground. The PBX and FXO are constantly monitoring the tip line for ground and the CO and FXS are constantly monitoring the ring line for ground. Battery (–48 VDC) is still connected to the ring line just as in loop-start signaling. Figure 17 shows a call originating from a PBX or FXO.



In the illustration, a PBX or FXO grounds the ring line to indicate to the CO or FXS that there is an incoming call. The CO or FXS senses the ring ground and then grounds the tip lead to let the PBX or FXO know that it is ready to receive the incoming call. The PBX or FXO senses the tip ground and respond by closing the loop between the tip and ring lines. It also removes the ring ground. This process completes the voice connection to the CO or FXS, and voice communication can start. Figure 18 displays a call coming from the CO or FXS.



In Figure 18, the CO or FXS grounds the tip line and then superimposes a 20-Hz 90-VAC ringing voltage over the ring line to alert the PBX or FXO of an incoming call. Figure 19 shows the final phase of ground-start signaling.



In the above illustration, PBX or FXO senses both the tip ground and ringing. When the PBX or FXO has available resources to make the connection, the PBX or FXO closes the loop between the tip and ring lines and removes the ring ground. 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 will have to reorder the call. This 100-ms timeout helps prevent glare.

Digital Ground-start Signaling

This mode of operation is most commonly used in foreign exchange (FX) trunk applications.

FXS originates:

Idle condition:

To FXS: A bit = 1, B bit = 1

From FXS: A bit = 0, B bit = 1

Step 1 : FXS originates the call. B bit from FXS goes to 0:

To FXS: A bit = 1, B bit = 1

From FXS: A bit = 0, B bit = 0 (FXS originating call)

Step 2 : A bit from FXO goes to 0:

To FXS: A bit = 0 (FXO responding), B bit = 1

From FXS: A bit = 0, B bit = 0

Step 3 : FXS responds by transmitting A=1, B=1 to FXO:

To FXS: A bit = 0, B bit = 1

From FXS: A bit = 1, B bit = 1

FXO Originates:

Step 1 : FXO changes the A and B bits from 1 to 0 (B bit follows ring cycle):

To FXS: A bit = 0, B bit = 0

From FXS: A bit = 0, B bit = 1

Step 2 : FXS responds by changing the A bit from 0 to 1. FXO responds by tripping the ring generator. When the ring generator is tripped, the FXO returns the B bit to 1:

To FXS: A bit = 0, B bit = 1

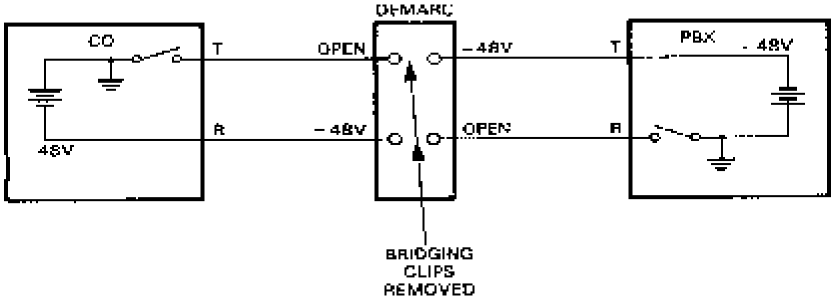
From FXS: A bit = 1, B bit = 1

Ground-Start Testing

Testing ground-start trunks is similar to testing loop-start trunks; however, some tests between the PBX and the demarc, with bridging clips removed, can usually be made.

Idle Condition (on hook)

The idle condition is represented in Figure 20. The bridging clips have been removed to isolate the PBX from the CO. Looking toward the PBX, –48V is observed on the T lead, and the R lead is open. Looking toward the CO, –48V is observed on the R lead, and the T lead is open.



