International Journal of Information & Computation Technology.
ISSN 0974-2239 Volume 4, Number 14 (2014), pp. 1373-1378

http://www. irphouse.com

# Call Routing through Analog Voice Ports and Issues at the Analog Voice Ports Connection Points

**Gowthami Priya Meruva**

*SREE VIDYANIKETHAN ENGINEERING COLLEGE TIRUPATHI*

**Abstract**

In present days, many business people where moving their telecommunication network from PSTN to VOIP. These two networks associate with two distinct types of signaling (i.e. analog and digital signal), to understand each other routers came with voice ports, signaling interfaces like FXS, FXO, and E&M are analog interfaces and T1, E1 and ISDN are digital interfaces. When associating these, need to do specific configuration or else issues will rise like echo and poor voice quality.

In router analog voice ports connection points have issues with FXO ports and supervisory disconnect signaling problems are associated with the analog signaling interfaces and ports.

**Key Words:** Public Switch Telephone Network (PSTN), Voice over Internet Protocol (VOIP), FXS, FXO, E&M and PBX.

## 1. INTRODUCTION:

Voice ports are found at the cross points of packet-based networks and traditional telephony networks [1], and they facilitate the passing of voice and call signals between the two networks. Actually, voice ports connect a router or access server to a line from a circuit-switched telephony device in a PBX [5] or the public switched telephone network (PSTN).

## 2. Different types of Calls in VOIP Network

### a. Local Calls:

Local calls take place among two telephones associated to one Voice supported router. This kind of call is tackled completely by the router and does not go through an external network. Both phones attached to Foreign Exchange Station (FXS) ports on the router.

**b. On-Net Calls:**

On-net calls arise between two phones on same/identical network, the calls can be conquering through one or more voice aided routers, but the calls remain on the same network. The border telephones attached to the network via FXS ports or via a PBX [3], Which typically linked to the network through a T1 connection. IP phones that linked to the network through switches place on-net calls through Cisco unified Communication manager. The link through the network can be a LAN connection, or WAN connection.

**c. Off-Net Calls:**

To get access to the PSTN [3], the user dials an access code, such as 9, from a phone directly linked to a Cisco voice-supported router or PBX [3]. The link to the PSTN is usually a single analog connection via a Foreign Exchange Office (FXO) port or a digital T1 or E1 connection.

**d. PLAR Calls:**

PLAR calls robotically connect a phone to a second phone when the first phone goes off hook. When this association occurs, the user does not get a dial tone, because the voice-supported port that phone is linked to is preconfigured with a specific number to dial. A PLAR link can work between any type of signaling, including E&M, FXO, FXS, or any permutation of analog and digital interfaces. For example, you might have met a PLAR link at an airline ticket counter where you pick up a handset and are directly connected with an airline representative.

**e. PBX-to-PBX Calls:**

PBX-to-PBX callsmake at a PBX [3] at one site and dismissed at a PBX at another site while using the network as convey between the two locations. Many business environments attach sites with private tie trunks. When migrating to a converged voice and data network, this same tie-trunk connection can be emulated across an IP network. Modern PBX [5] connections to a network are typically digital T1 or E1 with channel associated signaling (CAS) or Primary Rate Interface (PRI) signaling, although PBX connections can also be analog.

**f. Inter-cluster Trunk Calls:**

As part of an overall migration policy, a business might swap PBXs with Cisco Unified Communications Managers. This includes IP phones linked to the IP network. Cisco Unified Communications Manager performs the call-routing functions formerly provided by the PBX. When an IP phone call is finding using a configured Cisco Unified Communications Manager, the call is evaluated to see if the call is destined for another IP phone below its control or if the call must be routed to a remote Cisco Unified Communications Manager for call completion. Inter-cluster trunk calls, are routed between Cisco Unified Communications Manager clusters using a trunk.

**g. On-Net to Off-Net Calls:**

When planning a strong call-routing strategy, you might need to redirect calls via an alternative path should the primary path fail. An on-net to off-net call, initiates on a home network and is routed to an outside network, usually to the PSTN [1]. On-net to off-net call-switching functionality might be essential when a network link is down or if a network becomes overloaded and unable to handle all calls existing.

