Cisco SRS Telephony Configuration

This chapter explains the required and optional tasks for configuring Cisco SRS Telephony.

Contents

Prerequisites for SRS Telephony Configuration

The following are prerequisites that must be met before configuration:

- The Cisco SRS Telephony system must be configured with the correct IP phones, platforms, and Cisco CallManager (CCM) version. See the "Cisco IP Phone, Platform, Cisco CallManager, Signal, and Switch Support" section on page 1-4
- The correct memory requirements must be installed. Memory requirements are dependent on the platform and the number of supported Cisco IP phones. See the "Specifications" section on page 1-7 for details.
- The correct IP phone firmware must be installed. With CCM V3.1 and later, Cisco SRS Telephony is standard in IP phone firmware. Refer to CCM documentation for firmware requirements. For CCM 3.0.5, you must use the firmware version P003E302 or P004E302.
- IP routing must be enabled.
- The SRS Telephony router must be configured as the default router for the Cisco IP phones.

Note You must purchase a feature license to turn on this new feature. You also need an account on Cisco.com to access the Cisco IP phone load versions.

How to Configure Cisco SRS Telephony

This section contains the following procedures:

- Configuring SRS Telephony on Routers to Support IP Phone Functions, page 2-2 (required)
- Configuring Cisco SRS Telephony Optional Settings, page 2-4 (optional)
- Configuring Cisco SRS Telephony for Unity Voice-Mail Integration, page 2-9 (optional)

Configuring SRS Telephony on Routers to Support IP Phone Functions

Tip When the SRS Telephony feature is enabled, Cisco IP phones do not need to be reconfigured while in Cisco CallManager fallback mode because phones retain the same configuration that was used with Cisco CallManager.

To configure SRS Telephony on the routers to support the Cisco IP phone functions, use the following commands beginning in global configuration mode:

SUMMARY STEPS

1. **call-manager-fallback**
2. **ip source-address** *ip-address* [**port** *port*] [**any-match** | **strict-match**]
3. **max-dn** *max-directory-numbers*
4. **max-ephones** *max-phones*
5. **limit-dn** {**7910** | **7940** | **7960**} *max-lines*

DETAILED STEPS

	Command	Purpose

Step 1	`call-manager-fallback`
	
	Example:
	`Router(config)#`
		Enables SRS Telephony feature support and enters Cisco CallManager fallback mode.

Step 2	**ip source-address** *ip-address* [**port** *port*] [**any-match** | **strict-match**] **Example:** `Router(config-cm-fallback)# ip source-address 10.6.21.4 port 2002 strict-match`	Enables the router to receive messages from the Cisco IP phones through the specified IP addresses and provides for strict IP address verification. The default port number is 2000.

	Command	Purpose
Step 3	**max-dn** *max-directory-numbers* **Example:** `Router(config-cm-fallback)# max-dn 12`	Sets the maximum number of directory numbers or virtual voice ports that can be supported by the router. The default is 0. The maximum number is platform dependent. See the "Specifications" section on page 1-7 for further details. **Note** You must reboot the router in order to reduce the limit of the directory numbers or virtual voice ports after the maximum allowable number is configured.
Step 4	**max-ephones** *max-phones* **Example:** `Router(config-cm-fallback)# max-ephones 24`	Configures the maximum number of Cisco IP phones that can be supported by the router. The default is 0. The maximum number is platform dependent. See "Specifications" section on page 1-7 for further details. **Note** You must reboot the router in order to reduce the limit of the directory numbers or virtual voice ports after the maximum allowable number is configured.
Step 5	**limit-dn** {**7910** | **7940** | **7960**} *max-lines* **Example:** `Router(config-cm-fallback)# limit-dn 7910 2`	Limits the directory number lines on Cisco IP phones during CallManager fallback mode. **Note** You must configure this command during initial SRS Telephony router configuration, before any phone actually registers with the SRS Telephony router. However, you can modify the number of lines at a later time. The setting for maximum lines is from 1 to 6. The default number of maximum directory lines is set to 6. If there is any active phone with last line number greater than this limit, warning information is displayed for phone reset.

