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Using Analog E&M Ports to Interface to Overhead Paging Systems

Introduction

This document details the background theory and configurations that allow a router Ear and Mouth (E&M) voice port to interface to an overhead paging system.

Prerequisites

Requirements

Before attempting this configuration, please ensure that you meet these requirements:

- Analog E&M signaling theory and router voice port operation
- Cisco IOS® configuration and Cisco CallManager configuration

Components Used

The information in this document is based on these software and hardware versions:

- Cisco 2610 router
- Cisco IOS version 12.2.7a with an IP Plus feature set
- NM−2V voice carrier card and a VIC−2E/M (E&M Voice Interface Card (VIC))
- External paging amplifier

The information presented in this document was created from devices in a specific lab environment. All of the devices used in this document started with a cleared (default) configuration. If you are working in a live network, ensure that you understand the potential impact of any command before using it.

Conventions

For more information on document conventions, see the Cisco Technical Tips Conventions.

Background Information

Many sites with an existing PBX also have a paging system allowing users to call an extension on the PBX that forwards the audio broadcast to overhead loudspeakers. This concept is useful in workshops, parking lots, and open plan areas where a called party is not near a telephone handset. PBX manufacturers may provide dedicated line cards that interface with external paging amplifiers. These PBX paging cards have an isolated audio output (to prevent faults on the paging amplifier resulting in damage to the PBX), and a control or relay output that is used to activate the paging amplifier.

As IP−based PBXs and Voice over IP (VoIP) networks become more common, the need to integrate voice−equipped routers into legacy installations is apparent. New paging systems are available using loop start trunks that interface directly to PBX extension ports and have Voice Operated Relays (VOX) that control the paging amplifiers. Many customers do not wish to purchase new interfaces or replace existing hardware as they transition to IP−based systems. Fortunately, Cisco voice products are flexible enough to cover many of these cases. This document details the method of using analog E&M voice interfaces to provide an audio and control output to an interface with an external paging amplifier. Many dedicated PBX paging cards are based on normal PBX E&M line cards.

The difference between a conventional two wire telephone interface such as Foreign Exchange Station or Office (FXS or FXO) and an E&M interface is that the E&M interface will have wires that pass the audio signals plus additional wires to act as an input (to sense an incoming call) or an output (to indicate an outgoing call). These control leads are normally called the E lead (input) and the M lead (output). Depending on the type of E&M interface, the signaling leads could be controlled by connecting them to the ground, switching a negative 48Volt DC source, or completing a current loop between the two devices.

E&M interfaces normally will have the option of two or four wire operation. Rather than referring to the total number of physical connections on the port, two or four wire operation relates to the how audio is passed between the devices. Two wire operation means the transmitting and receiving audio signals are passed through a single pair of wires (one pair equals two wires). Four wire operation separates the direction of the signal and uses one pair for transmitting and another pair for receiving audio.

By default, Cisco E&M ports will use wink start signaling. Wink start operation dictates that when the voice port goes off hook (E leads the state changes from on hook to off hook), it expects to receive a 200 millisecond wink (on hook/off hook/on hook) transition on the M lead as the acknowledgement allows digits to be sent. The E lead stays in the off hook condition for the duration of the call.

A simpler form of E&M signaling is called immediate start. In this mode, when the voice port goes off hook (E lead changes from on hook to off hook), there is a brief pause. The router then sends the digits without any acknowledgment from the external device. As with wink start, the E lead stays in the off hook condition for the duration of the call.

When the voice port is configured as two wire with immediate start, an outgoing call (from the IP side toward the external device) will make the E lead change from an open circuit, to a short circuit, to the ground. Control leads can be used to switch a relay or push−to−talk control and the audio path opened on the transmit/receive (T/R) leads.

Configure

In this example, the customer has a requirement to interface an older paging system into a new Cisco CallManager installation. They will be using a Cisco 2610 router with an E&M VIC. The paging amplifier has an audio input and an external push−to−talk control input. The following interface pinouts are used between the router E&M voice port and the amplifier:

```
T1 (Pin 4) ---------------- Microphone audio input
R1 (Pin 5) ---------------- Microphone audio input
E lead (Pin 7) ------------- Push-to-talk control input
Ground (Pin 8) ------------ Push-to-talk control input
```

The router E&M voice port needs to be configured as a two wire, type 5, with immediate start operation.