A voltmeter connected from R to ground on the CO side of the demarc, or from T to ground on the PBX side, should read approximately –48V. An ohmmeter connected between T and ground on the CO side should read a very high resistance. Many PBXs have some voltage present between R and ground in the idle state. Erroneous measurements and damage to the meter may occur if resistance measurements are attempted. Refer to the PBX manufacturer's technical manual before measuring R-to-ground resistance on the PBX side of the demarc.

Outgoing (off hook)

To test a ground-start trunk for outgoing calls, remove the bridging clips and connect a test telephone and voltmeterp; then proceed as follows:

1. Observe the voltmeter. With the test telephone on hook, the meter should read near 0.0V.

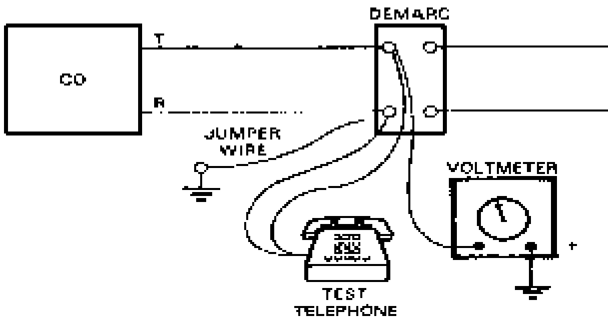
2. Go off hook and listen. There should be no dial tone.

3. Observe the meter. It should read near –48V.

4. Momentarily ground the R lead with a jumper wire and listen for a dial tone again. A dial tone should be heard shortly after the ground is removed.

5. Observe the voltmeter. The reading should be much lower than before, indicating that the CO is sending T ground.

6. Dial a station or a milliwaft test termination number. If the call completes, audio can be heard.



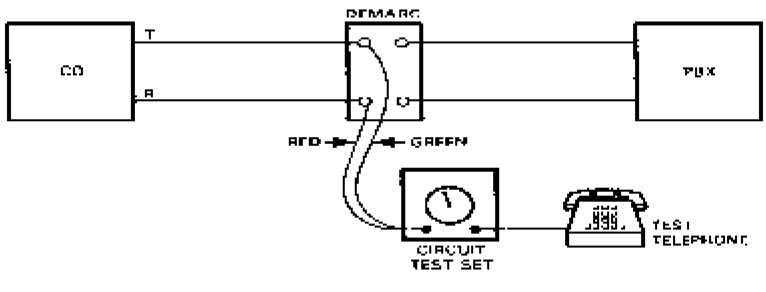
Incoming (ringing at destination)

Ground-start trunks may be tested for incoming call operation with a test telephone using exactly the same procedure as for loop-start trunks.

Loop Current Testing

For reliable operation, loop-start and ground-start trunks must have at least 23 milliamperes (mA) of direct current flowing when the loop is closed. Less than 23 mA results in erratic operation such as intermittent dropouts and inability to seize. If loop current is marginal, the trunk may test well with a test telephone, but operate erratically when connected to the PBX. Whenever a trunk operates erratically, loop current should be measured with a circuit test set.

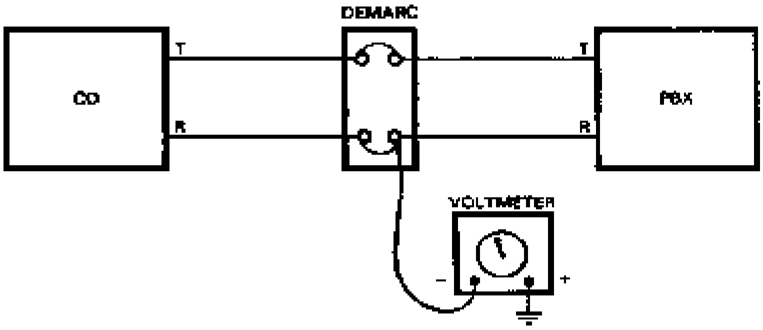
Figure 22 illustrates the test setup. With the bridging clips removed, connect the green test lead to T and the red test lead to R on the CO side of the demarc. The yellow lead is not used for this test.