Verifying Cisco SRS Telephony

To verify that the SRS Telephony feature is enabled, perform the following steps:

Step 1 Enter the **show run** command to verify the configuration.

Step 2 Enter the **show call-manager-fallback all** command to verify that SRS Telephony feature is enabled.

Step 3 Use the Settings display on the Cisco IP phones in your network to verify that the default router IP address on the phones matches the IP address of the SRS Telephony router.

Step 4 To temporarily block the TCP port 2000 Skinny Client Control Protocol (SCCP) connection for one of the Cisco IP phones to force the Cisco IP phone to lose its connection to the Cisco CallManager and register with the SRS Telephony router, perform the following steps:

a. Use the appropriate IP **access-list** command to temporarily disconnect a Cisco IP phone from the Cisco CallManager.

During a WAN connection failure, when SRS Telephony is enabled, Cisco IP phones display a message informing you that they are operating in Cisco CallManager fallback mode. The Cisco IP Phone 7960 and Cisco IP Phone 7940 display a "CM Fallback Service Operating" message and the Cisco IP Phone 7910 displays a "CM Fallback Service" message when operating in Cisco CallManager fallback mode. When the Cisco CallManager is restored, the message goes away and full Cisco IP phone functionality is restored.

b. Enter the **no** form of the appropriate **access-list** command to restore normal service for the phone.

c. Use the **debug ephone register** command to observe the registration process of the Cisco IP phone on the SRS Telephony router.

d. Use the **show ephone** command to display the Cisco IP phones that have registered to the SRS Telephony router.

Troubleshooting Tips

To troubleshoot the SRS Telephony feature, perform the following steps:

Step 1 To set keepalive debugging for the Cisco IP phone, use the **debug ephone keepalive** command.

Step 2 To set registration debugging for the Cisco IP phone, use the **debug ephone register** command.

Step 3 To set state debugging for the Cisco IP phone, use the **debug ephone state** command.

For further debugging, use the debug commands in the *Cisco IOS Debug Command Reference*.

Configuring Cisco SRS Telephony Optional Settings

Although following baseline Cisco SRS Telephony settings are not required, they are worth considering for possible configuration of the following features:

- Unmatched dial-peer routing
- Cisco IP phone date and time display formats
- Keepalive intervals
- Default destinations for incoming calls
- Global prefixes
- Call transfers from Cisco IP phones to other phone numbers
- Trunk access codes
- Message button phone numbers
- Class of restriction (COR) on the dial peers associated with directory numbers
- Call forwarding during a busy signal or no answer
- Translation rules for numbers dialed on Cisco IP phones
- Interdigit timeout value for all Cisco IP phones attached to the router
- Music on hold
- Dial-peer hunting
- Additional language options for IP phone display

SUMMARY STEPS

1. **call-manager-fallback**
2. **alias** *tag number-pattern* **to** *alternate-number* **preference** *preference-value*
3. **date-format** {**mm-dd-yy** | **dd-mm-yy**}
4. **time-format** {**12** | **24**}
5. **default-destination** *telephone number*
6. **keepalive** *seconds*
7. **dialplan-pattern** *tag prefix-pattern* **extension-length** *length* [**no-reg**]
8. **transfer-pattern** *transfer-pattern*
9. **access-code** {{**fxo** | **e&m**} *dial-string* | {**bri** | **pri**} *dial-string*} [**direct-inward-dial**]
10. **voicemail** *phone-number*
11. **cor** {**incoming** | **outgoing**} *cor-list-name* {*cor-list-number starting-number* **-** *ending-number* | **default**}
12. **call-forward busy** *directory-number*
13. **call-forward noan** *directory-number* **timeout** *seconds*
14. **translate** {**called** | **calling**} *translation-rule-tag*
15. **timeouts interdigit** *seconds*
16. **moh** *filename*
17. **user-locale** *country-code*

