Configuration — SIP Media Gateway Avaya Secure Router 2330/4134

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Chapter 1: Introduction

Purpose

This document describes the operation and configuration of the SIP Media Gateway features on the Secure Router 2330/4134 (SR2330/4134).

For information on SIP Survivability features, see *Avaya Secure Router 2330/4134 Configuration — SIP Survivability* (NN47263-510).

Related resources

Documentation

See the *Avaya Secure Router 2330/4134 Documentation Roadmap*, NN47263-103, for a list of the documentation for this product.

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Chapter 2: New in this release

The following section details what is new in *Avaya Secure Router 2330/4134 Configuration — SIP Media Gateway* (NN47263-508).

Features

See the following sections for information about supported features:

Configuring call restriction options

By default the gateway accepts SIP requests from any IP address. You can configure the Secure Router to accept incoming calls from a configured SIP Server only, or from session targets. For more information, see Configuring call restriction options on page 76.

Support for a Secondary SIP server

The Secure Router and the Advanced Gateway (SR/AG) support dual Session Managers at the headquarters. Then, if the Primary SIP server fails, the AG/SR fails over to the Secondary SIP server so there's no interruption in service. For more information, see Secondary SIP server on page 20.

Other changes

The changes in this section are not feature related.

Licensing

All licensing information was moved from this document and consolidated into a **Licensing** chapter in *Avaya Secure Router 2330/4134 — Commissioning* (NN47263-302).

Chapter 3: Media Gateway fundamentals

The Avaya Secure Router 2330/4134 (SR2330/4134) supports an optional voice subsystem that allows the router to operate as a Session Initiation Protocol (SIP) to public switched telephone network (PSTN) Media Gateway. With the SIP-PSTN Media Gateway features enabled, the SR2330/4134 can interconnect public switched telephone network (PSTN) and Session Initiation Protocol (SIP) based Voice over IP networks. The SR2330/4134 performs the conversion of PSTN signaling to SIP signaling and TDM voice to Real time Transport Protocol (RTP) packets and vice versa. To implement the voice subsystem, on the SR4134 you must install an internal Packetized Voice Module (PVM), while on the SR2330 you must install a Packetized Voice Internal Module (PVIM). In addition, you must install voice connection modules on the SR2330/4134. To provide access to the PSTN, the SR2330/4134 supports the following interfaces and signaling :

- Foreign Exchange Office (FXO)
- Centralized Automatic Message Accounting (CAMA)
- Direct Inward Dialing (DID)
- T1 Channel Associated Signaling (CAS)
- T1 Integrated Services Digital Network (ISDN) Primary Rate Interface (PRI)
- E1 R2
- E1 ISDN PRI
- ISDN Basic Rate Interface (BRI)

The SR2330/4134 also supports Foreign Exchange Station (FXS) interfaces to provide direct connections for analog phone, fax, and modem lines. It also supports Q Signaling (QSIG) over T1 or E1 to connect to legacy PBXs and key systems.

Important:

Slot 2 of the SR4134 supports only one port for any WAN data small module. Therefore, if you install any 2-port small module in this slot and use it for data connections, one port only is functional (port 1). This limitation also applies to the 2-port T1/E1 and ISDN BRI small modules if they are configured for voice traffic: only port 2/1 is functional. However, this limitation does not apply to FXS or FXO voice modules.

This limitation does not apply to the SR2330.

To expand the number of small slots available on the router, the SR4134 supports a Voice Carrier medium module (not supported on the SR2330). You can install this module into a medium slot to provide four additional small slots for FXS and FXO ports only. The Voice Carrier medium module can support up to 16 FXS/FXO ports.

Supported SIP servers

To route calls in a SIP VoIP network, the SR2330/4134 must operate within a larger telephony solution. The network must contain a SIP Server to provide the necessary call routing capabilities.

The SR2330/4134 can interoperate with the following SIP servers:

- Avaya Aura (version 5.2.1 and later)
- Avaya SES/CM (version 5.2.1 and later)
- Avaya CS 1000 (version 4.5 and later)
- Avaya CS 2100
- Genband C20
- Avaya SCS (version 3.0 and later)
- Microsoft OCS R1 & R2
- Broadsoft (software version 14.0 SP1)
- Asterisk (version 1.6)

The Avaya Secure Router 2330 (SR2330) and Avaya Secure Router 4134 (SR4134) are certified for Microsoft OCS. The SR2330 is certified as a basic gateway and the SR4134 is certified as a hybrid gateway as it can host the Microsoft Mediation Server on one of its modules. The SR2330/4134 can also support mixed Call Server deployments. One supported deployment is with Avaya CS1000 and Microsoft OCS.

Important:

In this document, the term SIP server is used generically to refer to any of the supported SIP call routing systems that provide call routing for the SIP Media Gateway. SIP server does not refer to any particular component within these SIP call routing systems. Any information or configuration that is specific to a particular SIP server type is identified in the document, as applicable.

SR2330/4134 with SIP server

The following figure shows the SR2330/4134 interoperating in a network with a central SIP server. This is a typical scenario, in which the SIP server is located at the main office and the SR2330/4134 is located at the branch office. The branch SR2330/4134 provides connections to analog phones and external PSTN trunks and the SIP server provides call routing.

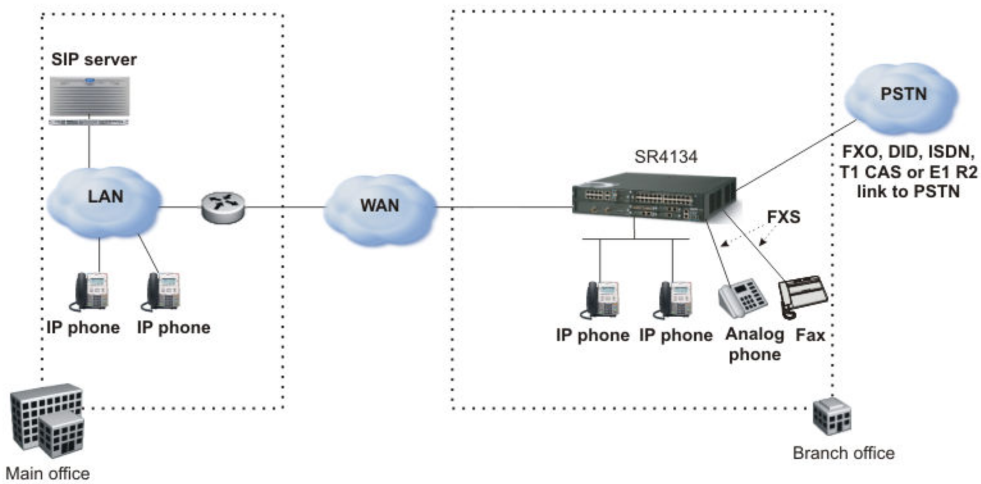


Figure 1: SR2330/4134 and SIP server interoperation

By default, all incoming analog calls received on the SR2330/4134 Media Gateway are forwarded to the central SIP server for routing. When a call is initiated from an analog phone or other POTS endpoint at the branch, the default SR2330/4134 operation is to build a SIP Uniform Resource Identifier (URI) and forward a SIP Invite to the central SIP server for call routing. The SIP server IP address is specified on the SR2330/4134 using the **`sip-ua sip-server`** command.

To route calls to and from the SR2330/4134, you must configure the SIP server with the appropriate routes to the SR2330/4134 endpoints.

When it receives a SIP Invite from the SR2330/4134, the SIP server uses configured call routing policies to signal the appropriate endpoint or media gateway. When the call is established, the media flows between the SR2330/4134 and the SIP endpoint across the IP network as shown in the following figure.

