Configuring Voice Functionality

This chapter provides information about configuring voice functionality on the Cisco 880 Series Integrated Services Routers (ISRs). The following ISRs have voice gateway capability:

- C881SRST and C888SRST: 4 FXS ports and 1 voice backup port
  - The C881SRST ISR has an FXO voice backup port.
  - The C888SRST ISR has a BRI voice backup port.
- C881-V has 4FXS ports, 2 BRI ports, and 1 backup FXO port
- C887VA-V and C887VA-V-W has 4FXS ports and 2 BRI ports.

Voice Ports

Analog voice ports (Foreign Exchange Station (FXS) ports) connect routers in packet-based networks to 2-wire or 4-wire analog circuits in telephony networks. Two-wire circuits connect to analog telephone or fax devices, and four-wire circuits connect to PBXs.

Digital voice ports are ISDN basic rate interface (BRI) ports.

Analog and Digital Voice Port Assignments

Analog and digital voice port assignments vary by model number. Table 1: Voice Port Assignments for Cisco 880 series ISRs, on page 2 lists the Cisco 880 series ISRs and their voice port assignments.

Table 1: Voice Port Assignments for Cisco 880 series ISRs

Model Number	Digital (BRI) Port Numbers	Analog (FXS) Port Numbers	Voice Backup Port Number
C881SRST	—	0–3	4 (FXO port)
C888SRST	—	0–3	4 (BRI port)
C881-V	2	4	1 (FXO port)
C887VA-V	2	4	—
C887VA-V-W	2	4	—

Voice Port Configuration

To configure analog and digital voice ports, see the following documents:

- Configuring Analog Voice Ports
- Basic ISDN Voice Interface Configuration

Call Control Protocols

SIP

Session Initiation Protocol (SIP) is a peer-to-peer, multimedia signaling protocol developed in the IETF (IETF RFC 2543). Session Initiation Protocol is ASCII-based. It resembles HTTP, and it reuses existing IP protocols (such as DNS and SDP) to provide media setup and teardown. See the Cisco IOS SIP Configuration Guide for more information.

For router configuration information under SIP, see the Basic SIP Configuration chapter of the Cisco IOS SIP Configuration Guide, Release 12.4T.

Cisco 880 Series ISR voice gateways provide voice security through SIP enhancements within the Cisco IOS Firewall. SIP inspect functionality (SIP packet inspection and detection of pin-hole openings) is provided, as well as protocol conformance and application security. The user is given more granular control on the policies and security checks applied to SIP traffic, and capability to filter out unwanted messages. For more information, see "Cisco IOS Firewall: SIP Enhancements: ALG and AIC".

MGCP

Media Gateway Control Protocol (MGCP) RFC 2705 defines a centralized architecture for creating multimedia applications, including Voice over IP (VoIP). See the Cisco IOS MGCP and Related Protocols Configuration Guide for more information.

Cisco 880 series voice gateway ISRs are configured primarily as residential gateways (RGWs) under MGCP. For residential gateway configuration information, see the Configuring an RGW section of the Basic MGCP Configuration chapter of the Cisco IOS MGCP and Related Protocols Configuration Guide.

H.323

International Telecommunications Union Recommendation H.323 defines a distributed architecture for creating multimedia applications, including Voice over IP.

For router configuration information, see the Configuring H.323 Gateways chapter of the Cisco IOS H.323 Configuration Guide, Release 12.4T.

Dial Peer Configuration

Configuring dial peers is the key to implementing dial plans and providing voice services over an IP packet network. Dial peers are used to identify call source and destination endpoints and to define the characteristics applied to each call leg in the call connection. For router configuration information, see Dial Peer Configuration on Voice Gateway Routers.

Other Voice Features

Real-Time Transport Protocols

Real-Time Transport Protocol (RTP) provides end-to-end network transport functions for applications that transmit real-time data.

Cisco Real-Time Transport Protocol (cRTP) uses the RTP protocol to forward Cisco-proprietary payload types.

Secure Real-Time Transport Protocol (SRTP) defines an RTP profile providing encryption, authentication, and replay protection.

RTP is used primarily with DTMF relay and is configured under dial peer configuration. For information on configuring RTP payload types, see the Dual-Tone Multifrequency Relay section of Dial Peer Configuration on Voice Gateway Routers.

For information on configuring SRTP on SIP-controlled platforms, see the Configuring SIP Support for SRTP chapter of the Cisco IOS SIP Configuration Guide, Release 4T.

For configuring RTP on MGCP-controlled platforms, see the Configuring an RGW section of the Basic MGCP Configuration chapter of the Cisco IOS MGCP and Related Protocols Configuration Guide.