The Cisco CallManager needs the Cisco 2610 router to be configured as a H323 gateway device. The extension number for the paging port is defined under the Cisco CallManager Route Pattern Configuration page, pointing to the Cisco 2610 H323 gateway.

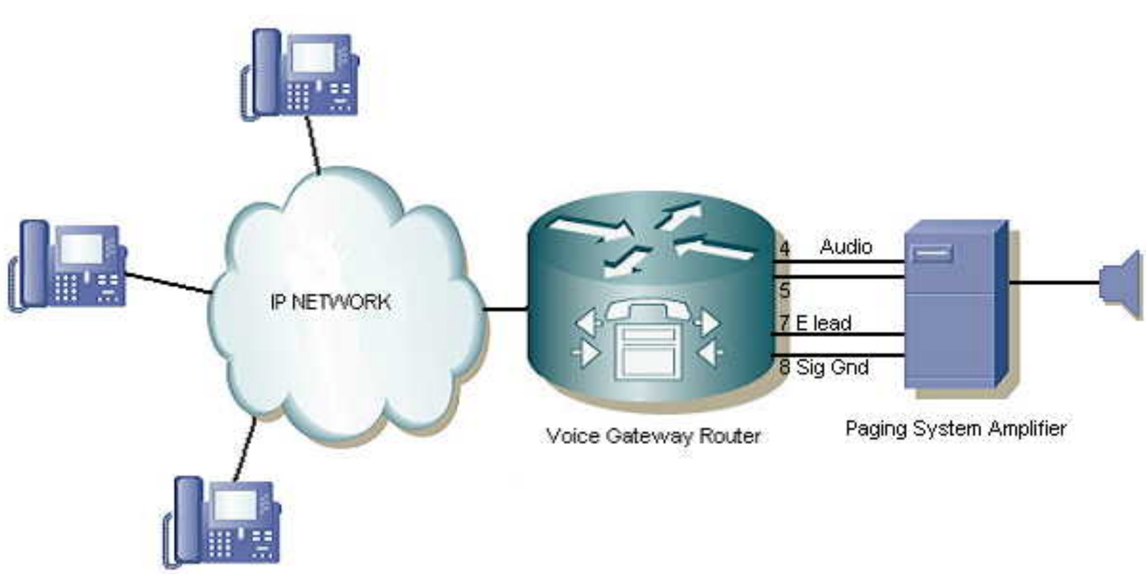
When the number for the paging system is dialed, a VoIP call is made between the IP handset to the E&M port on the gateway router. The voice port goes off hook. This is indicated by the E lead on pin 7 going from open circuit to closed circuit (with respect to the ground on Pin 8). This off hook condition activates the pager system's control input and the audio is sent on pins 4 and 5 of the voice port

Assuming a paging access number of 5555, the gateway router voice port and dial peer configuration would look similar to that found in the Configurations section of this document.

Note: To find additional information on the commands used in this document, use the Command Lookup Tool (registered customers only) .

Network Diagram

This document uses this network setup:



Configurations

This document uses these configurations:

Router voice port and dial peer configuration

```
!
voice-port 1/0/0
 operation 2-wire

!--- Only use pins 4 and 5 for audio.

 type 5

!--- Type 5 operation, the most basic mode.

 signal immediate

!--- Immediate start operation.

 auto-cut-through

!--- Send immediate answer back to the VoIP network.

!

!
dial-peer voice 5555 pots
destination-pattern 5555

!--- Match on 5555 access code.

port 1/0/0

!--- Send the call on E&M port 1/0/0.

forward-digits none

!--- Do not send any digits out of the port.

!
```

Verify

This section provides information you can use to confirm your configuration is working properly.

Certain **show** commands are supported by the Output Interpreter Tool (registered customers only) , which allows you to view an analysis of **show** command output.

