Building Residential VoIP Gateways: A Tutorial
Part Two: VoIP Telephony Interfaces

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When developing a VoIP system, one key area of consideration is the interface to an analog telephone. The designer must understand the telephony requirements that exist in the PSTN, as they must also be supported in VoIP systems. These articles are intended to provide engineers with design considerations for all major portions of their VoIP product. In this portion, we'll focus on the two most common interfaces to a standard POTS phone: Foreign eXchange Subscriber (FXS) and Foreign eXchange Office (FXO). It will describe the functionality provided by FXS and FXO circuits, cover some history of FXS and FXO, discuss industry standards, and highlight some of the challenges designers may face when supporting analog telephony interfaces on their VoIP residential gateway.

FXS and FXO are common terms in the world of analog telephony, but what is the difference between the two and why are they important in VoIP applications? In a traditional telephone connection over the PSTN, the telephone central office switch feeds battery and provides ringing to the phone. The phone itself completes the tip/ring circuit to request service or answer a call from the PSTN. For calls placed over the Internet, the FXS circuit emulates the telephone central office switch. The residential gateway "pretends" to be the switch, providing both battery and ringing to the phone and detecting loop current. The FXO circuit, on the other hand, emulates a phone, providing loop closure and detecting incoming ringing.

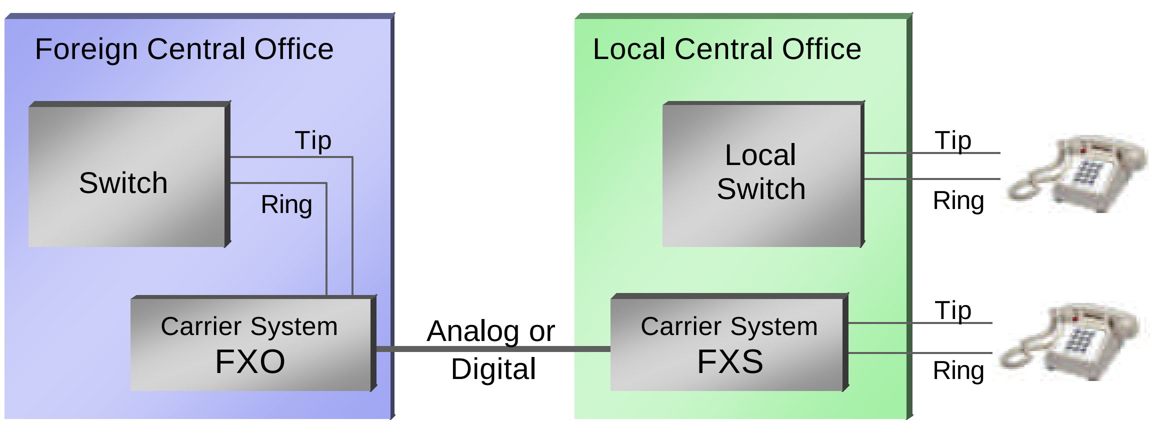


Fig. 1: Dial Tone From A Foreign Central Office

The terminology of FXS and FXO came from the desire to enlarge local calling areas. Before 800 toll-free calling was available, business customers seeking alternatives to expensive long distance charges were offered a foreign dial tone service. Carrier systems, first analog and then digital, were created to support this service, extending dial tone from a foreign central office (**F**oreign e**X**change **O**ffice) to multiple local central office sites

(**F**oreign e**X**change **S**tations). This application was one of the earlier uses for the FXO interface and is responsible for the terminology that still exists today.