To measure loop current, go off hook with the test telephone and listen for a dial tone. When testing a ground-start trunk, momentarily ground the R lead. When dial tone is obtained, press the push to measure button on the test set and read the current on the loop mA scale. The reading should be between 23 and 100 mA.

DID Trunk Testing

The idle condition is represented in Figure 23. Looking toward the PBX, ground is observed on the T, and battery is observed on the R lead. Looking toward the C0, a high-resistance loop is observed between T and R.



When the call is answered, the PBX places the battery on the T lead and ground on the R lead. This condition is known as a T-R reversal. This voltage reversal can be observed on the voltmeter. Because of the reversal of battery and ground on the T-R leads, this type of signaling is called loop reverse battery.

Call Disconnect

If the CO disconnects first, a brief voltage increase is observed while the loop in the CO switch goes from low to high resistance. This process is followed by a voltage reversal when the PBX goes on hook.

If the PBX disconnects first, a voltage reversal is observed, followed by an increase in voltage when the CO goes on hook and the CO loop goes from low to high resistance.

Several test calls should be made. After each test call, the bridging clips should be removed and the circuit tested to ensure that it has returned to the idle condition.

Demarc to PBX

Many PBXs can be tested for direct inward dial (DID) operation from the demarc with the bridging clips removed. Perform this lost as follows:

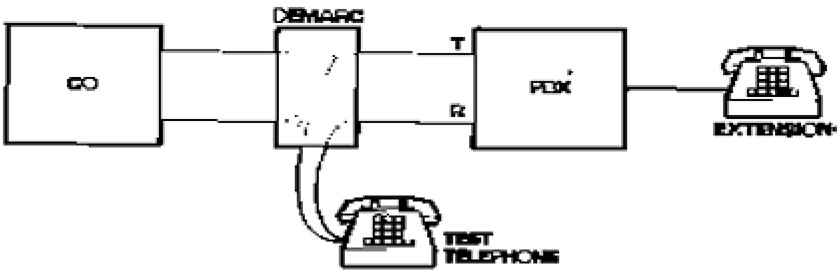
1. Go off hook with the test telephone.

2. Dial the 1- to 4-igit address of a PBX extension.

3. If the called extension rings, go to Step 4.

4. Attempt a conversation between the test telephone and the called extension. If good audio transmission occurs, then the PBX and trunk are functioning well as far as the demarc.

