POTS Signalling

This lecture describes the oldest access technology still in common use, the Plain Old Telephone Service (POTS). POTS is used to provide access to the Public Switched Telephone Network (PSTN).
After this lecture you should be able to: define the terms introduced in the lecture, describe the POTS services provided by the CO; and describe and recognize the waveform or impedance changes used by POTS signalling for: loop-start and ground-start line seizure, pulse and DTMF dialing, ringing, and basic call-progress tones (dial tone, ringback, busy).

Introduction

It's useful to understand how basic POTS ("landline") signalling works because cellular and internet-based telephony systems follow many of the same conventions.

The PSTN provides what is often called "plain old telephone service" (POTS). The PSTN allows its users ("subscribers") to place and receive phone calls to and from any of about 7 billion mobile phones and about 1 billion fixed telephone lines (as of 2017).

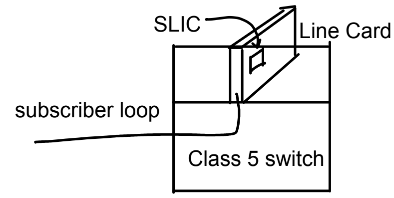
This lecture describes the POTS access technology, the physical interface between the subscriber's phone and the central office and signalling between the subscriber and the CO.

The PSTN interface is designed to carry an analog speech signal with frequency components from about 300 to 3400 Hz. It is also used with voice-band modems for data and fax.

POTS service is supplied using differential analog signalling over one twisted pair "loop" per subscriber. The pair can be several kilometers long, the maximum length depending on the wire gauge and use of "loading coils". One wire is labelled 'tip' and the other 'ring'. These terms originate with the jacks used in switchboards when switching was done manually by an operator.

POTS Services Provided by CO

The interface on a modern CO is provided by a "line card" that is installed in a (usually, very) large piece of equipment called a telephone "switch." The line card contains the required interface electronics. These are usually integrated into an IC called a Subscriber Line Interface Circuit (SLIC).

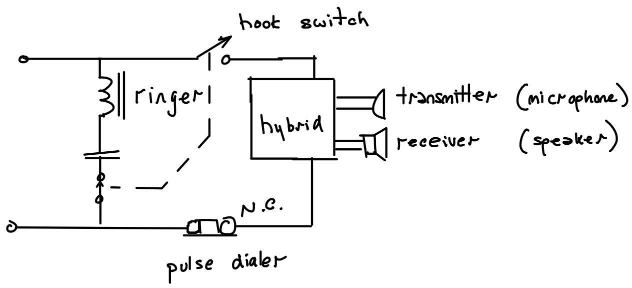


The SLIC provides a set of services that are often referred to by the acronym BORSCHT, which stands for:

- **B**attery: supplies -48V (relative to ground) on ring and 0V on tip. This allows the CO to detect when the subscriber "seizes" the line by going off-hook.
- **O**verload protection: the interface protects against contact with power lines or transient overvoltages (e.g. nearby lighting strikes).
- **R**inging: the CO notifies the subscriber of an incoming call by placing a 40–90 VAC signal at about 20 Hz on the line
- **S**upervision: the CO detects when the phone goes on- or off-hook to begin or end a call
- **C**oding: the line card converts the analog audio signal to/from digital a sampled digitized form since modern PSTN switches and trunks are digital
- **H**ybrid: the two directions, transmit and receive, are present simultaneously on the 2-wire pair but must be separated for transmission over the network
- **T**esting: for diagnostic purposes the loop voltage and current can be measured and often the transmit output can be looped back to the receive input

Telephone Schematic

The diagram below shows a simplified diagram of a telephone set.



When the phone is on-hook the ringer is connected across the line through a capacitor. Thus the phone appears as an open circuit at DC but AC current can flow through the ringer to ring the phone.