DETAILED STEPS

	Command	Purpose
Step 1	**call-manager-fallback** **Example:** `Router(config)#`	Enables SRS Telephony feature support and enters CallManager fallback mode.
Step 2	**alias** *tag number-pattern* **to** *alternate-number* **preference** *preference-value* **Example:** `Router(config-cm-fallback)#` **alias 1 60.. to 5001 preference 2**	(Optional) Allows routing of unmatched call destination to specific extension numbers with an associated dial-peer preference parameter.
Step 3	**date-format** {**mm-dd-yy** | **dd-mm-yy**} **Example:** `Router(config-cm-fallback)#` **date-format dd-mm-yy**	(Optional) Sets the date display format on all Cisco IP phones registered with the router. The default is mm-dd-yy.
Step 4	**time-format** {**12** | **24**} **Example:** `Router(config-cm-fallback)#` **time-format 24**	(Optional) Sets the time display format on all Cisco IP phones registered with the router. The default is 12 hours.

	Command	Purpose
Step 5	**keepalive** *seconds* **Example:** `Router(config-cm-fallback)# keepalive 60`	(Optional) Configures the time interval between Cisco IP phones keepalive messages sent to the router when SRS Telephony is enabled. The range is from 10 to 65535 seconds. The default timeout is 30 seconds.
Step 6	**default-destination** *telephone number* **Example:** `Router(config-cm-fallback)# default-destination 40802`	(Optional) Assigns the default destination number for incoming telephone calls.
Step 7	**dialplan-pattern** *tag prefix-pattern* **extension-length** *length* [**no-reg**] **Example:** `Router(config-cm-fallback)# dialplan-pattern 1 40855550.. extension-length 4 no-reg`	(Optional) Creates a global prefix that can be used to expand the abbreviated extension numbers into fully qualified E.164 numbers. The **extension-length** keyword enables the system to convert a full E.164 telephone number back to an extension number for the purposes of caller-ID display, received, and missed call lists. The **no-reg** keyword provides dialing flexibility and prevents the E.164 numbers in the dial peer from registering to the gatekeeper. You have the option not to register some specific numbers to the gatekeeper so that those numbers can be used for other telephony services.
Step 8	**transfer-pattern** *transfer-pattern* **Example:** `Router(config-cm-fallback)# transfer-pattern 52540..`	(Optional) Allows telephone call transfer from Cisco IP phones to other phone numbers both within the local IP network and outside of the local IP network.
Step 9	**access-code** {{**fxo** | **e&m**} *dial-string* | {**bri** | **pri**} *dial-string*} [**direct-inward-dial**] **Example:** `Router(config-cm-fallback)# access-code e&m 8`	(Optional) Configures trunk access codes for each type of line—Basic Rate Interface (BRI), E&M, Foreign Exchange Office (FXO), and Primary Rate Interface (PRI)—so that the Cisco IP phones can access the trunk lines while in Cisco CallManager fallback mode. The *dial-string* argument is used to set up temporary dial peers for each specified line type. The **direct-inward-dial** keyword enables you to set Direct Inward Dialing (DID) access for PRI and BRI trunk lines. **Note** The **access-code** command creates temporary dial peers in Cisco CallManager fallback mode. In many cases, you may already have the local PSTN ports configured with appropriate access codes provided by dial peers (for example, dial 9 to select an FXO PSTN line), in which case this command is not needed.
Step 10	**voicemail** *phone-number* **Example:** `Router(config-cm-fallback)# voicemail 914085551000`	(Optional) Configures the telephone number that is dialed when the message button on a Cisco IP phone is pressed.