Analog Phone And PSTN interfaces

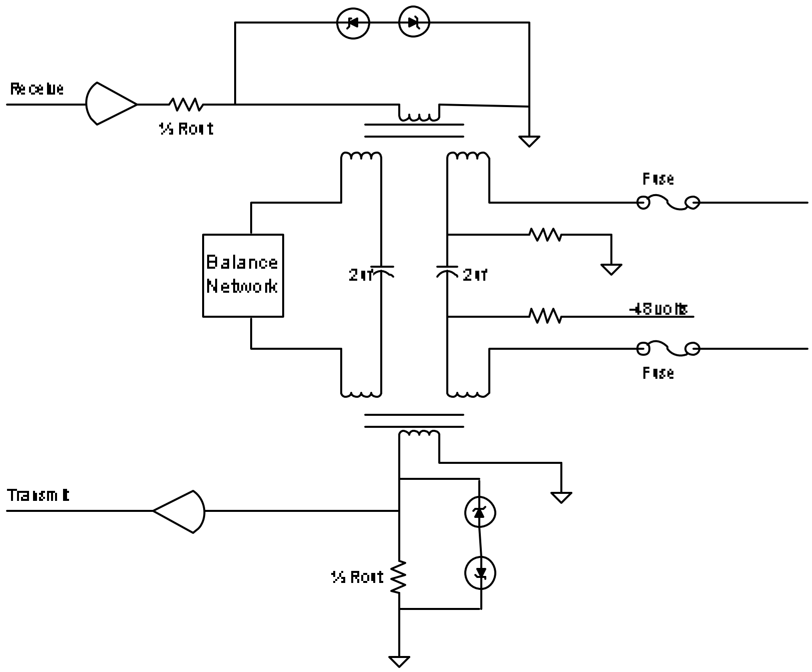
The FXS circuit consists of two parts, a CODEC and a SLIC (Subscriber Line Interface Circuit). A CODEC is comprised of an ADC and a DAC. The ADC converts the analog signal coming from the analog phone into a digital signal for transmission over the VoIP network. The DAC converts digital signals to analog levels to drive the analog phone. In order to achieve an audio bandwidth of 4 kHz, the sampling rate for both the ADC and DAC is usually around 8kHz. The SLIC device emulates PSTN voltage levels. It must detect if the phone is on-hook or off-hook and generate ringing voltages up to 120 V.

The circuitry for the FXO consists of a CODEC and a data access arrangement (DAA). The CODEC has the same functionality as in FXS, converting analog speech to digital signals, and vice versa. The DAA emulates a (POTS) phone. Its main function is to remove the high voltage dc bias, passing only the analog ac signal from the PSTN by applying a loop closure towards the PSTN.