Dual Tone Multi Frequency Relay

Using Dial Tone Multi Frequency (DTMF) Relay the local VoIP gateway listens for DTMF digits and sends the digits uncompressed as either RTP packets or H.245 packets to the remote VoIP gateway. The remote VoIP gateway regenerates the DTMF digits. This methodology prevents digit loss due to compression. For

information on configuring DTMF Relay, see the Dual-Tone Multifrequency Relay section of Dial Peer Configuration on Voice Gateway Routers.

For information on configuring DTMF that is specific to call control protocols, see the following:

- Configuring SIP DTMF Features
- Configuring DTMF Relay (H.323)
- Configuring Global MGCP Parameters

CODECs

The following CODECs are supported by the Cisco 880 series voice gateway routers.

- G.711 (a-law and mu-law)
- G.726
- G.729, G.729A, G.729B, G.729AB

For information on CODECs, see the following:

- Dial Peer Configuration Examples appendix of Dial Peer Configuration on Voice Gateway Routers.
- Cisco IOS SIP Configuration Guide, Release 4T
- Cisco IOS H.323 Configuration Guide

SCCP-Controlled Analog Ports with Supplementary Features

Cisco 880 series voice gateway ISRs support the Cisco Skinny Client Control Protocol (SCCP) that supplies supplementary features on analog voice ports that are controlled by Cisco Unified Communications Manager or by a Cisco Unified Communications Manager Express system. Supported features include:

- Audible message waiting indication
- Call forwarding options
- Call park/pickup options
- Call transfer
- Call waiting
- Caller ID
- 3-party conference calls
- Redial
- Speed dial options

For more information on the features supported and their configuration, see SCCP Controlled Analog (FXS) Ports with Supplementary Features in Cisco IOS Gateways .

Fax Services

The Cisco 880 series voice gateway ISRs support the following fax services:

Fax Pass-Through

Fax Pass-Through is the simplest way of transmitting faxes over IP, although it is not as reliable as Cisco Fax Relay. See the Configuring Fax Pass-Through chapter of the Cisco IOS Fax, Modem, and Text Services over IP Configuration Guide for more information.

Cisco Fax Relay

Cisco Fax Relay is a Cisco proprietary fax method that is turned on by default. Cisco Fax Relay allows the relay of a T.30 modulated signal across IP gateways in real-time on H.323 or SIP networks. See the Configuring Cisco Fax Relay chapter of the Cisco IOS Fax, Modem, and Text Services over IP Configuration Guide for more information.

T.37 Store-and-Forward Fax

The T.37 Store-and-Forward Fax mechanism allows a gateway to store and forward fax messages on H.323 or SIP networks. See the Configuring T.37 Store-and-Forward Fax chapter of the Cisco IOS Fax, Modem, and Text Services over IP Configuration Guide for more information.

T.38 Fax Relay

The T.38 Fax Relay provides an ITU-standard mechanism for real-time relay of fax signals. Gateway-controlled T.38 Fax Relay is available on MGCP networks. See the Configuring T.38 Fax Relay chapter of the Cisco IOS Fax, Modem, and Text Services over IP Configuration Guide for more information.

Unified Survival Remote Site Telephony

Cisco 880 Series voice gateway ISRs with Unified Survival Remote Site Telephony (SRST) include the following:

- Cisco C881SRST
- Cisco C888SRST

Unified SRST automatically detects a failure in the network and initializes the process of auto configuring the router. Unified SRST provides redundancy for the IP and FXS phones to ensure that the telephone system remains operational.

All the IP phones and analog phones connected to a telecommuter site are controlled by the headquarters office call control system, which uses Cisco Unified Communications Manager. During a WAN failure, the telecommuter router allows all the phones to reregister to the headquarter in SRST mode, allowing all inbound

and outbound dialing to be routed off to the PSTN (on a backup Foreign Exchange Office (FXO) or BRI port). Upon restoration of WAN connectivity, the system automatically returns communication to the primary Cisco Unified Communications Manager cluster.

Direct Inward Dialing (DID) is supported on the Cisco 880 series SRST voice gateway ISRs.

For general Unified SRST information, see the Cisco Unified SRST System Administrator Guide. Cisco Unified SRST is described in the Overview chapter.

- For information on how the H.323 and MGCP call control protocols relate to SRST, see the following sections of the Overview chapter in the Cisco Unified SRST System Administrator Guide.

For SIP-specific SRST information, see the Cisco Unified SRST System Administrator Guide. To configure SIP SRST features, see the 4.1 Features chapter.

Verification of Voice Configuration

Use the following procedures to verify voice port configurations:

- Verifying Analog and Digital Voice-Port Configurations
- Cisco IOS Voice Port Configuration Guide, Verify BRI Interfaces

To verify, monitor, and maintain SRST, see Monitoring and Maintaining Cisco Unified SRST.