	Command	Purpose
Step 11	**cor** {**incoming** | **outgoing**} *cor-list-name* {*cor-list-number starting-number* **-** *ending-number* | **default**} **Example:** `Router(config-cm-fallback)#` **`cor outgoing LockforPhoneC 1 5010 - 5020`**	(Optional) Configures a COR on the dial peers associated with directory numbers. COR is used to specify which incoming dial peer can use which outgoing dial peer to make a call. Each dial peer can be provisioned with an incoming and an outgoing COR list. The *cor-list-number* argument ranges from 1 to 6. The *starting-number* **-** *ending-number* argument provides a range of directory numbers. The **default** keyword instructs the router to use an existing default COR list.
Step 12	**call-forward busy** *directory-number* **Example:** `Router(config-cm-fallback)#` **`call-forward busy 50..`**	(Optional) Configures call forwarding to another number when the Cisco IP phone is busy. **Note** The E.164 number you enter can contain one or more "." wildcard characters. The wildcard characters correspond to the right-justified digits in the directory number extension.
Step 13	**call-forward noan** *directory-number* **timeout** *seconds* **Example:** `Router(config-cm-fallback)#` **`call-forward noan 5005 timeout 10`**	(Optional) Configures call forwarding to another number when no answer is received from the Cisco IP phone. The **timeout** keyword sets the waiting time, in seconds, before the call is forwarded to another phone. The *seconds* range is from 3 to 6000 seconds. **Note** The E.164 number you enter can contain one or more "." wildcard characters. The wildcard characters correspond to the right-justified digits in the directory number extension.
Step 14	**translate** {**called** | **calling**} *translation-rule-tag* **Example:** `Router(config-cm-fallback)` **`translate called 20`**	(Optional) Applies a translation rule to numbers dialed by Cisco IP phone users. The **called** keyword applies the translation rule to the outbound called party number. The **calling** keyword applies the translation rule to the inbound called party number. The *translation-rule-tag* argument is the reference number of the translation rule. Valid entries are from 1 to 2147483647. For further details, refer to the "Configuration Dial Plans, Dial Peers, and Digit Manipulation" chapter of the *Cisco IOS Voice, Video, and Fax Configuration Guide*, Release 12.2.

	Command	Purpose
Step 15	**timeouts interdigit** *seconds* **Example:** `Router(config-cm-fallback)# timeouts interdigit 5`	(Optional) Configures the interdigit timeout value for all Cisco IP phones attached to the router. The interdigit timeout specifies the number of seconds that the system waits after the caller has entered the initial digit or a subsequent digit of the dialed string. If the timeout ends before the destination is identified, a tone sounds and the call ends. **Note** This value setting is important when using variable-length dial peer destination patterns (dial plans). For more information on setting dial plans, see the "message URL Configuration Dial Plans, Dial Peers, and Digit Manipulation" chapter of the *Cisco IOS Voice, Video, and Fax Configuration Guide*, Release 12.2. The *seconds* argument is the interdigit timeout wait time in seconds. A valid entry is an integer from 2 to 120 seconds. The default is 10 seconds.
Step 16	**moh** *filename* **Example:** `Router(config-cm-fallback)# moh minuet.wav`	(Optional) Configures music on hold. This feature supports .au and .wav format music files. Music on hold works only for G.711 calls and on-net VoIP and PSTN calls. For all other calls, callers hear a tone. Internal calls between Cisco IP phones do not get music on hold, instead, callers hear a tone. **Note** The music on hold file can be in .wav or .au file format; however, the file format must contain 8-bit 8 kHz data, for example, Consultative Committee for International Telegraph and Telephone (CCITT) a-law or u-law data format.
Step 17	**user-locale** *country-code* **Example:** `Router(config-cm-fallback)# user-locale ES`	Selects a language for display on the Cisco IP Phone7940 and Cisco IP Phone 7960. • If you have Cisco CallManager V3.2, enter one of the following ISO-3166 codes: – Denmark—DE – France—FR – Germany—DE – Italy—IT – The Netherlands—NL – Norway—NO – Portugal—PT – Spain—ES – Sweden—SE – United States—US (default) • If you have software prior to Cisco CallManager V3.2, the only *country-code* option is US (United States).