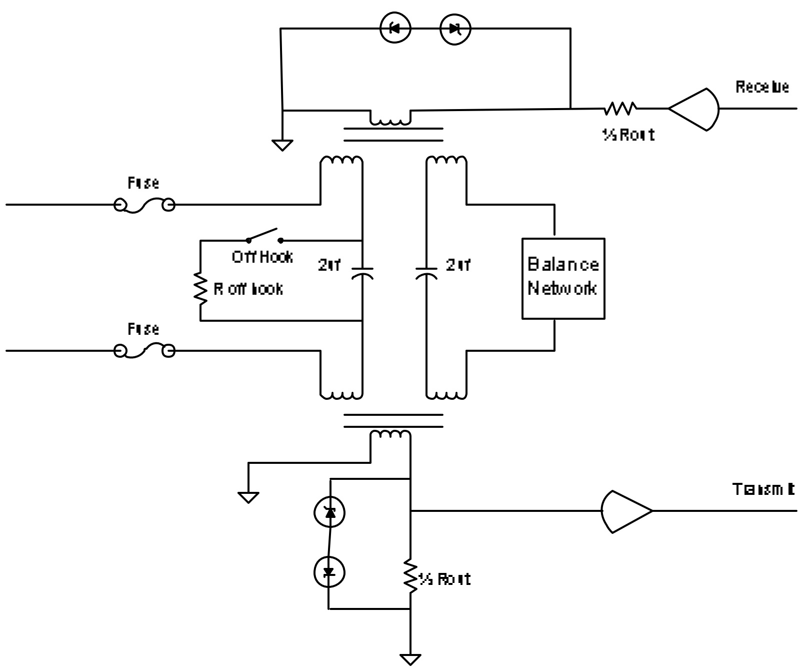
FXO Mirrors FXS

In VoIP gateways the FXS circuit is the primary interface for establishing outgoing calls and receiving incoming calls over the packet network. In a central office application the two-wire SLIC interface on a POTS line card serves as the FXS interface. In CPE applications, the FXS circuitry exists in the gateway, providing dial tone, battery current and ring voltages and detecting loop closure from the phone. Because this switch functionality resides at the CPE level, a direct connection to the PSTN is not necessary. There are cases, however, when a connection to the PSTN is useful using the FXO interface. It presents the same type of interface to the central office as an ordinary POTS telephone, with some improvements. Some important uses of the FXO port include:

- Lifeline for power failure: when there is no power to the voice gateway, the gateway is not able to connect to the packet network to place or receive a call. In this case, a relay can be used to connect the analog phone directly to the PSTN. When this situation occurs, the FXO circuit is intelligent and can detect a call is in progress, preventing that call from being disconnected once power is restored.
- Call re-direction: when the packet network is unavailable due to network congestion, the FXO circuit can remember the number dialed by the subscriber and route the call through the FXO circuit to the PSTN, to complete the call. This process prevents customers from having to redial the phone number when the packet network is down.
- Remote VoIP calling: when a VoIP customer is not at home, they can still make a VoIP call by calling their home number through the PSTN network. The voice gateway receives the call through the FXO port and forwards it to the VoIP network.



Receive
½ Rout
Fuse
Balance Network
2uf
2uf
-48 volts
Fuse
Transmit
½ Rout



Receive
½ Rout
Fuse
Off Hook
2uf
2uf
Balance Network
R off hook
Fuse
Transmit
½ Rout

B: Battery feed function found in an FXS linefeed interface. The complementary function in an FXO interface is battery sink. As seen in Fig. 3 a connection is made between the central office tip and ring leads by the FXO off-hook relay, with current limiting being provided by the FXO.
O: Over-voltage protection must be provided by the FXO due to exposure to lightning and power cross conditions. The SLIC's Tip and Ring inputs in the FXS circuit are designed to provide additional over-voltage protection.
R: Ringing is provided by the central office, but the FXO must be able to detect ringing and forward this information. The FXS circuit must provide ringing to the phone. A low-voltage ring signal generated by either the CODEC or SLIC is amplified by the SLIC and placed on the local loop to ring the phone.
S: Signaling refers to the ability of the FXO to receive on/off-hook information and present an off-hook on-command to the central office. It must also detect ringing and other conditions and transmit this information. The FXS must be able to detect on/off-hook states, detect and generate DTMF tones, and generate signals for caller ID.
C: Coding is a function of the CODEC devices that are part of both the FXS and FXO interfaces. It refers to the A-to-D and D-to-A coding of the voice signal.
H: Hybrid functionality is essential for stability and good voice quality and is equally important in both the FXS and FXO interfaces. Echo functions are detailed below.
T: Test is not normally an FXO function as automated maintenance and testing is provided by the central office. However, because the FXS circuit bypasses the PSTN, the required test and diagnostic functionality are included in the CODEC/SLIC.

Echo

The importance of stability and good voice quality are essential whether a call is made over the PSTN or packet network. The potential impact of echo is critical to the functions of both the FXS and FXO interfaces. Note too, that special hybrid functionality is required in both cases within the CPE device to handle the various line impedances in the world. Ordinary POTS telephones have relatively uncontrolled impedances between 200 O and 400 O. Since the current from office to subscriber is two-wire with no gain added, the impedance variations typically encountered do not affect performance. Stability and line echo issues can arise, however, when a carrier system uses two-to-four wire voice frequency (VF) hybrids on each end, as well as possible gain in the four-wire path.

