VoIP Configuration Examples

This section uses four different scenarios to demonstrate how to configure Voice over IP (VoIP). The actual VoIP configuration procedure depends on the topology of your voice network. The following configuration examples should give you a starting point, but you will need to customize them to reflect your network topology.

Configuration procedures are supplied for the following scenarios:

- FXS-to-FXS Connection Using RSVP, page C-1
- Linking PBX Users with E&M Trunk Lines, page C-7
- FXO Gateway to PSTN, page C-10
- FXO Gateway to PSTN (PLAR Mode), page C-12

FXS-to-FXS Connection Using RSVP

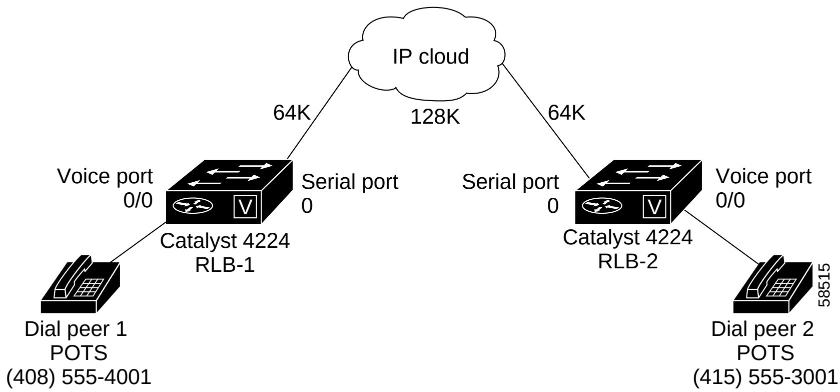
The following example shows how to configure VoIP for a simple FXS-to-FXS connection.

In this scenario, a very small company with two offices decides to integrate VoIP into its existing IP network. One basic telephony device is connected to Catalyst 4224 RLB-1; therefore, Catalyst 4224 RLB-1 is configured for one POTS dial peer and one VoIP dial peer. Catalyst 4224 RLB-w and Catalyst 4224 RLB-e establish the WAN connection between the two offices. Because one POTS telephony device is connected to Catalyst 4224 RLB-2, it is also configured for one POTS dial peer and one VoIP dial peer.

In this scenario, only the calling end (Catalyst 4224 RLB-1) is requesting RSVP.

Figure C-1 illustrates the topology of this FXS-to-FXS connection example.

Figure C-1 FXS-to-FXS Connection (Example)



Configuration for Catalyst 4224 RLB-1

```
hostname RLB-1

! Create voip dial-peer 2
dial-peer voice 2 voip

! Define its associated telephone number and IP address
 destination-pattern 14155553001
 sess-target ipv4:40.0.0.1

! Request RSVP
 req-qos controlled-load

! Create pots dial-peer 1
dial-peer voice 1 pots

! Define its associated telephone number and voice port
 destination-pattern 14085554001
 port 0/0

! Configure serial interface 0
interface Serial0
 ip address 10.0.0.1 255.0.0.0
 no ip mroute-cache

! Configure RTP header compression
 ip rtp header-compression
 ip rtp compression-connections 25

! Enable RSVP on this interface
 ip rsvp bandwidth 48 48
 fair-queue 64 256 36
 clockrate 64000

router igrp 888
 network 10.0.0.0
 network 20.0.0.0
 network 40.0.0.0
```

Configuration for Catalyst 4224 RLB-w

```
hostname RLB-w

! Configure serial interface 0
interface Serial0
 ip address 10.0.0.2 255.0.0.0

! Configure RTP header compression
 ip rtp header-compression
 ip rtp compression-connections 25

! Enable RSVP on this interface
 ip rsvp bandwidth 96 96
 fair-queue 64 256 3

! Configure serial interface 1
interface Serial1
 ip address 20.0.0.1 255.0.0.0

! Configure RTP header compression
 ip rtp header-compression
 ip rtp compression-connections 25

! Enable RSVP on this interface
 ip rsvp bandwidth 96 96
 fair-queue 64 256 3

! Configure IGRP
router igrp 888
 network 10.0.0.0
 network 20.0.0.0
 network 40.0.0.0
```

Configuration for Catalyst 4224 RLB-e

```
hostname RLB-e

! Configure serial interface 0
interface Serial0
 ip address 40.0.0.2 255.0.0.0

! Configure RTP header compression
 ip rtp header-compression
 ip rtp compression-connections 25

! Enable RSVP on this interface
 ip rsvp bandwidth 96 96
 fair-queue 64 256 3

! Configure serial interface 1
interface Serial1
 ip address 20.0.0.2 255.0.0.0

! Configure RTP header compression
 ip rtp header-compression
 ip rtp compression-connections 25

! Enable RSVP on this interface
 ip rsvp bandwidth 96 96
 fair-queue 64 256 3
 clockrate 128000

! Configure IGRP
router igrp 888
 network 10.0.0.0
 network 20.0.0.0
 network 40.0.0.0
```