Troubleshooting Tips

- To set detail debugging for the Cisco IP phones, use the **debug ephone detail** command.
- To set error debugging for the Cisco IP phones, use the **debug ephone error** command.
- To set call statistics debugging for the Cisco IP phones, use the **debug ephone statistics** command.
- To provide voice packet level debugging and print the contents of one voice packet in every 1024 voice packets, use the **debug ephone pak** command.
- To provide raw low-level protocol debugging display for all SCCP messages, use the **debug ephone raw** command.

For further debugging, you can use the debug commands in the *Cisco IOS Debug Command Reference*.

Configuring Cisco SRS Telephony for Unity Voice-Mail Integration

For dual tone multifrequency (DTMF) integrations, information on how to route incoming or forwarded calls is sent by the telephone system in the form of DTMF digits. The DTMF digits are in the form of a pattern and depend on the voice-mail system connected to the Cisco SRS Telephony router. These patterns are required for the DTMF integration with most voice-mail systems. The DTMF integration configuration on the Cisco SRS Telephony router works with any analog voice-mail system. Voice-mail systems are designed to respond to DTMF after the system has answered the incoming calls. The tasks described in the following sections are required:

- Configuring DTMF Patterns on the Router, page 2-9 (required)
- Configuring Integration Files on Legacy Voice-Mail Systems, page 2-11 (required)

Note FXO hairpin forwarded calls to voice mail must have disconnect supervision from the central office.

Configuring DTMF Patterns on the Router

The Cisco SRS Telephony router provides flexibility for the integration with any legacy voice-mail system. You can configure multiple tags and tokens for each pattern, depending on the voice-mail system and type of access. The *tag* in the configuration pattern must match the number defined in the voice-mail system's integration file to identify the type of call. The keywords—**CGN** (calling number), **CDN** (called number), and **FDN** (forwarding number)—define the type of call information sent to the voice-mail system.

SUMMARY STEPS

1. **vm-integration**
2. **pattern direct** *tag1* {**CGN** | **CDN** | **FDN**} [*tag2* {**CGN** | **CDN** | **FDN**}] [*tag3* {**CGN** | **CDN** | **FDN**}] [*last-tag*]
3. **pattern ext-to-ext busy** *tag1* {**CGN** | **CDN** | **FDN**} [*tag2* {**CGN** | **CDN** | **FDN**}] [*tag3* {**CGN** | **CDN** | **FDN**}] [*last-tag*]
4. **pattern ext-to-ext no-answer** *tag1* {**CGN** | **CDN** | **FDN**} [*tag2* {**CGN** | **CDN** | **FDN**}] [*tag3* {**CGN** | **CDN** | **FDN**}] [*last-tag*]
5. **pattern trunk-to-ext busy** *tag1* {**CGN** | **CDN** | **FDN**} [*tag2* {**CGN** | **CDN** | **FDN**}] [*tag3* {**CGN** | **CDN** | **FDN**}] [*last-tag*]

6. **pattern trunk-to-ext no-answer** *tag1* {**CGN** | **CDN** | **FDN**} [*tag2* {**CGN** | **CDN** | **FDN**}] [*tag3* {**CGN** | **CDN** | **FDN**}] [*last-tag*]