Line echo results from either the delayed "bleed-though" of the transmitted voice signal into the receive path at the hybrid (2-wire to 4-wire conversion point) or from reflections in the local loop due to impedance mismatches. Line echo is always present in the PSTN and is not necessarily a problem. In fact, some of your telephone's transmit signal is coupled into the receive path in order to generate sidetone. Sidetone lets the speaker hear his or her own voice in the receiver. Without sidetone, the speaker would be unsure if he or she was being heard on the other end, and could make for an awkward conversation.

When uncontrolled, however, excessive line echo can affect a caller's experience in two ways:

- The louder the echo, the more disruptive it will be during a voice call. Many times low levels of echo are present on the line, although they are not detectable by the user.
- The length of delay of the echo also greatly affects voice quality. This delay represents the time that elapses between when the user speaks, and when he or she hears their echo. Round trip echo delays greater than 25 ms will begin to affect voice quality.

The main function of the hybrid circuit that completes the 2-to-4-wire conversion and vice versa is to limit the amount of outgoing transmit signal that "bleeds" into the incoming receive path. As a result of transhybrid imbalance (hybrid component imperfections, impedance mismatches, etc.), some amount of Tx signal always gets into the Rx path (see Fig. 4).

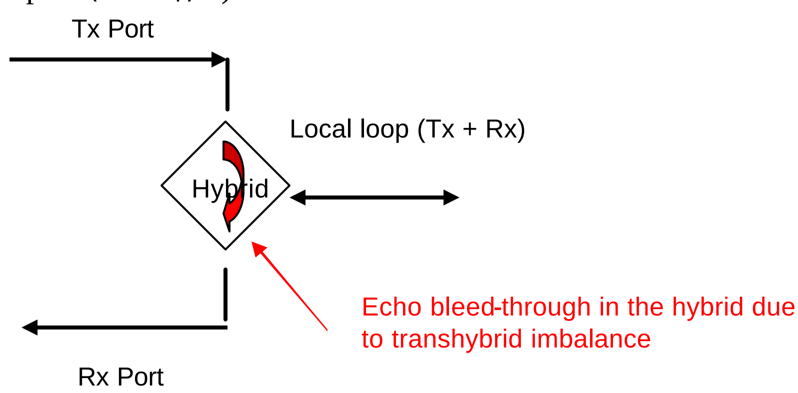


Fig. 4: Line Echo At 2-To-4-Wire Conversion

In addition to the echo caused by transhybrid imbalance, hybrid termination impedance mismatches can also cause line echo. If a line is not correctly terminated with its characteristic impedance, echoes will be generated. This echo is a result of the incoming signal from the 2-wire local loop hitting the hybrid termination resistance and reflecting back down the line (Fig. 5). Improperly terminated CPE equipment such as phones or modems can also generate echoes in the local loop.