5. It problems occur at Steps 3 or 4, then the DID operation is faulty and must be corrected.



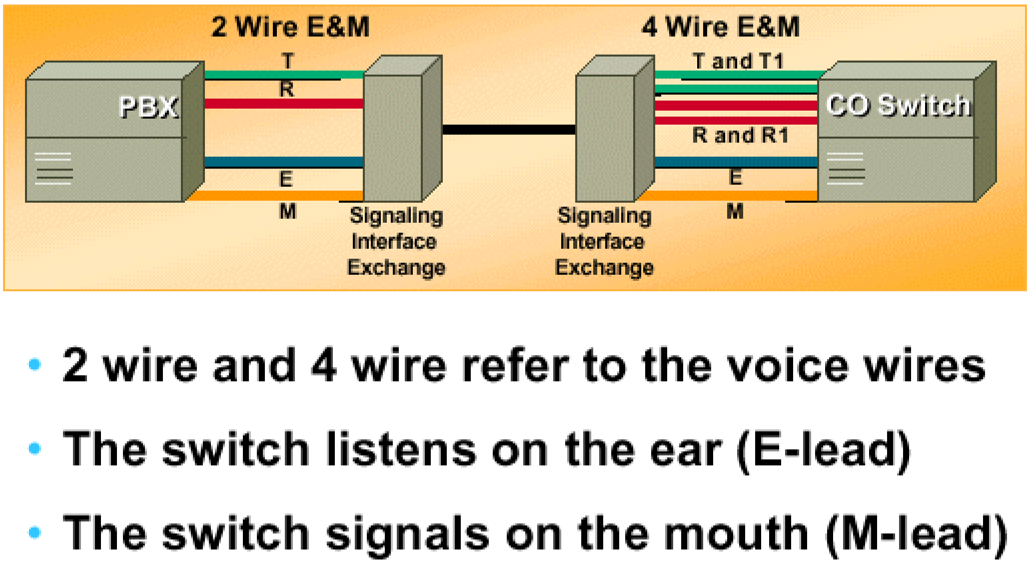
E&M Signailing

Another signaling technique used mainly between PBXs or other network-to-network telephony switches (Lucent 5 Electronic Switching System [5ESS], Nortel DMS-100, and so on.) is known as E&M. E&M is commonly referred to as ear and mouth or receive and transmit. There are five types of E&M signaling, as well as two different wiring methods (two wire and four wire). Table 1 shows that several of the E&M signaling types are similar.

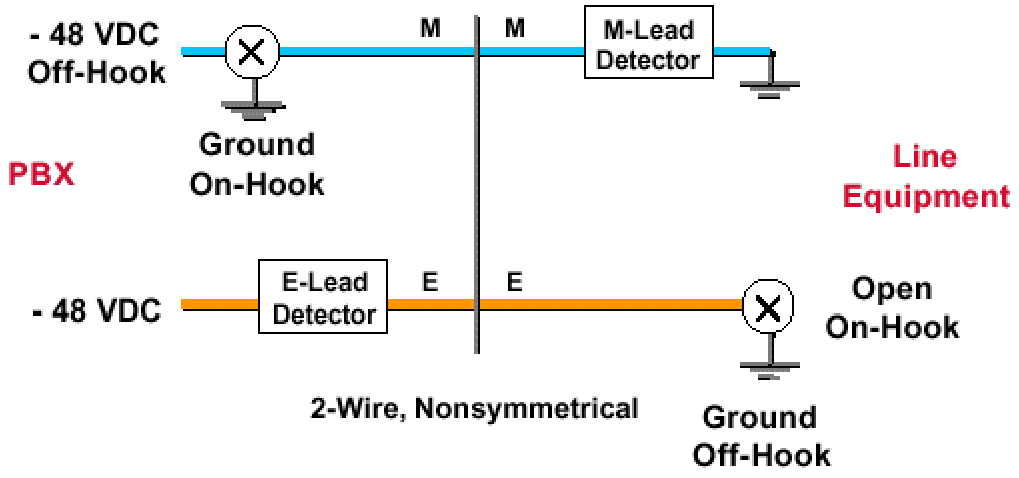
Table 1 E&M Signaling

Type	M Lead Off hook	M Lead On Hook	E Lead Off hook	E Lead On Hook
I	Battery	Ground	Ground	Open
II	Battery	Open	Ground	Open
III	Loop Current	Ground	Ground	Open
IV	Ground	Open	Ground	Open
V	Ground	Open	Ground	Open
SSDC5	Earth On	Earth Off	Earth On	Earth Off

Types I and II are the most popular E&M signaling in the Americas. Type V is used in the United States, and is very popular in Europe. Similar to type V, SSDC5A differs in that on- and off-hook states are backward to allow for fail-safe operation: if the line breaks, the interface defaults to off hook (busy). Of all the types, only types II and V are symmetrical (can be back to back using a crossover cable). SSDC5 is most often found in England. The Cisco 2600/3600 series currently supports types I, II, III, and V utilizing both two- and four-wire implementations. The following illustration depicts 2 wire and 4 wire E&M signaling connections. Voice travels over the tip and ring lines. Signaling occurs over E&M lines.