DETAILED STEPS

	Command	Purpose
Step 1	**`vm-integration`** **Example:** `Router(config) vm-integration`	Enters voice-mail integration mode and enables voice-mail integration with DTMF and analog voice-mail system.
Step 2	**`pattern direct`** *`tag1`* `{`**`CGN`** `|` **`CDN`** `|` **`FDN`**`} [tag2 {`**`CGN`** `|` **`CDN`** `|` **`FDN`**`}] [`*`tag3`* `{`**`CGN`** `|` **`CDN`** `|` **`FDN`**`}] [`*`last-tag`*`]` **Example:** `Router(config-vm-int) pattern direct 2 CGN *`	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system when the user presses the messages button on the phone. • The *tag* attribute is an alphanumeric string fewer than four DTMF digits in length. The alphanumeric string consists of a combination of four letters (A, B, C, and D), two symbols (* and #), and ten digits (0 to 9). The tag numbers match the numbers defined in the voice-mail system's integration file, immediately preceding either the number of the calling party, the number of the called party, or a forwarding number. The Cisco SRS Telephony router supports a maximum of four tags. • The keywords—**CGN**, **CDN**, and **FDN**—configure the type of call information sent to the voice-mail system, such as calling number (CGN), called number (CDN), or forwarding number (FDN).
Step 3	**`pattern ext-to-ext busy`** *`tag1`* `{`**`CGN`** `|` **`CDN`** `|` **`FDN`**`} [tag2 {`**`CGN`** `|` **`CDN`** `|` **`FDN`**`}] [`*`tag3`* `{`**`CGN`** `|` **`CDN`** `|` **`FDN`**`}] [`*`last-tag`*`]` **Example:** `Router(config-vm-integration) pattern ext-to-ext busy 7 FDN * CGN *`	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an internal extension attempts to connect to a busy extension and the call is forwarded to voice mail.
Step 4	**`pattern ext-to-ext no-answer`** *`tag1`* `{`**`CGN`** `|` **`CDN`** `|` **`FDN`**`} [tag2 {`**`CGN`** `|` **`CDN`** `|` **`FDN`**`}] [`*`tag3`* `{`**`CGN`** `|` **`CDN`** `|` **`FDN`**`}] [`*`last-tag`*`]` **Example:** `Router(config-vm-integration) pattern ext-to-ext no-answer 5 FDN * CGN *`	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an internal extension fails to connect to an extension and the call is forwarded to voice mail.

	Command	Purpose
Step 5	**pattern trunk-to-ext busy** *tag1* {**CGN** | **CDN** | **FDN**} [tag2 {**CGN** | **CDN** | **FDN**}] [*tag3* {**CGN** | **CDN** | **FDN**}] [*last-tag*] **Example:** `Router(config-vm-integration) pattern trunk-to-ext busy 6 FDN * CGN *`	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system once an external trunk call reaches a busy extension and the call is forwarded to voice mail.
Step 6	**pattern trunk-to-ext no-answer** *tag1* {**CGN** | **CDN** | **FDN**} [tag2 {**CGN** | **CDN** | **FDN**}] [*tag3* {**CGN** | **CDN** | **FDN**}] [*last-tag*] **Example:** `Router(config-vm-integration) pattern trunk-to-ext no-answer 4 FDN * CGN *`	Configures the DTMF digit pattern forwarding necessary to activate the voice-mail system when an external trunk call reaches an unanswered extension and the call is forwarded to voice mail.

Although it is unlikely that you will use multiple instances of the **CGN**, **CDN**, or **FDN** keyword in a single command line, it is permissible to do so.

Configuring Integration Files on Legacy Voice-Mail Systems

To configure the integration files for a legacy voice-mail system, follow the instructions in the voice-mail system's analog voice mail integration configuration guide or recommended documents. You must design the DTMF integration patterns appropriately, so that the voice-mail system and the Cisco SRS Telephony router work with each other. For a configuration example, see the "Configuring Cisco SRS Telephony for Unity Voice-Mail Integration" section on page 2-9.