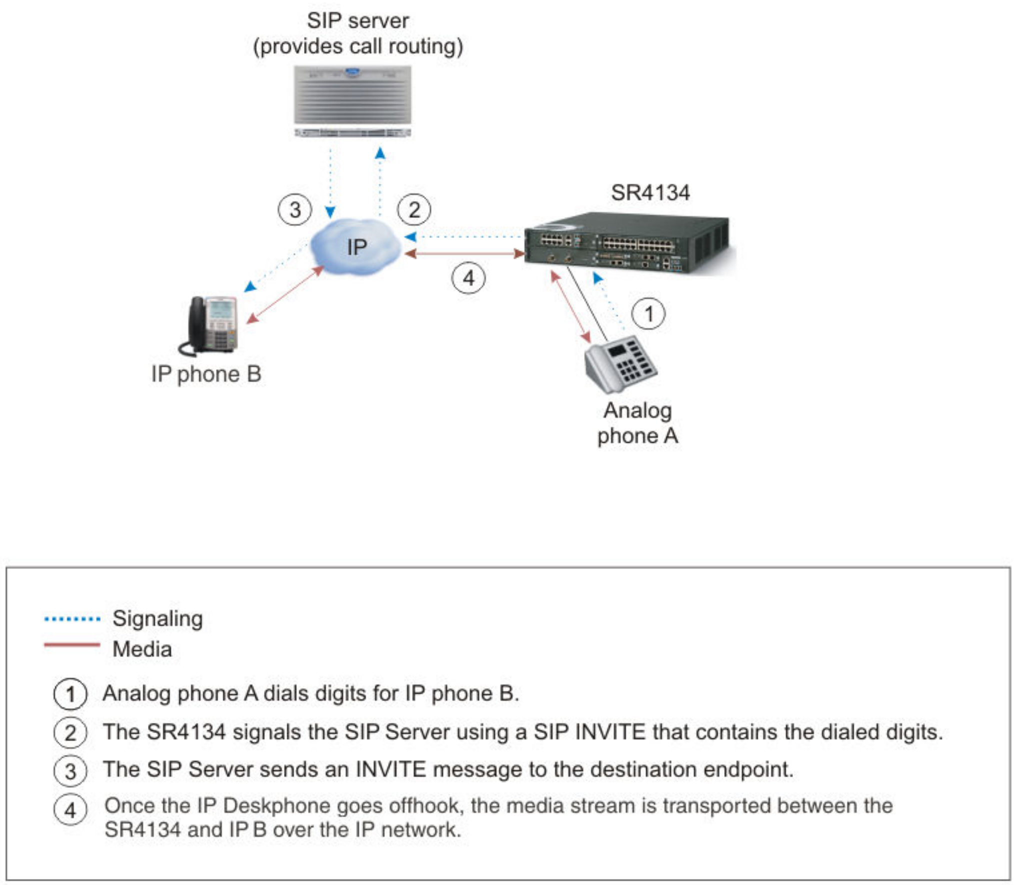


Figure 2: SIP server routing to remote endpoint

If the call destination is a POTS or PSTN interface on the originating SR2330/4134, the SIP server, using the configured call routing, sends a SIP Invite back to the SR2330/4134 that instructs the SR2330/4134 to complete the call locally. When the call is established, the media flows locally through the SR2330/4134, as shown in the following figure.

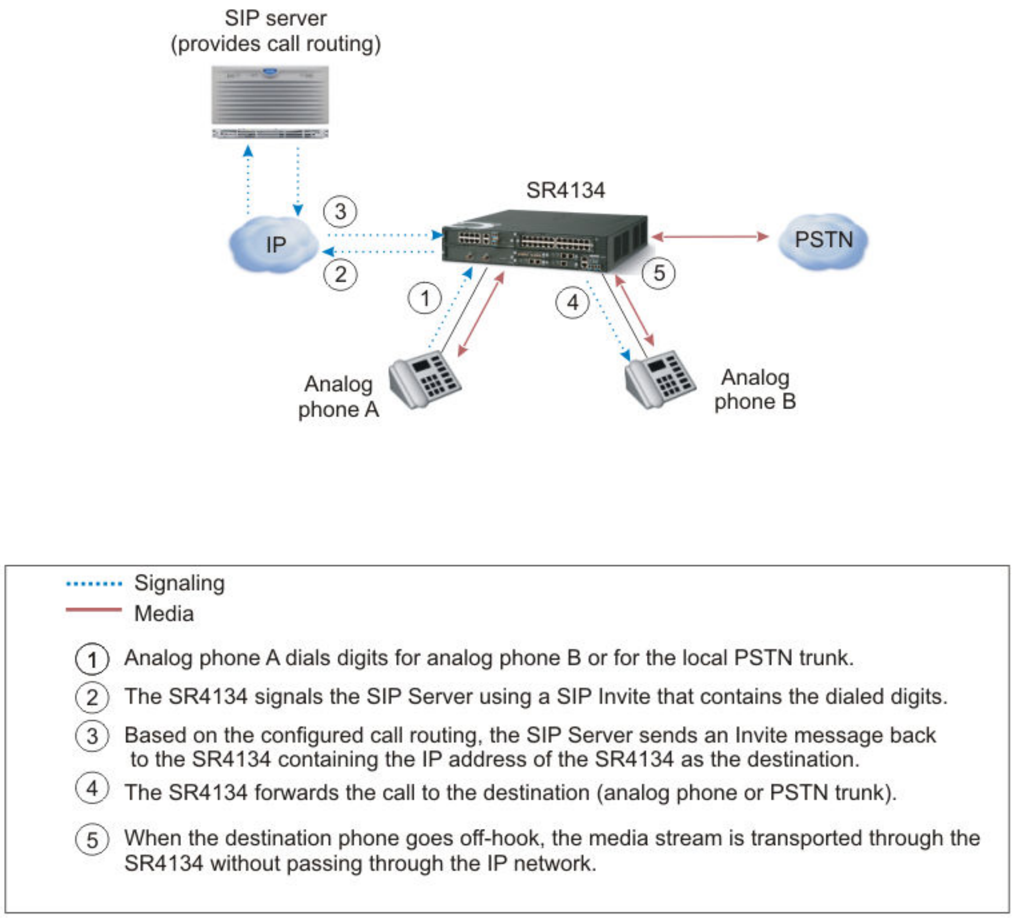


Figure 3: SIP server routing to local SR2330/4134 endpoint

The SIP server also handles the routing of all IP Phone calls, whether initiated at the main office or the branch. For example, when a branch IP phone that is connected to the SR2330/4134 initiates a call, the SR2330/4134 treats the traffic as standard data traffic and forwards it to the main office SIP server for call routing.

As shown in the following figure, you can also use the SR2330/4134 to connect calls between a legacy TDM PBX and the SIP network.

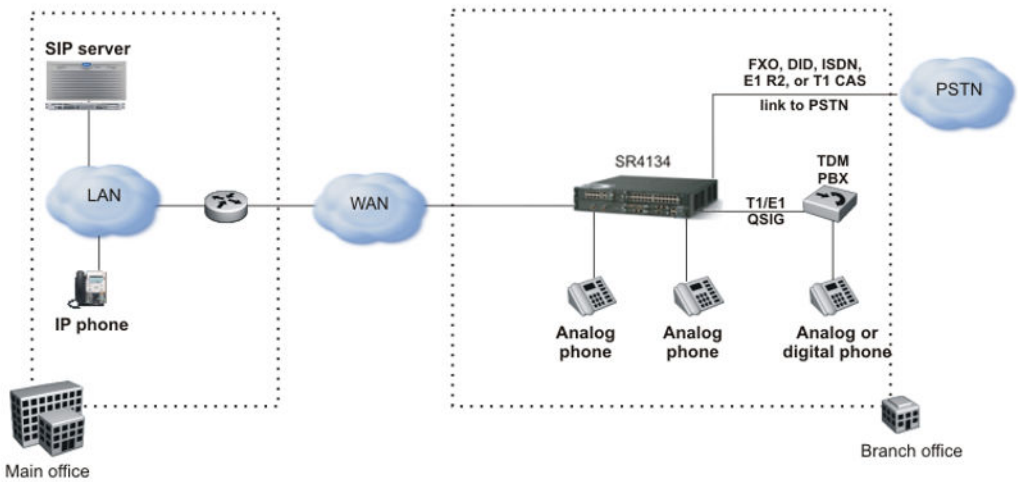


Figure 4: SR2330/4134 interoperation with SIP server legacy PBX

Secondary SIP server

The Secure Router and the Advanced Gateway (SR/AG) can serve as a branch Gateway with dual Session Managers (SM) at the headquarters. This feature works as follows:

1. If both the Primary SIP server and the Secondary SIP server are UP, the FXS lines register with Primary SIP server.
2. If the Primary SIP server goes DOWN, the SR/AG fails over to the Secondary SIP server and the FXS lines registered earlier with Primary SIP server now register with the Secondary SIP server.
3. When the Primary SIP server comes back UP, the AG/SR falls back to the Primary SIP server and the FXS lines register back with the Primary SIP server.
4. If the Primary SIP server is UP and the Secondary SIP server is DOWN, the FXS lines remain registered with the Primary SIP server.
5. If the Primary SIP server and the Secondary SIP server go DOWN, none of the FXS lines will be registered.
6. If the Primary SIP server is DOWN and the Secondary SIP server comes UP, the FXS lines register with the Secondary SIP server.
7. If the Primary SIP server comes UP and the Secondary SIP server is UP, the FXS lines register with the Primary SIP server.

Note:

If SSM is configured as a Tertiary SIP server and if both the Primary and Secondary SIP servers go DOWN, the SR/AG fails over to the Tertiary SIP server (SSM).

If you configure SSM as a Tertiary SIP server, Avaya recommends configuring the default-gateway in SSM with the gateway's bind IP, port and transport. This enables SSM to forward calls for users that are not registered with SSM to the gateway.

For configuration information and an example, see Specifying a Secondary SIP server on page 59.

Dial peers

Unlike PSTN voice calls, which use dedicated circuits from end to end, VoIP calls consist of a number of separate call legs. Within an end-to-end connection, each call leg defines a logical connection between two points (for example, between two Secure Routers, or between a Secure Router and an analog phone).

As shown in the following figure, an end-to-end VoIP call between Secure Routers is comprised of four call legs: two at the source Secure Router, and two at the destination Secure Router.