Configuration for Catalyst 4224 RLB-2

```
hostname RLB-2

! Create pots dial-peer 2
dial-peer voice 2 pots

! Define its associated telephone number and voice-port
 destination-pattern 14155553001
 port 0/0

! Create voip dial-peer 1
dial-peer voice 1 voip

!Define its associated telephone number and IP address
 destination-pattern 14085554001
 sess-target ipv4:10.0.0.1

! Configure serial interface 0
interface Serial0
 ip address 40.0.0.1 255.0.0.0
 no ip mroute-cache

! Configure RTP header compression
 ip rtp header-compression
 ip rtp compression-connections 25

! Enable RSVP on this interface
 ip rsvp bandwidth 96 96
 fair-queue 64 256 3
 clockrate 64000

! Configure IGRP
router igrp 888
 network 10.0.0.0
 network 20.0.0.0
 network 40.0.0.0
```

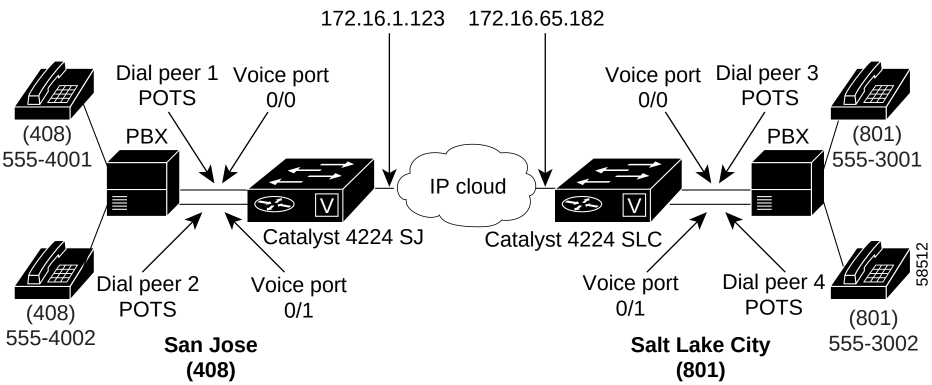
Linking PBX Users with E&M Trunk Lines

The following example shows how to configure VoIP to link PBX users with E&M trunk lines.

In this scenario, a company decides to connect two offices: one in San Jose, California, and the other in Salt Lake City, Utah. Each office has an internal telephone network using a PBX connected to the voice network by an E&M interface. Both the Salt Lake City and the San Jose offices are using E&M Port Type II, with four-wire operation and Immediate Start signaling. Each E&M interface connects to the Catalyst 4224 using two voice interface connections. Users in San Jose dial *801-555* and then the extension number to reach a destination in Salt Lake City. Users in Salt Lake City dial *408-555* and then the extension number to reach a destination in San Jose.

Figure C-2 illustrates the topology of this scenario.

Figure C-2 Linking PBX Users with E&M Trunk Lines (Example)



Note This example assumes that the company has already established a working IP connection between its two remote offices.

Router San Jose Configuration

```
hostname router SJ

!Configure pots dial-peer 1
dial-peer voice 1 pots
 destination-pattern 1408555....
 port 0/0

!Configure pots dial-peer 2
dial-peer voice 2 pots
 destination-pattern 1408555....
 port 0/1

!Configure voip dial-peer 3
dial-peer voice 3 voip
 destination-pattern 1801555....
 session target ipv4:172.16.65.182
 ip precedence 5

!Configure the E&M interface
voice-port 0/0
 signal immediate
 operation 4-wire
 type 2

voice-port 0/1
 signal immediate
 operation 4-wire
 type 2

!Configure the serial interface 0
interface serial0
 ip address 172.16.1.123
 no shutdown
```

Router Salt Lake City Configuration

```
hostname router SLC

!Configure pots dial-peer 3
dial-peer voice 3 pots
 destination-pattern 1801555....
 port 0/0

!Configure pots dial-peer 4
dial-peer voice 4 pots
 destination-pattern 1801555....
 port 0/1

!Configure voip dial-peer 1
dial-peer voice 1 voip
 destination-pattern 1408555....
 session target ipv4:172.16.1.123
 ip precedence 5

!Configure the E&M interface
voice-port 0/0
 signal immediate
 operation 4-wire
 type 2

voice-port 0/1
 signal immediate
 operation 4-wire
 type 2

!Configure the serial interface 0
interface serial0
 ip address 172.16.65.182
 no shutdown
```

Note PBXs should be configured to pass all dual tone multifrequency (DTMF) signals to the router. Cisco recommends that you do not configure, store, or forward tone.

Note If you change the gain or the telephony port, make sure that the telephony port still accepts DTMF signals.