Monitoring and Maintaining SRS Telephony

To monitor and maintain the router with SRS Telephony feature, use the following commands in EXEC mode:

Command	Purpose
`Router#` **show run**	Displays the configuration.
`Router#` **show call-manager-fallback all**	Displays the detailed configuration of all the Cisco IP phones, voice ports, and dial peers of the SRS Telephony router.
`Router#` **show call-manager-fallback dial-peer**	Displays the output of the dial peers of the SRS Telephony router.
`Router#` **show call-manager-fallback ephone-dn**	Displays Cisco IP phone destination number.
`Router#` **show call-manager-fallback voice-port**	Displays output for the voice ports.
`Router#` **show ephone dn** *dn-tag*	Displays Cisco IP phone destination number.
`Router#` **show ephone 7910**	Displays Cisco 7910 phone status.
`Router#` **show ephone 7940**	Displays Cisco 7940 phone status.
`Router#` **show ephone 7960**	Displays Cisco 7960 phone status.
`Router#` **show ephone offhook**	Displays Cisco IP phone status for all phones that are off hook.

Command	Purpose
Router# **show ephone registered**	Displays Cisco IP phone status for all phones that are currently registered.
Router# **show ephone remote**	Displays Cisco IP phone status for all nonlocal phones (phones that have no Address Resolution Protocol [ARP] entry).
Router# **show ephone ringing**	Displays Cisco IP phone status for all phones that are ringing.
Router# **show ephone summary**	Displays a summary of all Cisco IP phones.
Router# **show ephone telephone-number** *phone-number*	Displays Cisco IP phone status for a specific phone number.
Router# **show ephone unregistered**	Displays Cisco IP phone status for all unregistered phones.
Router# **show ephone-dn** *tag*	Displays Cisco IP phone destination number.
Router# **show ephone-dn summary**	Displays a summary of all Cisco IP phone destination numbers.
Router# **show ephone-dn loopback**	Displays Cisco IP phone destination numbers in loopback mode.
Router# **show voice port summary**	Displays a summary of all voice ports.
Router# **show dial-peer voice summary**	Displays a summary of all voice dial peers.

Configuration Examples

This section provides the following configuration examples:

- Basic and Optional Configuration Example, page 2-12
- Voice-Mail Integration Configuration Example, page 2-15

Basic and Optional Configuration Example

This section provides the following configuration example for the basic and optional SRS Telephony configurations described in this chapter:

```
!
version 12.2
no parser cache
no service single-slot-reload-enable
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
!
!
logging rate-limit console 10 except errors
!
!
!
memory-size iomem 30
ip subnet-zero
!
!
!
!
```

```
ip dhcp pool 2600
   network 10.2.0.0 255.255.0.0
   option 150 ip 10.0.0.1
   default-router 10.0.0.1
!
no ip dhcp-client network-discovery
lcp max-session-starts 0
!
!
!
translation-rule 1
 Rule 0 85... 919785
!
translation-rule 2
 Rule 0 408734.... 4
!
!
!
interface FastEthernet0/0
 ip address 10.0.0.2 255.255.0.0
 duplex auto
 speed auto
!
interface FastEthernet0/1
 ip address 10.0.0.1 255.255.0.0
 duplex auto
 speed auto
!
router eigrp 100
 network 10.0.0.0
auto-summary
 no eigrp log-neighbor-changes
!
ip classless
ip route 0.0.0.0 0.0.0.0 10.0.0.1
ip http server


snmp-server packetsize 4096
snmp-server manager
call rsvp-sync
!
voice-port 1/1/0
!
voice-port 1/1/1
!
mgcp modem passthrough voip mode ca
no mgcp timer receive-rtcp
!
mgcp profile default
!
dial-peer cor custom
 name call911
 name call1800
 name call1900
!
!
dial-peer cor list allowall
 member call911
 member call1800
 member call1900
!
!
!
```

```
dial-peer cor list allow1800
 member call1800
!
!
dial-peer cor list alloww1800and1900
 member call1800
 member call1900
!
dial-peer cor list allow1900
 member call1900

!
dial-peer voice 1 voip
 destination-pattern 919715....
 translate-outgoing called 2
 session target ipv4:10.0.0.5
!
dial-peer voice 2 voip
 shutdown
 destination-pattern 6....T
 session target ipv4:10.0.0.6
 codec g711ulaw
!
dial-peer voice 3 voip
 destination-pattern 65087.....
 session target ipv4:10.0.0.7
 codec g711ulaw
!
dial-peer voice 90 voip
 corlist outgoing allow1900
 destination-pattern 9000
 session target ipv4:10.0.0.8
!
dial-peer voice 45 pots
 destination-pattern 9
 port 1/1/0

!
call-manager-fallback
 ip source-address 10.0.0.1 port 2000
 max-ephones 10
 max-dn 10
 dialplan-pattern 1 408735.... extension-length 4 no-reg
 dialplan-pattern 2 919785.... extension-length 4 no-reg
 voicemail 4001
 no huntstop
 alias 2 3... to 5555
 translate called 1
 call-forward busy 5001
 call-forward noan 5001 timeout 8
 cor incoming allowall default
 cor incoming allowall 1 4000 - 4999
 cor incoming allowall 2 4000 - 5000
 moh minuet.au
 time-format 12
 date-format mm-dd-yy
 transfer-pattern 1...
 transfer-pattern 2...
 keepalive 30
 interdigit timeout 5
!
```

```
line con 0
line aux 0
line vty 0 4
 login
line vty 5 15
 login
!
!
end
```