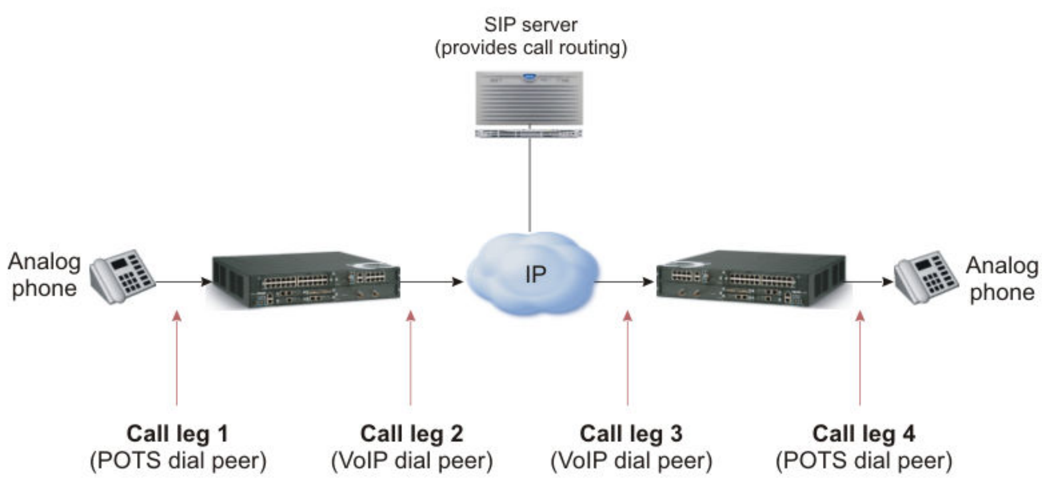


Figure 5: Dial peers in analog-to-VoIP call

To define the properties associated with each call leg and to identify call origin and destination, the SR2330/4134 uses logical dial peers.

There are two basic kinds of dial peers:

- Plain old telephone service (POTS) dial peer

  Describes the characteristics of a traditional telephony network connection. POTS dial peers associate a string of dialed digits with a voice port that connects the SR2330/4134 to the PSTN, to a local PBX, or to an analog telephone.

- Voice-network dial peer (for VoIP)

VoIP dial peers describe the characteristics of an IP network connection and associate a string of dialed digits with a remote network device, such as a remote Media Gateway connected to a destination telephone.

With the SR2330/4134, the remote network device specified in the VoIP dial peer is typically the SIP server, which provides the necessary call routing.

By configuring the appropriate dial peer properties, you can apply attributes to a call leg such as fax rate (for VoIP dial peers) and calling-line ID (for POTS dial peers).

Dial peers are relevant only to the gateway on which they are configured.

Dial peer example

The following figure shows dial peers configured on two Secure Routers.

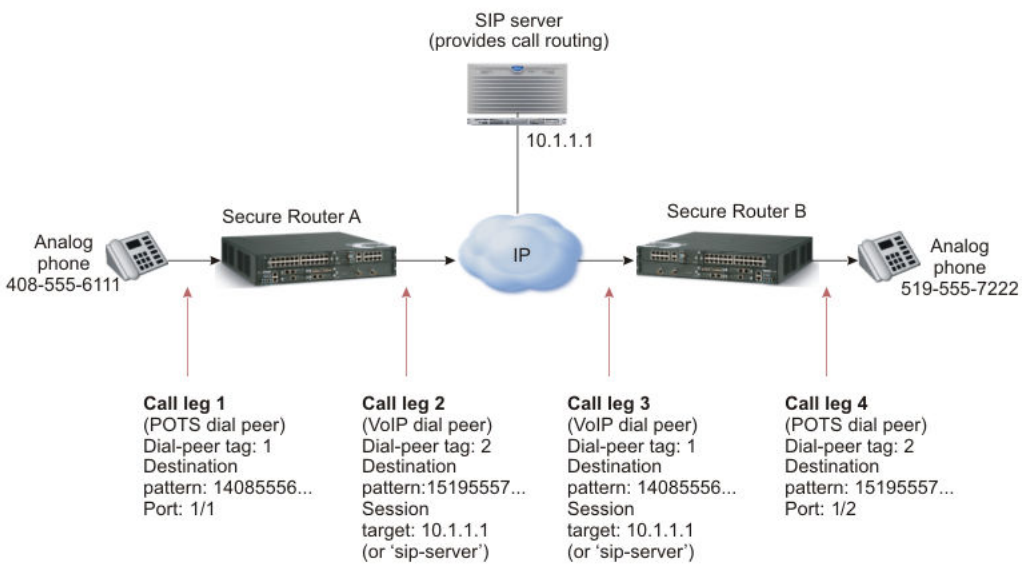


Figure 6: Dial peer example

To establish calls between the two analog phones, you must configure a POTS dial peer on each SR2330/4134. The POTS dial peer specifies the parameters and dialed digits for the analog phone connection.

To define VoIP call parameters, you can also configure a VoIP dial peer on each SR2330/4134. The VoIP dial peer specifies the parameters and dialed digits for the VoIP connection, as well as the address of the session target used to route outgoing calls (the SIP server).

You are not strictly required to configure VoIP dial peers to route outgoing VoIP calls. If you do not configure a VoIP dial peer, by default, the SR2330/4134 forwards outgoing calls to the SIP server (specified by the **`sip-ua sip-server`** command) for routing.

Whether or not VoIP dial peers are configured, the SIP server must be configured appropriately to route the destination patterns to the appropriate gateway. In this case, the SIP server must have a route with 4085556... pointing to Advanced Gateway A as the destination and a route with 5195557... pointing to Advanced Gateway B as the destination.

SR2330/4134 call routing

The default behavior of the SR2330/4134 is to route all incoming TDM calls to the central SIP server for routing. Dialed numbers received from a directly connected endpoint are forwarded to the SIP server for routing, even if the destination is a locally connected POTS or PSTN interface.

If you have configured VoIP dial peers, the SR2330/4134 first attempts to match the dialed digits to one of the VoIP dial peers before forwarding the call to the central SIP server. If it finds a match, the SR2330/4134 forwards the call to the target address specified in the dial peer. If it finds no match, the SR2330/4134 forwards the call to the central SIP server. The following sections provide more details about this call routing behavior.

Incoming POTS call routing

When the SR2330/4134 receives an incoming POTS call, the gateway compares the full E.164 called number against the destination patterns configured for each local VoIP dial peer (but not POTS dial peers). The SR2330/4134 then performs one of the following actions:

- Matching VoIP dial peer

  If the SR2330/4134 can match the called number to a VoIP dial peer, it routes the call to the SIP server address or target network address specified by the VoIP dial peer. It applies the appropriate POTS dial peer connection attributes for the incoming call leg and the matched VoIP dial peer attributes for the outbound leg.

- No matching VoIP dial peer

  If the SR2330/4134 finds no VoIP dial-peer match, it forwards the call to the central SIP server (specified using the **`sip-ua sip-server`** command) for further routing.

For numbers other than the matched VoIP dial peers, the SR2330/4134 routes the call to central SIP server assuming that it has the necessary information for routing. Whether they are intra-site calls, inter-site calls reachable through extension dialing, calls to subscribers within the service provider domain, calls to subscribers outside the service provider domain, or calls to the PSTN network, the central SIP server has the routing rules to route the calls.

SIP URI

When routing to the SIP Network, the SR2330/4134 builds a SIP URI to embed in the SIP messages. To create a SIP URI, the SR2330/4134 suffixes the managed service domain name to the entered digits. The managed service domain name is the domain name of the service provider that provides hosted PBX services to the enterprise. For example, if the user dials destination number 1-408-555-1000 and the service provider domain is example.com, the request URI in the outgoing Invite is sip:14085551000@example.com.

The SR2330/4134 supports SIP URI only with numeric usernames and routes calls based on the username.

SIP server routing

To route calls to and from the SR2330/4134, you must configure the central SIP server with the appropriate routes to the SR2330/4134 endpoints.

When the SIP server receives a SIP Invite from the SR2330/4134, using the configured call routing policies, it signals the appropriate endpoint or media gateway to complete the call. When the call is established, the media flows between the SR2330/4134 and the SIP endpoint across the IP network as shown in the following figure.

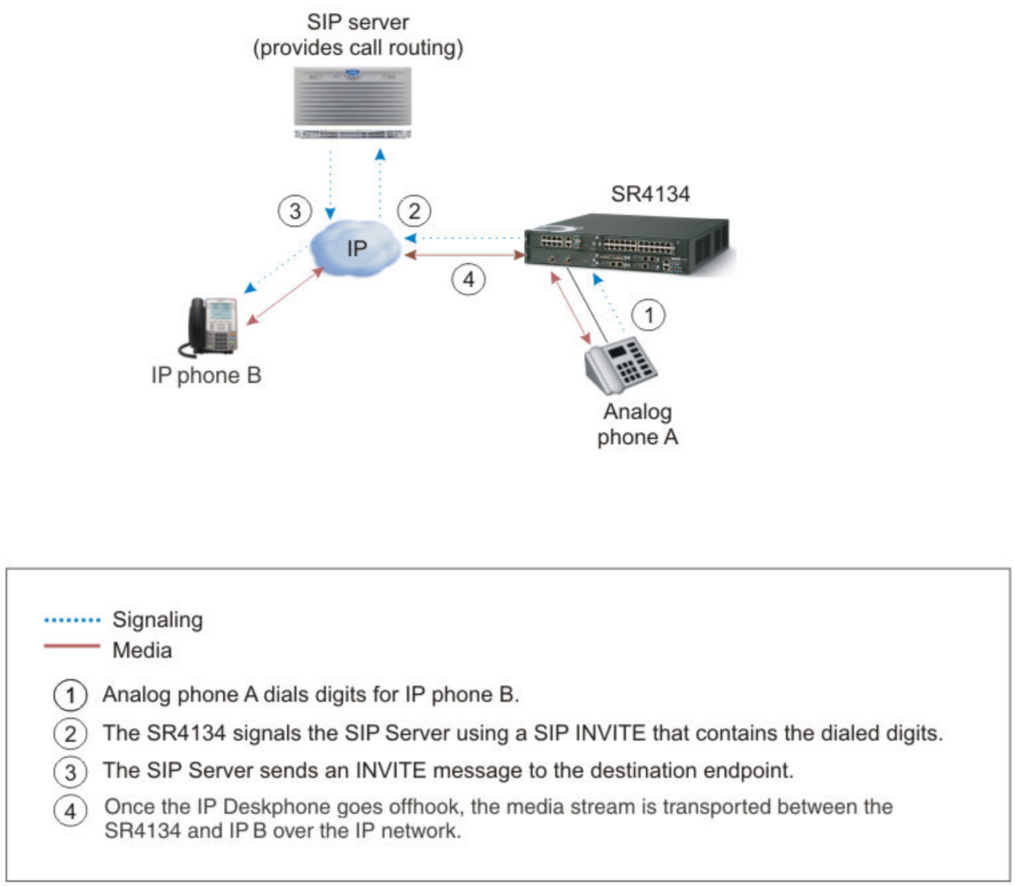


Figure 7: SIP server routing to remote endpoint

If the call destination is a POTS or PSTN interface on the originating SR2330/4134, the SIP server, using the configured call routing policies, sends a SIP Invite back to the SR2330/4134 that instructs the SR2330/4134 to complete the call locally. When the call is established, the media flows locally through the SR2330/4134, as shown in the following figure.