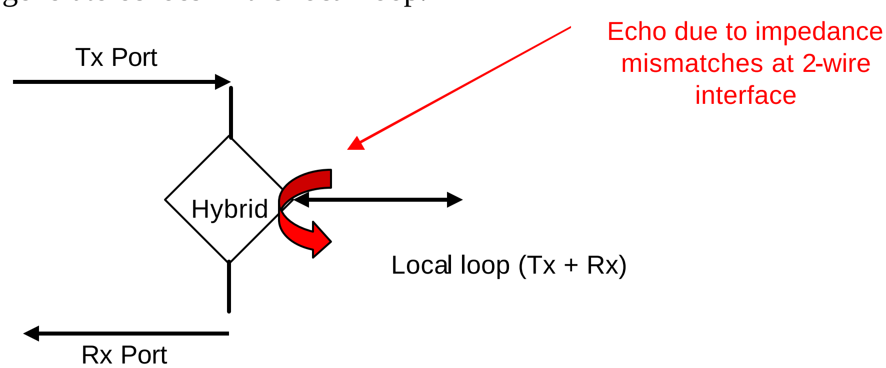
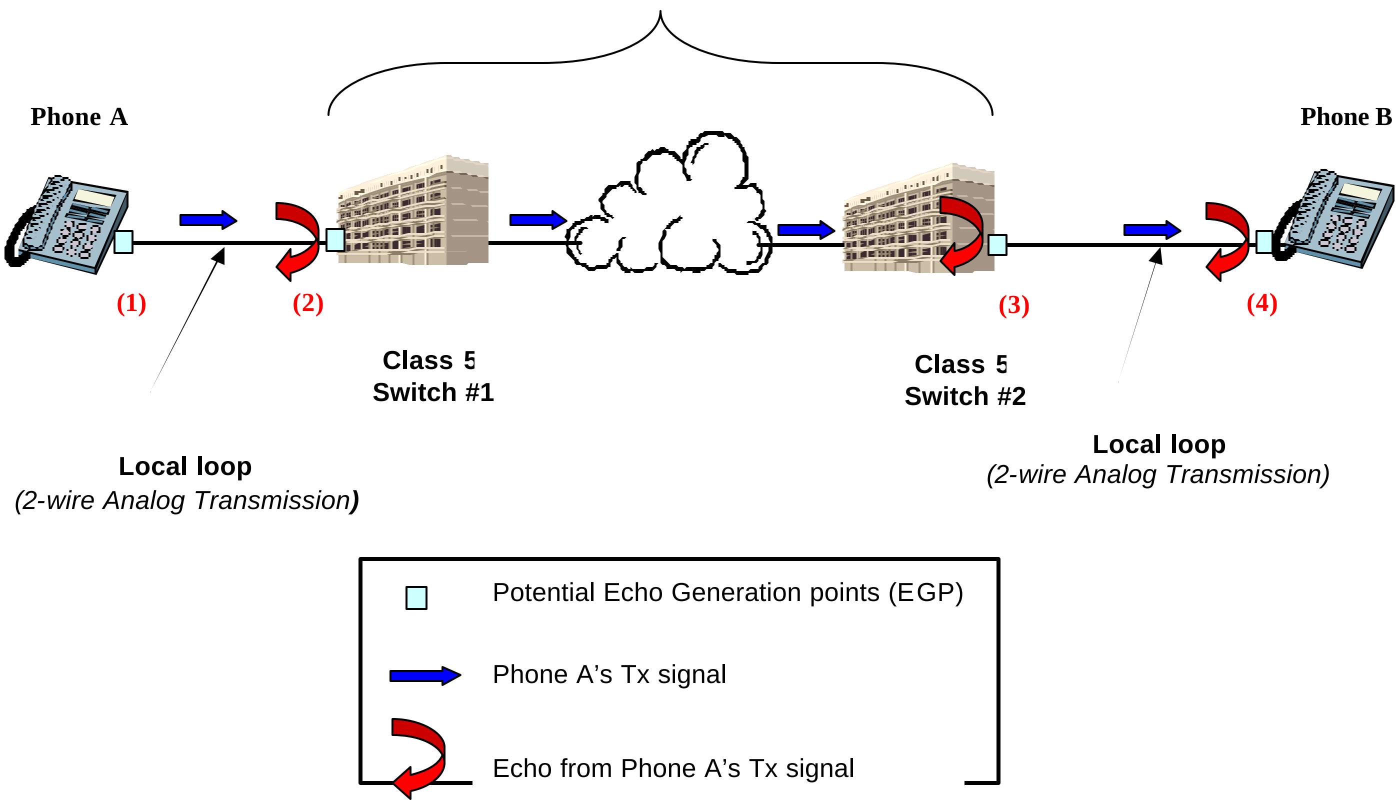


Fig. 5: Line Echo At The Local Loop

To get a better understanding of the sources of line echo, some background information on the PSTN is required. The telephone network consists of two basic sections: (1) the switching and transport core and (2) the local loop.