## 3. Voice Ports

Voice ports on routers and access servers match physical telephony switch links so that voice calls and their associated signaling can be transferred entire between a packet network and a circuit-switched network or device. For a voice call to occur, certain data must be passed between the telephony devices at either end of the call, such as the on-hook status of the devices, the accessibility of the line, and whether an incoming call is trying to reach a device. This information is referred to as signaling, and to progression it correctly, the devices at both ends of the call segment, which are straight associated to each other, must use the same type of signaling and unacceptably coordinated electrical components can cause echo and create poor audio quality [6].

### 3.1 Signaling Interfaces:

Voice ports on routers and access servers considerably link the router, access server, or call control device to telephony devices such as telephones, fax machines, PBXs [5], and PSTN [1] central office (CO) switches via signaling interfaces.

These signaling cross points generate information about things such as

- On-hook status
- Ringing
- Line seizure

The signaling interfaces include FXO, FXS, and E&M, which are types of analog interfaces. Digital signaling interfaces include T1, E1, and ISDN. Some digital associations follow FXO, FXS, and E&M interfaces. It is vital to know which signaling method the telephony side of the connection is using and to match the router configuration and voice interface hardware to that signaling method.

### 3.2 Analog Voice Ports:

Analog voice port lines attach routers in packet-based networks to analog two-wire or four-wire circuits in telephony networks. Two-wire circuits tie to analog telephone or fax devices, and four-wire circuits connect to PBXs [3]. Relations to the PSTN CO [1] are typically made with digital interfaces. Three types of analog voice interfaces are supported by Cisco gateways.

The following is a detailed enlightenment of each of the three types of analog voice interfaces:

A. FXS: An FXS interface ties the router or access server to end-user equipment such as telephones, fax machines, or modems. The FXS crossing point supplies ring, voltage, and dial tone to the station and includes an RJ-11 connector for basic telephone equipment, key sets, and PBXs.

B. FXO: An FXO crossing point is used for trunk, or tie-line, connections to a PSTN CO or to a PBX that does not support E&M signaling (when the local telecommunications authority permits). This interface is of value for off- site station applications. A standard RJ-11 modular telephone cable connects the FXO voice interface card to the PSTN or PBX [5] via a telephone wall outlet.
C. E&M: Trunk circuits join telephone switches to one a new. They do not tie end-user equipment to the network. The most common form of analog trunk circuit is the E&M interface, which uses special signaling paths that are separate from the trunk audio path to convey information about the calls. The signaling paths are known as the E-lead and the M-lead. E&M links from routers to telephone switches or to PBXs are preferable to FXS and FXO connections because E&M provides better answer and disconnect supervision.

## 4. Analog Signaling

The human voice produces sound waves, and the telephone conveys the sound waves into electrical signals, analogous to sound. Analog signaling is not healthy/strong because of line noise. Analog transmissions are strengthening by amplifiers because the signal becomes reduced the farther it travels from the CO. As the signal is strengthened, the noise is also strengthened, which often causes an unfeasible connection.

In digital networks, signals are transmitted over great distances and coded, regenerated, and decoded without disgrace of quality. Repeaters amplify the signal and clean it to its original condition. Repeaters then establish the original sequence of the signal levels and send the clean signal to the next network destination. Signaling techniques can be placed into one of three categories:

A. Supervisory: Involves the discovery of changes to the status of a loop or trunk. When these changes are discovered, the supervisory circuit generates a fixed response. A circuit (loop) can close to connect a call, for example.
B. Addressing: Involves passing dialed digits (pulsed or tone) to a PBX or CO [5]. These dialed digits provide the switch with an association path to another phone or customer premises equipment.
C. Informational: Provides audible tones to the user, which indicates certain situation such as an incoming call or a busy phone.

### 4.1 FXS and FXO Supervisory Signaling:

FXS and FXO interfaces indicate on-hook or off-hook status and the seizure of telephone lines by one of two access signaling methods: loop-start or ground-start. The type of access signaling is determined by the type of service from the telephone company's CO. Standard home telephone lines use loop-start, but business telephones can order ground-start lines instead.