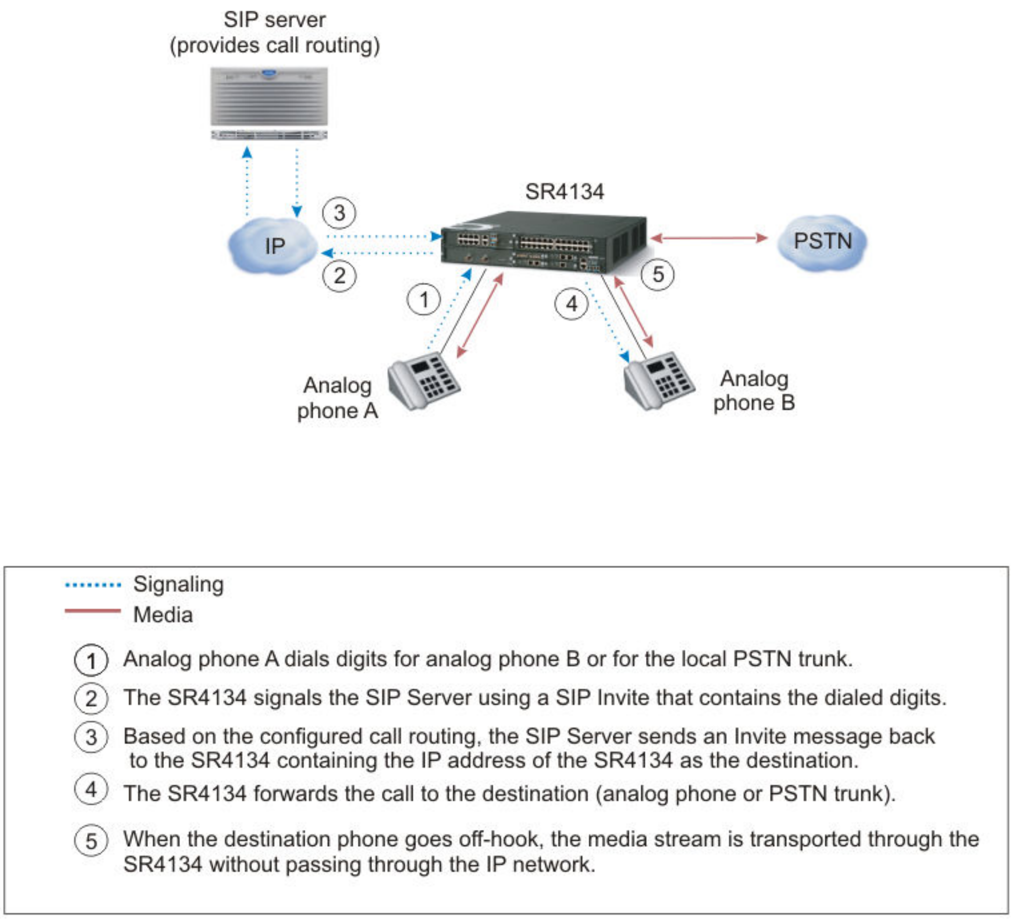


Figure 8: SIP server routing to local SR2330/4134 endpoint

Outgoing POTS call routing

When the SR2330/4134 receives a SIP Invite from the SIP server, the SR2330/4134 compares the user digits from the SIP URI against the destination patterns of the local POTS dial peers. If the SR2330/4134 finds a POTS dial peer to match to the incoming digits, the SR2330/4134 connects the call to the voice port or PSTN interface specified by the outgoing POTS dial peer. It also applies the appropriate POTS (for local POTS or PSTN connections) or VoIP (for external VoIP connections) dial peer attributes for the incoming call leg and the matched POTS dial peer attributes for the outbound leg.

If the SR2330/4134 finds no dial peer match, it rejects the call.

Pass-through prefix digit for SIP server failure

If the central SIP server is not configured or is otherwise unreachable (for example, if the SIP server fails or the link to the SIP network is lost), the SR2330/4134 rejects all standard calls. As a result, the SR2330/4134 can no longer route any incoming or outgoing calls.

As a workaround to SIP server failure, you can configure a pass-through prefix digit that can force a call to be routed to one of the local directly-connected PSTN interfaces without going through the central SIP server. In this case, the SR2330/4134 treats the call as internal, and no SIP Invite is sent to the SIP server.

The following figure shows the flow for an example call that uses the pass-through prefix digit.

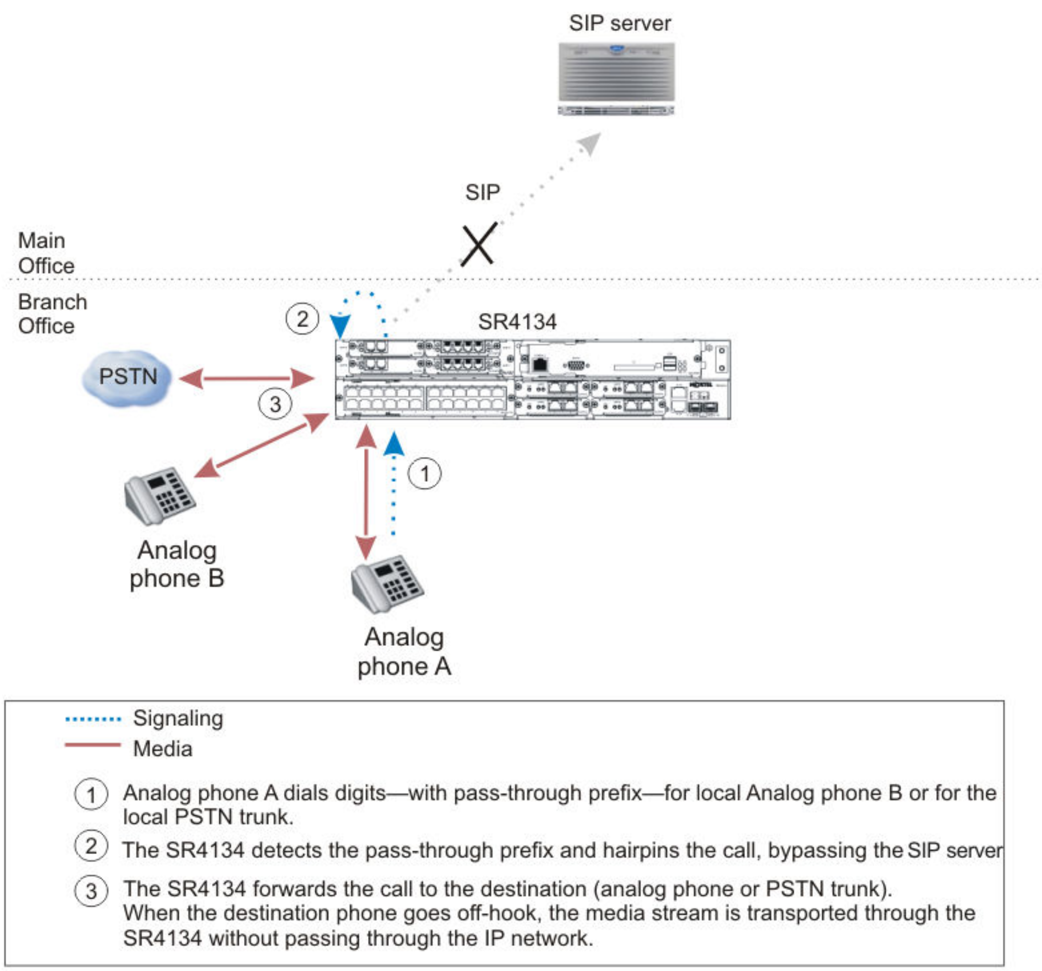


Figure 9: Pass-through prefix routing

Emergency call handling

The SR2330/4134 can route 911 calls. The SR2330/4134 treats emergency calls like pass-through prefix calls in that it first attempts to match the dialed digits of the emergency call to a local POTS dial peer. If the SR2330/4134 can match the dialed digits to a POTS dial peer, then the call is routed directly to the associated trunk without signaling the central SIP server. However, if no matching POTS dial peer is found, then the SR2330/4134 forwards the call to the SIP server for routing.