When the phone is taken off-hook, the "hook switch" allows current to flow through the transmitter and receiver. The varying resistance of the transmitter (microphone) causes the current to vary at the audio frequency. This time-varying current causes the corresponding sound the be reproduced at both the near and far ends of the loop.

In an actual phone there is a network called a hybrid that provides some isolation between the transmitter and receiver. Some feedback of the subscriber's voice (called "sidetone") is desirable. It is possible to get the user to adjust their speaking volume by varying the sidetone level.

The hook switch may also disconnect the ringer to avoid loading the voice circuit and to avoid the ringing due to dial pulses.

Some phones use pulse dialing to transmit the called number to the CO. The dialer breaks the loop current to encode the number as described below. The dialer also shorts out the transmitter and receiver to avoid distorting the pulses or causing annoying 'clicks' to be heard in the receiver.

Signalling

Signalling is the transfer of control information necessary to manage a communication link. PSTN signalling between the CO and subscriber is analog.

Since there is only one pair between the CO and the subscriber, signalling must be done over the same pair that is used to carry the call. This is called in-band signalling. Most other PSTN signalling, such as that between switches, is done over dedicated channels that carry control information between switches. This is called Common Channel Signalling (CCS).

POTS Signalling

Signalling between the subscriber and CO can use four techniques:

- the subscriber can switch the loop current off or on to indicate that the receiver is on- or off-hook;
- the subscriber can interrupt the loop momentarily or send tones to the CO to "dial" a number
- the CO can place a low-frequency (20 Hz), high-voltage (80 VAC) ringing signal across the pair to indicate an incoming call
- the CO can put audio-frequency (1-2 kHz) low-voltage (-9 dBm) tones on the loop current to report the state of the call

Hook State and Current Signalling

When the phone is not in use it appears as an open circuit at DC. When the subscriber picks up the handset the hook switch causes loop current to flow. This indicates that the subscriber wants to place or answer a call.

The CO supplies a loop voltage of -48 VDC from a lead-acid battery (nominally (4×6) cells $\times 2$ volts). The voltage on the tip lead is 0V and on the ring is -48V relative to ground. A negative voltage relative to ground is used so that any leakage currents to ground will result in deposition of copper onto rather than away from the copper wires.

The phone line can be "seized" by completing the tip and ring circuit (e.g. by the hook switch). This is called "loop start." In specialized applications the ring can be grounded instead. This is called "ground start."

There are current-limiting resistors in the CO of 400 to 800 ohms (depending on the loop length). The phone itself has a DC resistance of 100 to 400 ohms.

The loop current when the phone goes off-hook will depend on the resistance of the loop and the phone. Typical off-hook currents are 20-120 mA. The

CO has minimum threshold for detecting off-hook current, on the order of 6 to 25 mA.

Exercise 1: Assuming a zero-length loop and CO current-limiting resistors of 400 ohms and phone resistance of 200 ohms, what is the loop current? What are the voltages relative to ground assuming the CO resistance is split into two 200 ohm resistors?

Exercise 2: If the battery voltage is 48V and the loop current is 48 μA, what is the loop resistance? Is the phone on-hook or off-hook? What if the loop current is 48 mA?

The DC resistance of a 24-gauge loop (both conductors in series) is about 52 ohms per kft ("kilofeet" or 1000 feet, a term still used in the US) but will vary with the gauge.

Exercise 3: Assume the CO and telephone in the above example are now operating over a 15kft (about 5km) 24-gauge loop. What is the loop current?

Dialing

The subscriber transmits the phone number of the desired party using either pulses or dual-tone multi-frequency (DTMF) signalling.

To dial a number using pulses, the loop current is interrupted at a rate of 10 pulses per second with a minimum gap between digits of 400–900ms. The open/close ratio is 60 to 66%. A pause of several seconds before the called number is finished causes the call to be released.

The timing of the pulses was designed to drive electromechanical switches and is enforced by a mechanical clutch mechanism that controls the speed at which the dial can turn (in both directions).