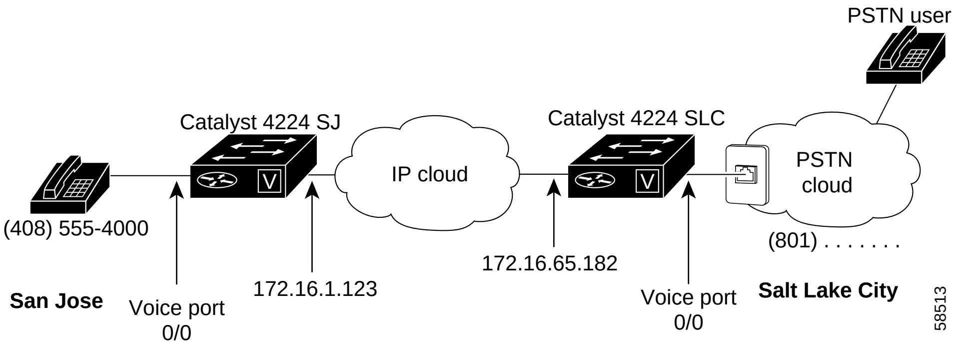
FXO Gateway to PSTN

Foreign Exchange Office (FXO) interfaces provide a gateway from the VoIP network to the analog public switched telephone network (PSTN) or to a PBX that does not support Ear and Mouth (E&M) signaling.

In this scenario, users connected to Catalyst 4224 SJ in San Jose, California, can reach PSTN users in Salt Lake City, Utah, via Catalyst 4224 SLC. Router SLC in Salt Lake City is connected directly to the PSTN through an FXO interface.

Figure C-3 illustrates the topology of this scenario.

Figure C-3 FXO Gateway to PSTN (Example)



Note This example assumes that the company has already established a working IP connection between its two remote offices.

Router San Jose Configuration

```
hostname router SJ

! Configure pots dial-peer 1
dial-peer voice 1 pots
 destination-pattern 14085554000
 port 0/0

! Configure voip dial-peer 2
dial-peer voice 2 voip
 destination-pattern 1801.......
 session target ipv4:172.16.65.182
 ip precedence 5

! Configure serial interface 0
interface serial0
 clock rate 2000000
 ip address 172.16.1.123
 no shutdown
```

Router Salt Lake City Configuration

```
hostname router SLC

! Configure pots dial-peer 1
dial-peer voice 1 pots
 destination-pattern 1801.......
 port 0/0

! Configure voip dial-peer 2
dial-peer voice 2 voip
 destination-pattern 14085554000
 session target ipv4:172.16.1.123
 ip precedence 5

! Configure serial interface 0
interface serial0
 ip address 172.16.65.182
 no shutdown
```

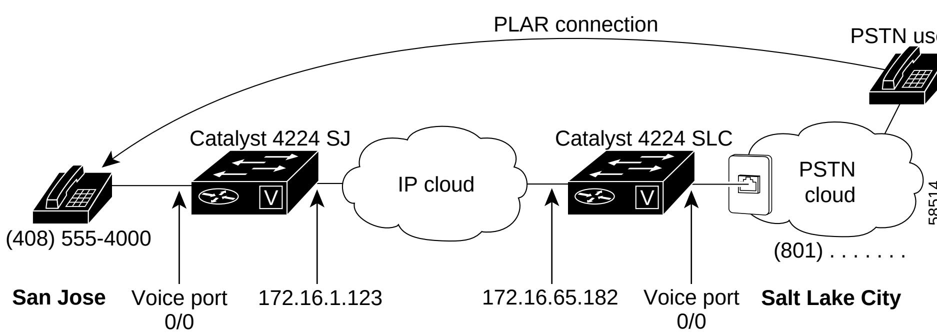
FXO Gateway to PSTN (PLAR Mode)

The following scenario shows an FXO gateway to PSTN connection in PLAR mode.

In this scenario, PSTN users in Salt Lake City, Utah, can dial a local number and establish a private line connection in a remote location. As in the previous scenario, Catalyst 4224 SLC in Salt Lake City is connected directly to the PSTN through an FXO interface.

Figure C-4 illustrates the topology of this scenario.

Figure C-4 FXO Gateway to PSTN (PLAR Mode) (Example)



Note This example assumes that the company has already established a working IP connection between its two remote offices.

Router San Jose Configuration

```
hostname router SJ

! Configure pots dial-peer 1
dial-peer voice 1 pots
 destination-pattern 14085554000
 port 0/0

! Configure voip dial-peer 2
dial-peer voice 2 voip
 destination-pattern 1801.......
 session target ipv4:172.16.65.182
 ip precedence 5

! Configure the serial interface 0
interface serial0
 clock rate 2000000
 ip address 172.16.1.123
 no shutdown
```

Router Salt Lake City Configuration

```
hostname router SLC

! Configure pots dial-peer 1
dial-peer voice 1 pots
 destination-pattern 1801.......
 port 0/0

! Configure voip dial-peer 2
dial-peer voice 2 voip
 destination-pattern 14085554000
 session target ipv4:172.16.1.123
 ip precedence 5

! Configure the voice port
voice port 0/0
connection plar 14085554000

! Configure the serial interface 0
interface serial0
 ip address 172.16.65.182
 no shutdown
```