The switching and transport section is responsible for the transport and routing of calls, call services, billing, etc. Within this section, all voice and data signals are transmitted digitally, using separate paths for the Tx and Rx signals. This makes it much easier to transmit long distance signals, allowing for the use of repeaters, microwave transmission towers, etc. The local loop consists of the “last mile” of copper that connects the central office to the subscriber. For illustrative purposes consider a simplified PSTN diagram.

Fig. 6: PSTN Switching/Transport Network
(4-wire Digital Transmission)



If we trace the Tx signal of Phone A in the diagram, the light blue squares indicate potential echo generation points (EGPs) due to either transhybrid imbalance or impedance mismatches. In this example, we’ll assume that these EGPs actually produce large enough echoes to be problematic. You can see from the diagram that there are 3 places where echo will be generated for Phone A (EGP 2, 3 and 4). The echo at EGP 2 is a result of impedance mismatch of the hybrid to the line and results in an echo being sent back to Phone A. This echo is not problematic as the round trip delay is not very long and will not be noticed by the end user. This echo is also masked by the sidetone. The problematic echoes are the echoes generated at EGP 3 and 4 since their round trip delay will be large (>25 ms). It is characteristic for the echoes generated at the “far end” of the network to cause the most problems. The echo at EGP 3 is due to transhybrid imbalance, while the echo at EGP 4 is due to the impedance mismatch of the phone to the line.

To help eliminate the unwanted echoes, telephone service providers will typically install echo cancellers in the PSTN. In order to eliminate the echoes on Phone A’s connection, an echo canceller would need to be installed just before or in the #2 Class 5 switch. The echo canceller is inserted into the 4-wire section of the network. Its job is to model the echo characteristics of the local loop section containing EGP 3 and 4 and, along with knowledge of the Phone A’s Tx signal, cancel the echo generated at EGP 3 and 4 prior to transmitting Phone B’s Tx signal back to Phone A. In VoIP systems, the echo must be handled completely in the VoIP gateway for calls originating and terminating at VoIP CPE gateway. For calls that must be routed to the PSTN, the PSTN will continue to handle the echo cancellation function.

Standards

Although placing VoIP calls over the packet network does not require a direct connection to the PSTN, it does not preclude a system designer from meeting the requirements of the PSTN. A VoIP residential gateway designer must understand the telephony requirements that exist today and apply them to the FXS and FXO interfaces of their systems. There are four primary standards that are essential for understanding the behavior and requirements in the PSTN:

1. GR57 specifies the behavior of a Digital Loop Carrier system from the tip/ring pair at the central office to the tip/ring pair at the subscriber end of the system.
2. TA909 describes the behavior of Fiber-to-the-Curb systems from a digital interface at the central office (T1 for North America) to the tip/ring interface at the subscriber end of the system. Fiber-to-the-Curb systems are typically 16- to 24-channel systems.
3. GR303 discusses the behavior of “integrated” Digital Loop Carrier systems. Behavior is specified from the digital interface at the Central Office (T1) to the tip/ring interface at the subscriber end of the system. Systems of this type tend to be large with several hundred tip/ring pairs at the subscriber end.
4. GR1089 specifies the environmental standards for telephone infrastructure in North America. This includes specifying immunity to lightning, power cross, and EMC. It also specifies limits to unintended RF radiation (EMI).

Summary

As interest in VoIP service increases, phones specifically designed for voice calls over the packet network (i.e. IP phones) will eventually replace standard analog POTS phones. Until that time, however, designers of VoIP gateways must consider how their equipment will interface to these standard analog phones. The FXS and FXO circuits provide a means to this end.

For more information, visit www.ti.com/voip

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As a product manager in TI's VoIP group, Kim is responsible for the strategy for interfacing analog telephony solutions to TI's VoIP gateway products. Since joining TI in 1995, Kim has been a product manager for analog modem, central office line card and CPE gateway products. She earned her BS in industrial engineering from Iowa State University and has an MBA from Southern Methodist University.

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