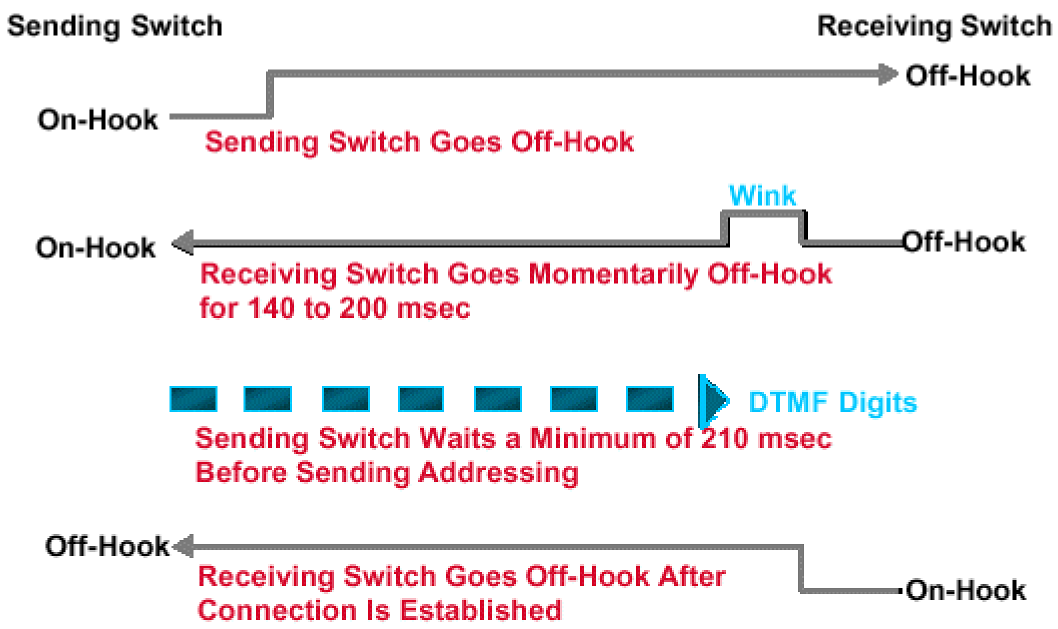


The next figure illustrates type 1 E&M signaling with a 2 wire line.

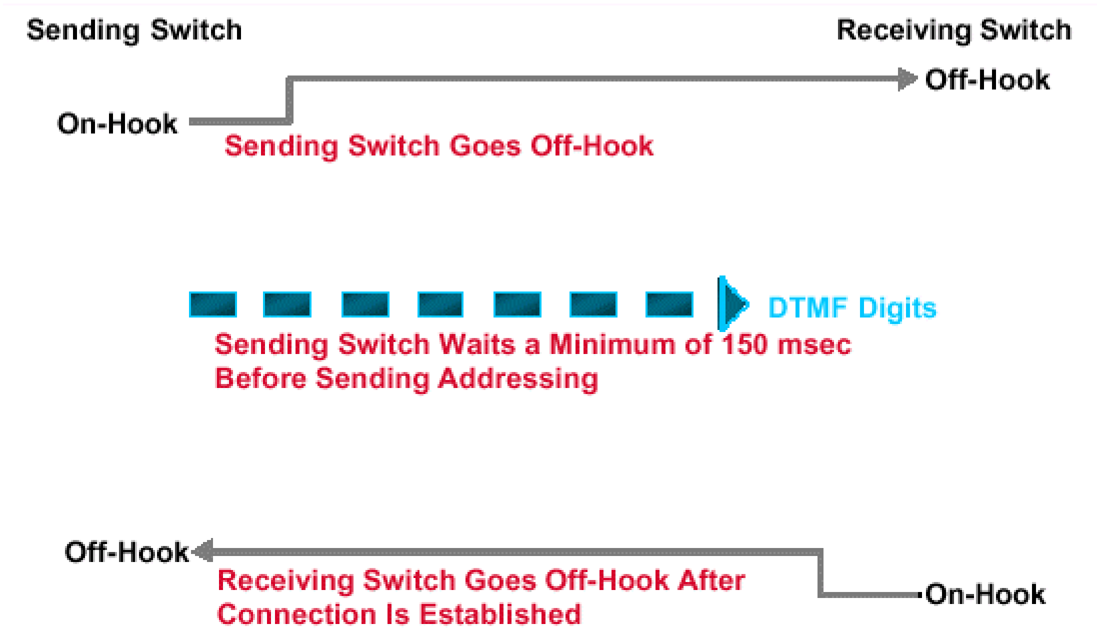


- **Common ground must exist between PBX and line equipment**

The following illustration shows the process that takes place during wink start signaling.