To enable emergency call routing, you must configure a POTS dial peer for the required digits and specify the same digits as the emergency number (using the **`emergency-number`** command under the **`SR/configure/voice/service/voip#`** command tree) . Otherwise, the call is treated as a standard call and is forwarded to the SIP server for routing.

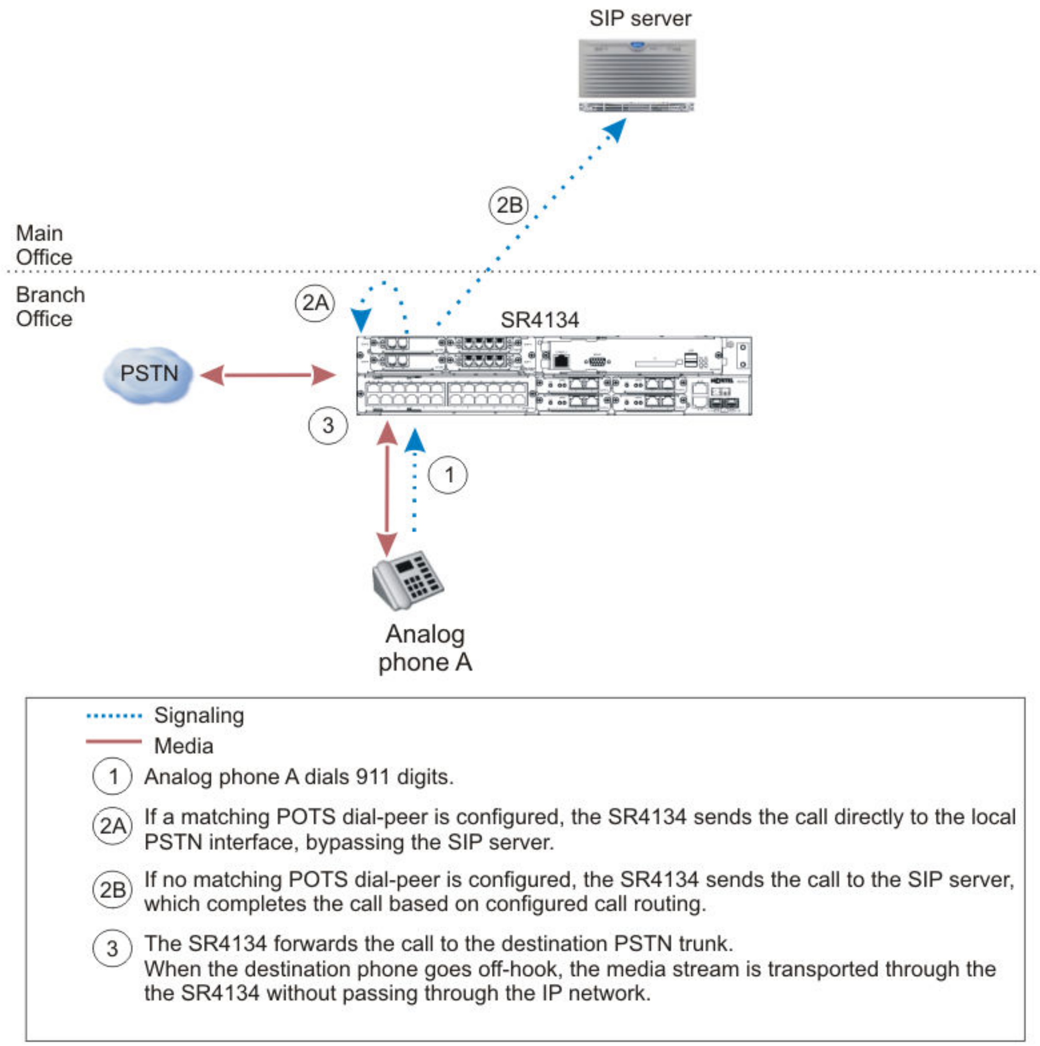


Figure 10: Emergency call routing

E911 emergency call routing with CAMA trunks

The SR2330/4134 supports direct connections to the PSAP in the E911 network using CAMA trunks. This meets recently enacted legislation that requires enterprises to connect directly to the E911 network (this requirement is expected to expand to all US states).

CAMA restrictions

The SR2330/4134 CAMA trunks have the following restrictions:

- Direct trunking is not supported
- Automatic location information (ALI) / Data Management Systems (DMS) Reverse ALI lookup features of E911 are not supported
- Alternate routing for busy traffic and night service for power failure are not supported

Call routing logic

The following figure shows the main SR2330/4134 call routing logic for calls coming from TDM ports.

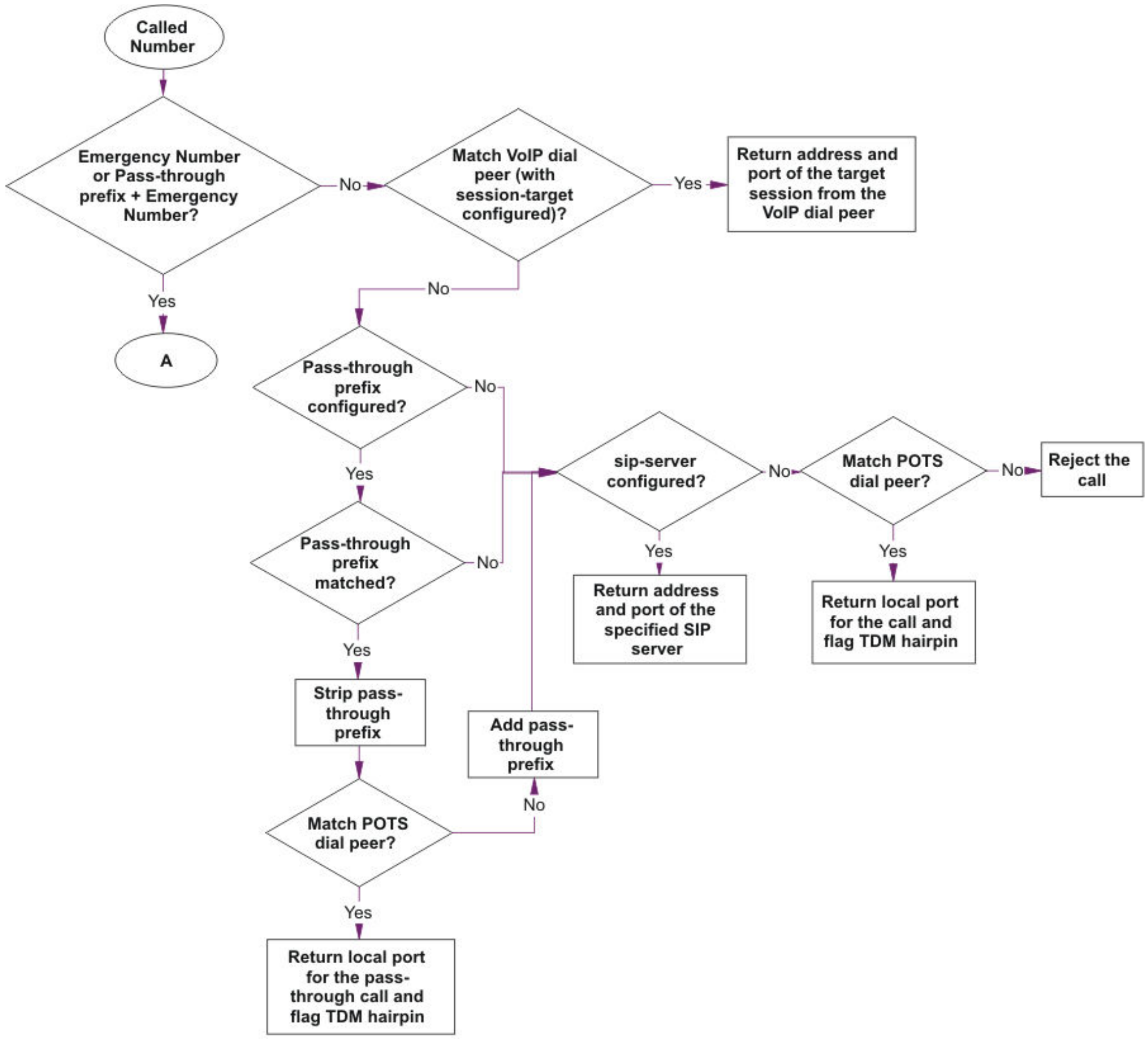


Figure 11: SR2330/4134 internal call routing logic for incoming calls from TDM ports

Note that if the SIP server is down (rather than not configured on the SR2330/4134), a user must dial a pass-through prefix (or emergency number) to route directly to a local POTS connection.

The following figure shows the SR2330/4134 call routing logic for emergency calls coming from POTS endpoints.