For DTMF signalling, two tones are transmitted, one at a lower frequency (697, 770, 852 or 941 Hz) and one at a higher frequency between (1209, 1339, 1447 or 1633 Hz). The frequency tolerance is $\pm 1.5\%$. The minimum tone duration is 50 ms with a 50 ms pause between digits.

Exercise 4: What is the maximum number of DTMF digits that could be sent in the time it takes to dial a '5' and wait until the start of the next digit?

The high-frequency tones are sent at a level of -4 to -9 dBm. The lower-frequency tones are sent at a level about 2 dB lower.

DTMF signalling has various advantages: faster dialing, lower signalling voltages, and, unlike loop current based dialing, the digits can be transmitted end-to-end.

Signal levels on telephone circuits are often given in dBm under the assumption that the impedance level is 600 ohms (although it often isn't).

Exercise 5: What is the amplitude of a -6 dBm tone?

Ringing

The CO places an AC voltage on the line to indicate an incoming call. The ringer is AC-coupled and generates a sound in response to the ringing voltage.

In North America the ringing waveform is 20Hz at approximately 86 V (voltage depends on loop length) with a 2 s on, 4 s off cadence.

The ringer is capacitively coupled so it does not draw current at DC. The AC ringing voltage causes the ringer to sound, but the DC (average) current is zero.

When the phone goes off-hook, the line is connected in series with the transmitter/receiver and the CO detects the unbalanced current due to the DC loop current being superimposed on the AC ringing current.

Call Progress Tones

When the phone is off-hook the CO can transmit various tones to indicate the status of a call. These include:

- dial tone: the subscriber may start dialing
- ring-back: the call has not been set up
- busy: the phone at the remote end is off-hook (possibly due to a call in progress)
- other: various other tones indicate problems with the network (all trunks busy, no such number, toll call, etc)

The following definitions are taken verbatim from the Wikipedia articles:

- dial tone is a continuous tone having frequencies of 350 and 440 Hz at a level of −13 dBm
- ringback tone is defined as comprising frequencies of 440 and 480 Hz at a level of −19 dBm and a cadence of 2 seconds ON and 4 seconds OFF

- busy tone is defined as having frequency components of 480 and 620 Hz at a level of −24 dBm and a cadence of half a second ON and half a second OFF
- reorder tone, also called "fast busy" tone, contains the same frequency components as busy tone at a similar level but with a cadence of 0.25 of a second on and 0.25 of a second off;

Calling Party ID

The phone number of the calling party can be transmitted from the CO to the subscriber using audio-frequency FSK modulation between the first and second rings.

Trunk Interfaces

Trunk interfaces often work differently than those used for subscribers.

E&M is an alternative signaling method that simplifies a telephony interface by separating the voice and signalling circuits. It is typically used between a PBX (private branch exchange, a privately-owned switch) and a trunk to a service provider's CO. It uses two circuits (labelled E and M) that allow either side to seize the associated voice circuit. Various configurations are possible. Some use pairs and others use ground return. Some signal by grounding the line and others by applying battery voltage. Since the trunk interfaces can be configured to operate in many different ways it is important that both sides of the interface be configured to be compatible.

Some trunks are "four-wire" as opposed to "two-wire." Four-wire connections use a different pair for each direction (typically labelled T&R, T1&R1). This avoids the need to use a hybrid on each end of the loop.

Government Regulations and Industry Standards

Unlike many other areas of telecommunications, governments have set technical standards for telephony equipment.

In the US, CFR 47 (FCC Regulations) Part 68, also published by the Telecommunications Industry Association (TIA) as standard TIA-968, specifies terminal equipment (telephones, modems, etc.) interoperability requirements such as connectors, impedances, signal levels and frequencies to ensure telephones and other devices from different manufacturers can be connected to the PSTN.

In Canada, Industry Canada sets similar standards in regulation CS-03, which is based in large part on TIA-968.