The next figure displays the immediate wink start signaling process.

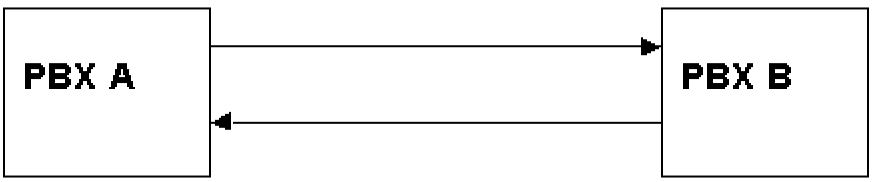


Digital E&M Signaling

Digital E&M signaling is a two-state signaling scheme (on hook/off hook) commonly used on digital four-wire CO and tie trunks. "A bit" signaling transmits the signaling state. The "B bit" (or B, C, D bits in the case of extended superframe [ESF]) follows the same state as the A bit.

Idle Condition

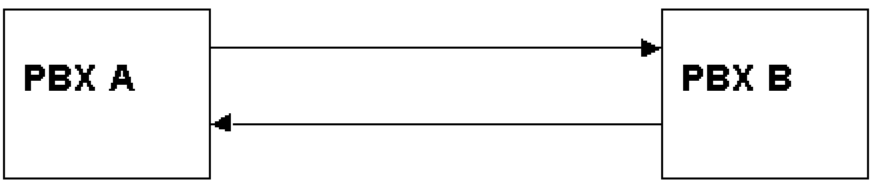
To PBX B: A bit = 0, B bit = 0



From PBX B: A bit = 0, B bit = 0

PBX A Goes Off Hook

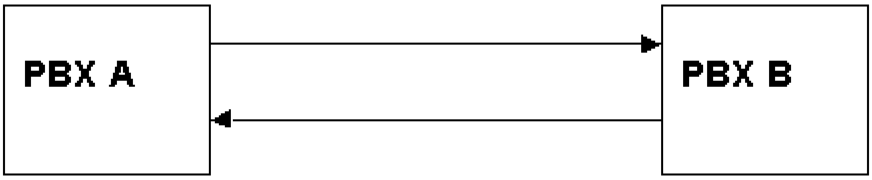
To PBX B: A bit = 1, B bit = 1



From PBX B: A bit = 0, B bit = 0

PBX B Answers

To PBX B: A bit = 1, B bit = 1



From PBX B: A bit = 1, B bit = 1

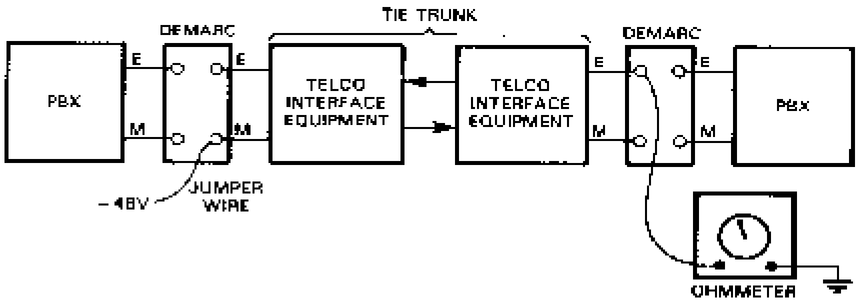
Note: *The originating switch may receive dial tone or wink back from the distant end after initiating the call depending on the application.*

E&M Tie Trunk Testing

Since the PBXs at both ends of the tie trunk are part of the same private network, private network technicians can perform end-to-end tests on the trunk, even though the transmission path may include leased facilities in the public network. Technicians at both ends of the trunk work together, coordinating their activities by talking over each other's facilities. The following testing procedures cover testing of only E&M signaling types I and II.