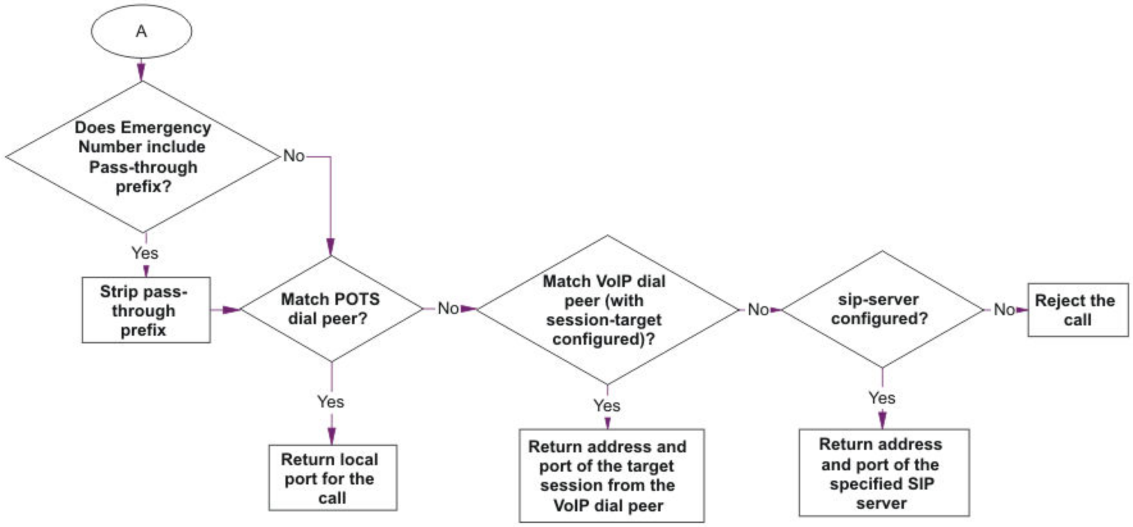


Figure 12: SR2330/4134 internal call routing logic A: Emergency calls from TDM ports

The following figure shows the SR2330/4134 call routing logic for calls coming from the SIP network.

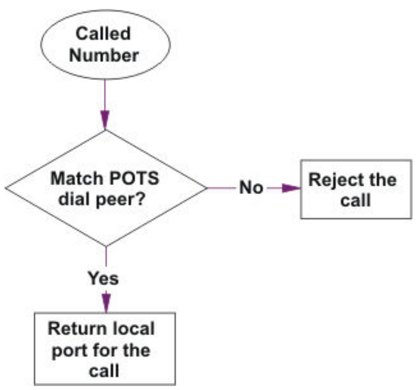


Figure 13: SR2330/4134 internal call routing logic for calls coming from the SIP network

Call routing for SIP endpoints connected to SR2330/4134 Ethernet ports

The SR2330/4134 treats any SIP endpoints that are directly connected to SR2330/4134 Ethernet ports as standard data traffic. The central SIP server controls the call routing for these endpoints.

Note:

If the SIP Survivability module (SSM) is enabled on the SR2330/4134, then the SSM can function as the backup proxy for these SIP endpoints.

Dial peer destination patterns

Each dial peer is associated with a destination pattern that specifies either a prefix or a full E.164 telephone number. The pattern you configure is used to match dialed digits to a dial peer.

When an SR2330/4134 receives voice data, it compares the called number (the full E.164 telephone number) in the packet header with the number configured as the destination pattern for the dial peer. The SR2330/4134 then strips out the left-justified numbers that correspond to the destination pattern. If you have configured a prefix, the prefix is appended to the front of the remaining numbers, which creates a dial string that the SR2330/4134 dials.

Valid entries for the destination pattern are the digits 0 through 9, and the characters shown in the following table.

Table 1: Valid characters for dial peer destination patterns

Character	Description
Period (.)	Matches any entered digit (this character is used as a wildcard).
Percent (%)	Indicates that the preceding digit occurred zero or more times; similar to the wildcard usage.
Plus (+)	Indicates that the preceding digit occurred one or more times
Brackets ([ ])	Indicate a range. A range is a sequence of characters enclosed in the brackets; only numeric characters from 0 to 9 are allowed in the range.
T	Indicates that the destination pattern value is a variable-length dial string. In some areas of the world (for example, some European countries), valid telephone numbers can vary in length. Use the optional control character T to indicate that a particular destination pattern value is a variable-length dial string. In this case, the system does not match the dialed numbers until the interdigit timeout value expires.

The plus (+) character is sometimes as a leading character for international calls. However, the SR2330/4134 uses the plus (+) character as a wildcard in the destination pattern. To route calls properly, the SR2330/4134 ignores the leading plus (+) character in the dialed digits when performing dial-peer matching.

Supported dial plan schemes

You can input the following dial patterns when using numeric destination addresses.

- Extension format: Calls within the enterprise are established using n-digit extension numbers. The calls can be within the same site or inter-site. However, extension dialing from the PSTN is not supported. The central SIP infrastructure performs the necessary expansion from the n-digit extension to a full number and routes the call to the final destination.
- External number format: Calls to numbers outside the enterprise can be any of the formats below.
    - Full 10-digit E.164 numbers
    - 1+ 10-digit E.164 number
    - 011+ 14-digit international number
- Locally routed external number format: A user can force a call to be routed locally to the PSTN using a directly connected PSTN interface by entering a special pass-through digit prefix. In this case, the SR2330/4134 treats the call as internal, and no SIP Invite is sent to the SIP server. For example, if the prefix value is 6, an FXS port can dial another FXS port at extension 1234 by dialing 61234. This is useful when the SR2330/4134 cannot reach the SIP server.

  You can configure this prefix using the **`pass-through-call-prefix`** command under the **`SR/configure/voice/service/voip#`** command tree.

Number translation

Number translation allows you to modify the telephone numbers that enter or leave the SIP Media Gateway. For example, you can add an area code to a number that must be routed to the PSTN, or remove an area code for a number that is routed to an internal company site. You can also use number translation to add or strip the plus (+) character that is sometimes used for international calls.

You can apply number translation to the incoming or outgoing call leg. As well, you can choose to modify the calling party number, the called party number, or both.

Translation rules

To implement number translation, you must create translation rules. Each rule contains a match pattern to match against the called or calling number, and a replacement pattern that is applied to the number when a match is found.

Translation rules are collected together into translation-rule groups. Each translation-rule group can contain up to 10 individual translation rules. The translation-rule group lists the translation rules in order of priority.

When searching for a match in the translation-rule group, the Media Gateway compares the telephone number against each listed rule in the order of priority. When a match is found, the specified replacement pattern is applied to the number.

The SR2330/4134 supports a maximum of 15 translation-rule groups.

Translation profile

When you have configured the translation-rule groups, you must create a translation profile and associate it with the desired translation-rule groups. Each translation profile can be associated with up to two translation-rule groups:

- One translation-rule group for calling number translation
- One translation-rule group for called number translation

There are a maximum of 128 translation profiles allowed in the system.

Applying the translation profile

When you have configured a translation profile, you must apply it to a dial peer, specifying whether the profile is applied in the incoming or outgoing direction.

Translation profiles with incoming dial peers

To apply a translation profile to an incoming dial peer, the SR2330/4134 must first match the incoming number to the appropriate dial peer. The SR2330/4134 identifies incoming VoIP and POTS dial peers using two separate processes.

Identification of incoming VoIP dial peer

The SR2330/4134 identifies the incoming VoIP dial peer by matching the calling number with the appropriate VoIP dial peer destination pattern.

Identification of incoming POTS dial peer

The SR2330/4134 identifies the incoming POTS dial peer using the following order:

1. First, the SR2330/4134 attempts to match the calling party number by comparing it against the destination patterns of the POTS dial peers.
2. If no match is found in step 1, the SR2330/4134 searches for a dial peer that is directly associated with the voice port that is receiving the incoming call.
3. If no dial peer is found in step 2, the SR2330/4134 searches for a dial peer associated with a trunk group that includes the incoming voice port.

In steps 2 or 3, if multiple matches are found, the dial peer with the highest tag is selected.

Applying the incoming profile

After the incoming number is matched to the appropriate dial peer, if the dial peer is associated with incoming translation profiles, the called and calling numbers are translated as required. After the numbers are translated, the SR2330/4134 matches the new called number to the appropriate outgoing dial peer to complete the call.

Translation profile with outgoing dial peers

Translation profiles are applied to outgoing dial peers as follows: when the SR2330/4134 processes the outgoing call leg, it first matches the outgoing (called) number to the appropriate dial peer. Then, if the dial peer is associated with outgoing translation profiles, the called and calling numbers are translated as required. After the numbers are translated, the SR2330/4134 uses the new called number to initiate the call on the outgoing dial peer.

Additional notes

When configuring number translation, be aware of the following:

1. There is no need to change any of the existing dial-peer destination patterns.
2. Match and replace patterns only allow the period (.) as a wildcard character and it cannot be followed by any additional digits. For example, 613……. is allowed but 613…4567 is not.
3. Wildcards are used for pattern matching, whereas the replace pattern only replaces the non-wildcard entries of the matched-pattern. For example, while dial string 6135555001 matches with the following rule:

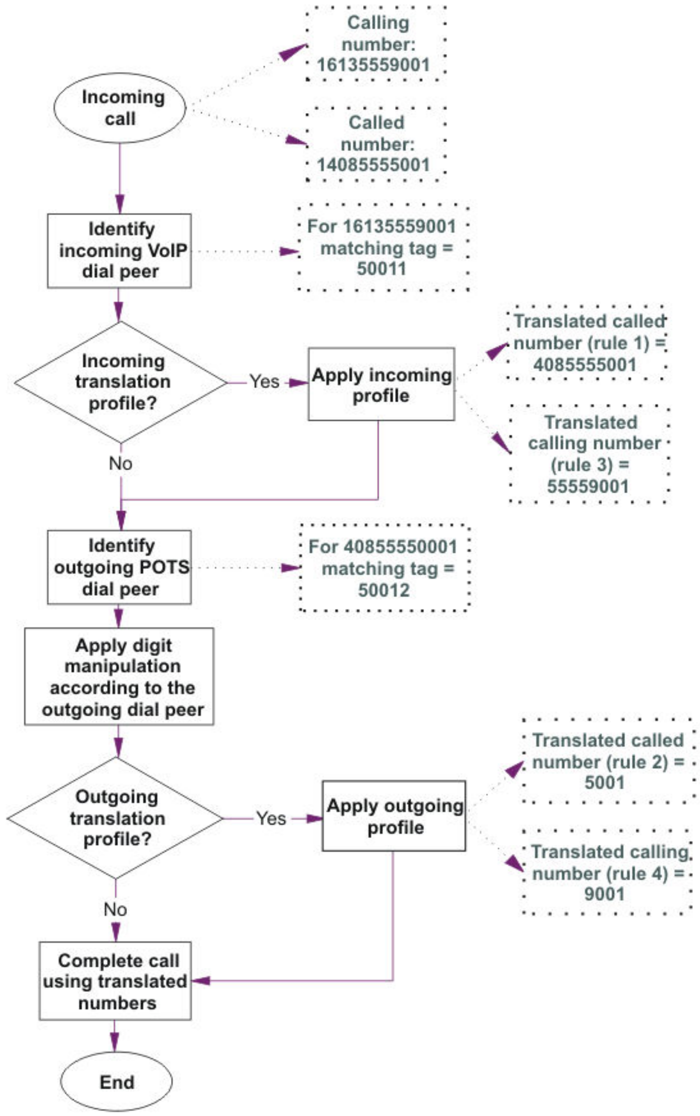
   ```
   rule 1 /613......./ //
   ```

   the replace pattern in this rule replaces 613 with nothing, so that the translated string becomes 5555001.

   On the other hand, dial-string 6135555 is shorter than the match pattern above and therefore does not match with the rule. As a result there is no translation.
4. You can implement + stripping by associating an appropriate translation profile with an incoming dial peer.
5. You can add a + prefix by associating an appropriate translation profile with an outgoing dial peer.
6. The SR2330/4134 does not allow + as a prefix in the dial-peer destination pattern.
7. As the SR2330/4134 does not allow + as a prefix in the dial-peer destination pattern, the SR2330/4134 ignores the + prefix when matching dial peers.

Number translation examples

The following figures show the basic process that the SR2330/4134 follows to apply number translation to sample SIP-to-TDM and TDM-to-SIP calls.



Example configuration:

```
voice translation-rule 5001
rule 1 /1408/ /408/
rule 2 /408555..../ //
rule 3 /16135559001/ /5559001/
rule 4 /5559001/ /9001/
exit

voice translation-profile SIP2TDM
translate called 5001
translate calling 5001
exit

dial-peer voice voip 50011
destination-pattern .T
translation-profile incoming SIP2TDM
session target ipv4:192.168.9.202
no shut
exit

dial-peer voice pots 50012
destination-pattern 4085555001
port 1/1
translation-profile outgoing SIP2TDM
no digit-strip
no shut
exit
```

Figure 14: SIP-to-TDM number translation example

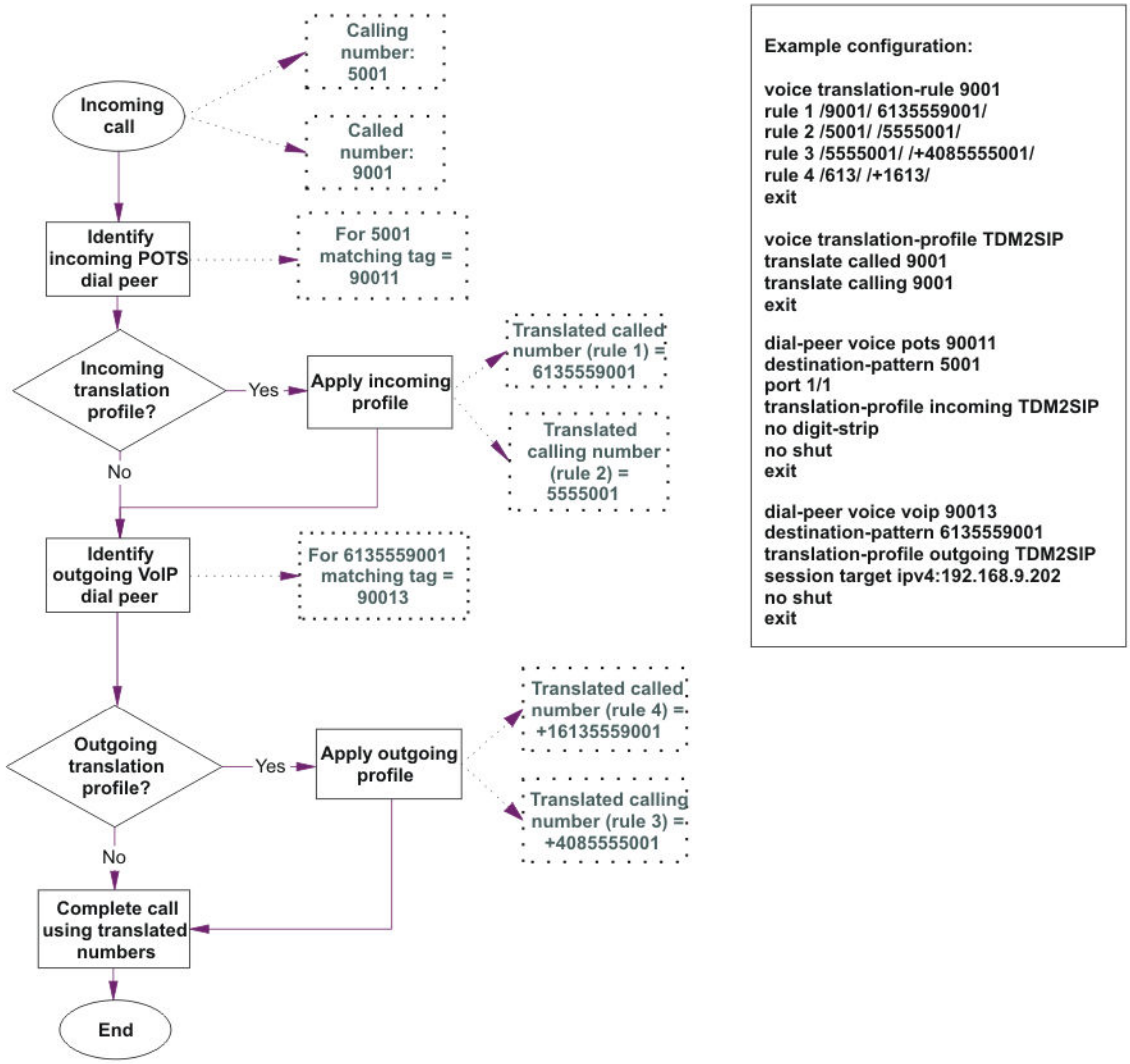


Figure 15: TDM-to-SIP number translation configuration example

Dial peer trunk groups

Trunk groups allow you to direct incoming calls to a destination trunk that belongs to a configured trunk group. If one or more trunks in the group is busy, the incoming call is routed to one of the free lines. You can assign any T1 CAS, ISDN PRI, ISDN BRI, or FXO port to a trunk group. You cannot add FXS ports to the group.

The SR2330/4134 chooses the destination trunk from within the group in a round-robin fashion. You cannot specify a preferred destination trunk within the group.

To direct calls to a trunk group, you must create the trunk group, add interfaces to it, and then associate a dial peer with the trunk group. You cannot associate a dial peer to a voice port and a trunk group simultaneously.

A voice port can belong to only one trunk group at a time. You cannot add a voice port to multiple trunk groups.

Caller ID on FXS and FXO ports

The SR2330/4134 supports caller ID on FXS and FXO ports. You can configure FXS ports to send caller ID and FXO ports to receive it.

With FXS ports, you can specify the station number to send as caller ID.

With FXO ports, you can also specify a station number if caller ID information is expected to be received from the CO. This supports situations in which a call is placed from the CO to the FXO interface and continues to a far-end FXS port through an on-net call. If the FXO port receives no caller ID information from the CO, the FXO port forwards the configured station number to the far-end FXS port.

Caller ID with dial peers

The SR2330/4134 also supports calling-line ID (CLID) information on dial peers. The CLID configured for POTS dial peers is applicable only for POTS-to-SIP calls. The configuration specifies the calling party number that is entered in the From field of the SIP Invite message. While you can configure the CLID on POTS dial peers for any port type, it is typically applied to incoming trunk calls.

If you configure the CLID network-number for a dial peer, the configured number appears in the outgoing SIP Invite message, replacing the incoming calling party number and name. You can also choose a restrict the CLID, in which case calls from the dial peer are routed to the SIP network as anonymous (the calling party number is added under the P-Asserted-Identity Header in accordance with RFC 3325). You can also configure a substitute display name. In this case, if the SR2330/4134 receives an incoming trunk call with no CLID, it can use the configured substitute display name.

DSP properties

On all voice ports, you can configure DSP properties to specify digital signal properties that are applied to the ports. The configurable properties include:

- comfort noise
- compand type
- echo cancellation
- input gain (for FXS or FXO ports only)
- output attenuation (for FXS or FXO ports only)

Resource management

The SR2330/4134 supports 1 call per second (CPS) of voice traffic.