Voice-Mail Integration Configuration Example

The examples in this section show how to configure analog voice-mail integration. They include configuration examples for local and central voice-mail systems.

Local Voice-Mail System Example

The "Dial-Peer Configuration for Integration for Voice-mail System" section of the example shows a legacy dial-peer configuration for a local voice-mail system. The "Cisco SRS Telephony Voice-mail Integration Pattern Configuration" part is a compatible Cisco SRS Telephony configuration.

```
! Dial-Peer Configuration for Integration for Voice-mail System
!
dial-peer voice 101 pots
 destination-pattern 14011
 port 3/0/0
!
dial-peer voice 102 pots
 preference 1
 destination-pattern 14011
 port 3/0/1
!
dial-peer voice 103 pots
 preference 2
 destination-pattern 14011
 port 3/1/0
!
dial-peer voice 104 pots
 destination-pattern A14012
 port 3/1/1
!
! Cisco SRS Telephony configuration
!
call-manager-fallback
 max-ephones 24
 max-dn 144
 ip source-address 1.4.214.104 port 2000
 voicemail 14011
!
! Cisco SRS Telephony Voice-mail Integration Pattern Configuration
!
vm-integration
 pattern direct 2 CGN *
 pattern ext-to-ext no-answer 5 FDN * CGN *
 pattern ext-to-ext busy 7 FDN * CGN *
 pattern trunk-to-ext no-answer 4 FDN * CGN *
 pattern trunk-to-ext busy 6 FDN * CGN *
```

Central Location Voice-Mail System Example

The "Dial-Peer Configuration for Integration with Voice-Mail System" section of the example shows a legacy dial-peer configuration for a central voice-mail system. The "Cisco SRS Telephony Voice-mail Integration Pattern Configuration" section is a compatible Cisco SRS Telephony configuration.

Note MWI integration is not supported for PSTN access to voice-mail systems at a central locations.

```
! Dial-Peer Configuration for Integration with Voice-Mail System
! located in central location
!
dial-peer voice 101 pots
 destination-pattern 14011
 port 3/0/0
!
! Cisco SRS Telephony configuration
!
call-manager-fallback
 max-ephones 24
 max-dn 144
 ip source-address 1.4.214.104 port 2000
 voicemail 14011
!
! Cisco SRS Telephony Voice-mail Integration Pattern Configuration
!
vm-integration
 pattern direct 2 CGN *
 pattern ext-to-ext no-answer 5 FDN * CGN *
 pattern ext-to-ext busy 7 FDN * CGN *
 pattern trunk-to-ext no-answer 4 FDN * CGN *
 pattern trunk-to-ext busy 6 FDN * CGN *
```