To confirm the configuration and settings of the voice port, the **show voice port** <***card/slot/port***> command provides information about the status of the router's voice ports, as shown in the following example:

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

recEive And transMit 1/0/0 Slot is 1, Sub-unit is 0, Port is 0
   Type of VoicePort is E&M
   Operation State is DORMANT
   Administrative State is UP
   No Interface Down Failure
```

```
  Description is not set
  Noise Regeneration is enabled
  Non Linear Processing is enabled
  Non Linear Mute is disabled
  Non Linear Threshold is −21 dB
  Music On Hold Threshold is Set to −38 dBm
  In Gain is Set to 0 dB
  Out Attenuation is Set to 0 dB
  Echo Cancellation is enabled
  Echo Cancellation NLP mute is disabled
  Echo Cancellation NLP threshold is −21 dB
  Echo Cancel Coverage is set to 8 ms
  Playout−delay Mode is set to default
  Playout−delay Nominal is set to 60 ms
  Playout−delay Maximum is set to 200 ms
  Playout−delay Minimum mode is set to default, value 40 ms
  Playout−delay Fax is set to 300 ms
  Connection Mode is normal
  Connection Number is not set
  Initial Time Out is set to 10 s
  Interdigit Time Out is set to 10 s
  Call Disconnect Time Out is set to 60 s
  Ringing Time Out is set to 180 s
  Wait Release Time Out is set to 30 s
  Companding Type is u−law
  Region Tone is set for US

Analog Info Follows:
  Currently processing none
  Maintenance Mode Set to None (not in mtc mode)
  Number of signaling protocol errors are 0
  Impedance is set to 600r Ohm
  Station name None, Station number None
  Translation profile (Incoming):
  Translation profile (Outgoing):

Voice card specific Info Follows:
  Operation Type is 2−wire
  E&M Type is 1
  Signal Type is immediate
  Dial Out Type is dtmf
  In Seizure is inactive
  Out Seizure is inactive
  Digit Duration Timing is set to 100 ms
  InterDigit Duration Timing is set to 100 ms
  Pulse Rate Timing is set to 10 pulses/second
  InterDigit Pulse Duration Timing is set to 750 ms
  Clear Wait Duration Timing is set to 400 ms
  Wink Wait Duration Timing is set to 200 ms
  Wait Wink Duration Timing is set to 550 ms
  Wink Duration Timing is set to 200 ms
  Delay Start Timing is set to 300 ms
  Delay Duration Timing is set to 2000 ms
  Dial Pulse Min. Delay is set to 140 ms
  Percent Break of Pulse is 60 percent
  Auto Cut−through is disabled
  Dialout Delay is 300 ms
  Paging_Router#
```

Troubleshoot

This section provides information you can use to troubleshoot your configuration.

Monitoring the Voice Port Signaling and Audio Outputs

Follow these instructions to monitor the voice port signaling and audio outputs:

1. Operation of the E&M port is best verified using a multimeter to measure continuity (cable test, ohms or resistance setting).

   If a voice call is placed to the E&M port, the E lead (pin 7) is switched from open circuit to ground (pin 8), and the meter will show a change from high resistance to zero resistance. The E lead switching to ground can be seen by the paging amplifier as a push−to−talk signal. It then broadcasts the audio signal coming in on the audio pairs. The following is an example of a typical digital multimeter:

2. The voice port audio output can be checked with a telephone technician's test set (sometimes called a 'Butt Set" or 'Buttinski').

   Any outgoing audio can be heard on the test set's earpiece, therefore confirming if the router is sending a signal to the paging amplifier. The monitor leads of the Butt set would be clipped across the T and R wires (pins 4 and 5) on the router voice port. The following is an example of a typical telephone test handset:

For more information on E&M interfaces and signaling, refer to the Technical Assistance Center Analog Signaling (E & M, DID, FXS, FXO) Technical Support Page.

Related Information

* **Conferencing Software**
* **Voice Technologies**
* **Voice, Telephony and Messaging Devices**
* **Voice Software**
* **Voice, Telephony and Messaging Technical Support eLearning Solutions**
* **Recommended Reading: Troubleshooting Cisco IP Telephony Cisco Press, ISBN 1587050757**
* **Technical Support − Cisco Systems**