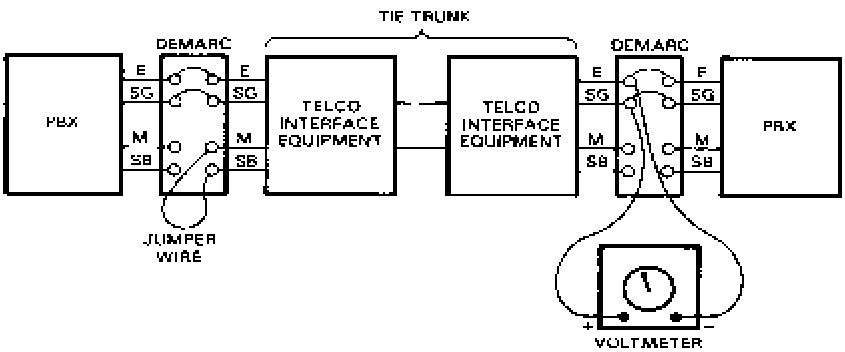
Type I:

To test Type I E&M signaling, bridging clips are removed from the E and M leads at both ends. Ohmmeters are connected between the E leads and ground. When the M lead at one end of the trunk is jumpered to –48V, the ohmmeter reading at the other end should go from open to a very low resistance, indicating E-lead ground. (See Figure 27.)



Type II:

The test setup for Type II is illustrated in fIGURE 28. Bridging clips are removed only from the M and signal battery (SB) leads. Voltmeters are connected between E and signal ground (SG). Under idle conditions the voltmeters should read battery voltage from the PBX, approximately –48V. When a jumper wire is connected between M and SB at one end of the trunk, the voltmeter reading at the far end should decrease to a low value, indicating E-lead ground.



ITU-T Signaling System 7

Common Channel Signaling Systems

Common channel signaling (CCS) systems are usually High-level Data Link Control (HDLC)-based message-oriented signaling systems. Within the United States PSTN, the original implementation of CCS started in 1976, and was known as CCIS (common channel interoffice signaling). This signaling is similar to ITU-T's Signaling System 6 (SS6). The CCIS protocol operated at relatively low bit rates (2.4K, 4.8K, 9.6K), but transported messages that were only 28 bitslong. However, CCIS could not adequately support an integrated voice and data environment. Therefore, a new HDLC-based signaling standard and ITU-T recommendation was developed: Signaling System 7.

First defined by the ITU-T in 1980, the Swedish Post, Telephone, and Telegraph (PTT) started SS7 trials in 1983 and some European countries are now entirely SS7-based.

Within the United States, Bell Atlantic began implementing SS7 in 1988, among the first Bell operating companies (BOCs), if not the first, to do so.

Currently, most of the long-distance networks and local-exchange-carrier networks have migrated to implementations of ITU-T's Signaling System 7 (SS7). By 1989, AT&T had converted its entire digital network to SS7; and US Sprint is SS7 based. However, many LECs are still in the process of upgrading their networks to SS7 because the number of switch upgrades required for SS7 support impacts the LECs much more heavily than the ICs. The slow deployment of SS7 within the LECs is also, in part, responsible for delays incorporating ISDN within the United States.

There are three versions of SS7 protocols at the present time:

- ITU-T version (1980, 1984) detailed in ITU-T Q.701 - Q.741
- ATT and Telecom Canada (1985)
- ANSI (1986)

Signaling System 7 U.S. PSTN Features

SS7 currently provides support for POTS through the use of a telephony user part (TUP), which defines the messages that are used to support this service. An additional ISDN user part (ISUP) has been defined that supports ISDN transport. Eventually, since the ISUP includes translations from POTS to ISDN, the ISUP is expected to replace the TUP. Figure 29 shows where SS7 takes control of the voice network.