### 4.2 E&M Signaling:

E&M is another signaling technique used mainly between PBXs [3] or other network-to- network telephony switches. E&M signaling supports tie-line type facilities or signals between voice switches. Instead of superimposing both voice and signaling on the same wire, E&M uses separate paths, or leads, for each.

## 5. Understand Supervisory Disconnect Signaling Methods

### 5.1 Ground-start Signaling Disconnect

Ground-start signaling can be used on the FXO port of the router if the switch is able of providing a ground-start association. When configured, the switch take off the ground from the connection and the FXO port goes on-hook.

### 5.2 Power Denial-based Supervisory Disconnect:

Power denial discovery is a disruption of line power from the switch or PBX to the FXO port, which lasts at least 350 ms. The FXO interface on the router finds that power is no longer there and interprets this as a supervisory disconnect indication.

### 5.3 Battery Reversal:

Battery reversal is established by reversing the battery division on the PBX. This is done primarily when the call is associated (far-end answer), with the polarity reversed throughout the entire discussion. When the far-end disconnects, the battery polarity is changed back to normal to indicate call cut off. PBX uses the battery reversal indication to start billing.

### 5.4 Tone-based Supervisory Disconnect:

Supervisory Tone is the audible frequencies that a PBX [5] can produce to indicate that a call has been released (caller back on-hook) and the connection should be disconnected. The tones are different in most countries. The router's FXO port can be configured to interpret the tones as Supervisory Disconnect and disconnect the call.

## 6. Issues at the Analog Voice Port connection point

When loop-start signaling is used, a router's FXO crossing point looks like a phone to the switch it linked to. The FXO crossing point closes the loop to specify off-hook. The switch always provides a battery so there is no disconnect management from the switch side. Since a switch expects a phone user (example of an FXO interface) to hang up the phone when the call is finished (on either side), it also hope the FXO port on the router to hang-up. This "human involvement" is not built into the router. The FXO port hopes the switch to tell it when to hang-up (or remove the battery to specify on-hook). Because of this, there is no warranty that a near-end or far-end FXO port cut-Off the call once either end of the call hangs-up.

The most common sign of this problem are phones that continue to ring when the caller has cleared, or FXO ports that stay busy after the previous call should have been finished.

## 7. CONCLUSION

Call routing via analog Voice Ports helps VOIP/PSTN [1] call connecting vice versa, here Router voice ports plays a key role in the converting the signaling and as well as the voice and data in between Packet switched network devices and Circuit switched network devices. And there are some issues with the analog voice ports (FXO) connection points to the router.

## 8. REFERENCES

[1] Webopedia, “The Difference between VoIP and PSTN Systems”, Available: http://www.webopedia.com/DidYouKnow/Internet/2008/VoIP_POTS_Difference_Between.asp

[2] G. Krzysztof, K. Aleksander, W. Jozef, N. Krzysztof, “Testbed analysis of video and VoIP transmission performance in IEEE 802.11 b/g/n networks,” Telecommunication Systems, SpringerNetherlands, vol. 48, no 3-4, pp. 247-260, December 2011.

[3] Freeny, S.L.; Bell Telephone Labs., N.J.; Kieburtz, R.; Mina, K.; Tewksbury, S.K.,” Systems Analysis of a TDM-FDM Translator/Digital A-Type Channel Bank”, Communication Technology, IEEE Transactions on (Volume:19, Issue: 6 ), 06 January 2003.

[4] YouhongLu; Critical Commun. Syst. Div., Bosch Commun. Syst., Burnsville, MN; Guodong Shi; Youlian Zhu; Weige Tao “Echo Cancellation of FXO line card expansion“, Publisher:IEEE,Page(s): 3590 - 3594,25-27 June 2008.

[5] Jo, K.Y.; US Defense Commun. Eng. Center, Reston, VA, USA; Thomas, J.E., “Interconnection of heterogeneous defense data networks”, IEEE

[6] Sugiyama, A.; Yamaji, T.,”An efficient multichannel line echo canceler algorithm for PSTN and VoIP/VoDSL applications”, Acoustics, Speech, and Signal Processing, 2001.