If licensed for sufficient DSP channels, the SR4134 supports a maximum of 128 IP to TDM channels with G.711 20ms codec and SR2330 supports a maximum of 64 IP to TDM channels with G.711 20ms codec. As described in the following table, the DSP capacity is lower if the SR2330/4134 is running more complex codecs.

Table 2: DSP channel capacity for different codecs

Codec	SR2330 Maximum channel density	SR4134 Maximum channel density
G.711 (10ms)	64	96
G.711 (20ms)	64	128
G.726 (10ms)	24	64
G.726 (20ms)	32	64
G.723.1	24	64
G.729AB	32	64
T38	32	32

Call Admission Control

Call Admission Control (CAC) support is limited to a simple calculation based on the number of calls. You must statically configure the bandwidth of the uplink connecting to the SIP server using the SR2330/4134 QoS CLI based on the number of voice calls to support. The bandwidth of a voice call is dependent on the codec negotiated. The worst case bandwidth requirement is while using the G711 codec.

If the uplink goes down then all the active calls on that link are dropped.

The SR2330/4134 limits the number of calls based on the configured bandwidth value.

Supported voice modules

With the SR2330/4134, the voice subsystem is an optional feature. To implement the voice subsystem, on the SR4134 you must install an internal Packetized Voice Module (PVM), while on the SR2330 you must install a Packetized Voice Internal Module (PVIM). The PVM or PVIM can arrive factory-installed with new orders, or you can install it in the field on previously purchased routers. For information about field installation, see *Avaya Secure Router 2330/4134 Installation — Hardware Components* (NN47263-301).

The PVM and PVIM modules provide the voice conversion from Time-Division Multiplexing (TDM) signals to IP Real-time Transport Protocol (RTP) packets and vice versa. They also provide a number of digital signal processing (DSP) functions including echo cancellation, voice activity detection (VAD), comfort noise generation (CNG), tone detection and generation, and dual-tone multifrequency (DTMF) digit collection. They can support fax over IP using T.38 fax relay or fax pass-through as well as modem over IP using modem pass-through. They also have an in-built timeslot interchanger for TDM to TDM switching.

In addition to the internal PVM and PVIM modules, the SR2330/4134 supports a number of voice-capable modules to provide external voice connections.

The following modules provide analog voice interface connections:

- 2-port and 4-port FXS/DID small module
- 2-port and 4-port FXO/CAMA small module

The following modules provide pulse-code modulation (PCM) encoded voice channels. With these modules, you can configure each individual port in a slot to operate in voice mode and you can enable or disable each port separately.

- 1-port T1/E1 small module (T1 configured as ISDN PRI or T1 CAS; E1 configured as ISDN PRI or E1 R2)
- 2-port T1/E1 small module (T1 configured as ISDN PRI or T1 CAS; E1 configured as ISDN PRI or E1 R2)
- 2-port ISDN BRI S/T small module (only supported in point-to-point mode; not supported in point-to-multipoint mode)

The following table describes the maximum supported voice ports on each SR2330/4134 chassis type.

Table 3: Maximum supported voice ports

Interface type	SR2330 maximum supported ports	SR4134 maximum supported ports
T1/E1 ports	2	4
BRI ports	6	7
FXS/DID ports	12	64 (See Note 1)
FXO/CAMA ports	12	64 (See Note 1)
Note 1: With the Voice Carrier Medium module, the SR4134 Media Gateway can accommodate a total of up to 64 FXS/FXO ports. However, only 32 total FXS/FXO ports are tested and qualified for this release.		

FXS/DID modules

The FXS module allows the SR2330/4134 to connect to a conventional analog POTS phone, fax, or modem. The SR2330/4134 emulates PBX or CO behavior for the connected device. The SR2330/4134 provides dial tone and ring, collects digits, and applies call progress tones to indicate the progress of the call to the user. The SR2330/4134 then converts the POTS signaling into SIP signaling and vice versa.

SR2330/4134 FXS ports support the following features:

- loop-start signaling
- ground-start signaling
- battery reversal (with optional answer supervision)
- caller ID
- Direct Inward Dialing (DID), which provides off-premise DID connections from a central office (calls from the PSTN only) and supports caller ID

SIP signaling uses en bloc address signaling while digits on POTS lines arrive one at a time. As a result, the SR2330/4134 uses a digit map to analyze the digits received on the POTS interface against valid dial plans. If it finds a matching dial plan, the SR2330/4134 forwards the dialed digits in the Invite when initiating the new call. If it cannot find a matching dial plan, the SR2330/4134 rejects the call.

The SR2330/4134 can apply call hold on POTS lines if the hold is initiated from the VoIP network (any PSTN or POTS line can be put on hold by an external SIP endpoint). The SR2330/4134 can initiate call hold/resume from FXS lines, but cannot initiate call hold requests received from POTS or PSTN lines.

The SR2330/4134 does not support voice mail. The SIP server, in tandem with a voice mail server, can provide voice mail for the FXS phones configured on the SR2330/4134.

The SR2330/4134 FXS module is interoperable with FCC Class A device compliant phones.

FXO/CAMA modules

The FXO module allows the SR2330/4134 to connect to the PSTN CO or to a PBX using an analog connection. The SR2330/4134 emulates POTS phone behavior for the CO; it detects ring and tones, dials digits, and collects caller ID.

The SR2330/4134 converts FXO signaling into SIP signaling and vice versa.

SR2330/4134 FXO ports support the following features:

- loop-start signaling
- ground-start signaling
- battery reversal (with optional answer supervision)
- supervisory disconnect (always enabled for FXO ports)
- caller ID
- Centralized Automatic Message Accounting (CAMA) for E911 service
- Private Line Automatic Ringdown (PLAR)

FXO modules are interoperable with FCC Class A device FXO interfaces.

ISDN BRI modules and ISDN PRI on T1/E1 small modules

The SR2330/4134 supports ISDN PRI and BRI signaling to provide user-side digital voice connections to a PSTN CO or to a PBX. The SR2330/4134 supports ISDN BRI modules and ISDN PRI signaling on T1/E1 small modules in T1 and E1 mode. Both BRI and PRI connections use data (D) channels for signaling and bearer (B) channels for transporting voice traffic. The SR2330/4134 converts ISDN signaling for voice into SIP signaling and vice versa.

The SR2330/4134 supports ISDN BRI connections using the ISDN BRI S/T small module only in point-to-point mode. Each BRI module consists of two 64 kilobits per second (Kb/s) B channels for voice and one 16 Kb/s D channel for control information.

ISDN PRI is supported on the 1-port and 2-port T1/E1 small modules in T1 and E1 modes. T1 ISDN PRI connections provide 23 B-channels for carrying voice traffic and 1 D-channel for signaling. E1 ISDN PRI connections provide 30 B-channels for carrying voice traffic and 1 D-channel for signaling.

The following sections describe some of the supported ISDN standards for voice.

ISDN overlap receiving for voice

The SR2330/4134 supports overlap receiving as an alternative signaling method to en bloc signaling for call establishment of ISDN voice calls. While most modern switches use en bloc signaling, many European countries continue to support overlap signaling. The SR2330/4134 supports overlap receiving on BRI ports configured for Euro ISDN and on E1 PRI ports configured for Euro ISDN or QSIG.

Important:

The SR2330/4134 does not support overlap sending. The SR2330/4134 always uses en bloc sending.

Important:

The SR2330/4134 supports overlap receiving on BRI ports configured for Euro ISDN and on E1 PRI ports configured for Euro ISDN or QSIG. You cannot enable overlap receiving on other ISDN switch types.

QSIG

The SR2330/4134 supports QSIG on E1 and T1 PRI interfaces. Q Signaling (QSIG) is a variant of ISDN Q.921 and ISDN Q.931 D-channel signaling for use in private integrated services networks. QSIG operates between nodal entities known as private integrated network exchanges (PINX), such as PBXs or key systems.

With QSIG signaling, the SR2330/4134 can emulate PSTN functionality. The SR2330/4134 can route incoming PINX voice calls across a WAN connection to a peer SR2330/4134, which can then forward the call to another PINX. The QSIG messages pass transparently across the WAN link between the QSIG SR2330/4134s and the call flow remains compliant with Q.931. This functionality provides toll bypass between the two PINXs.

The following figure shows a sample topology that uses QSIG connections to route PBX calls over a WAN link.

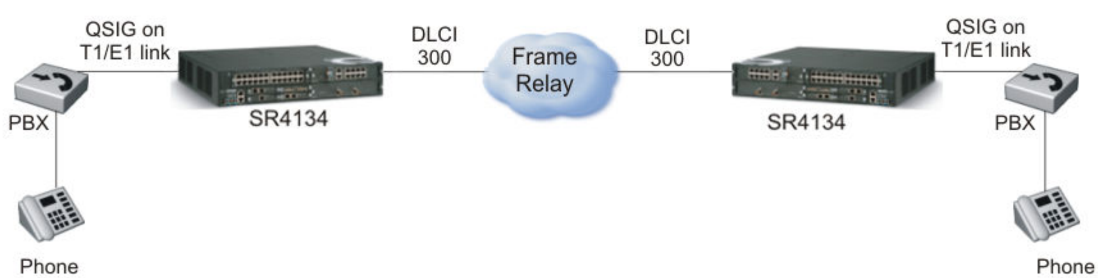


Figure 16: QSIG sample topology

The SR2330/4134 supports only basic QSIG services; it does not support supplementary services.

The QSIG protocol was originally specified by the European Computer Manufacturers Association (ECMA). It has since been adopted by European Telecommunications Standards Institute (ETSI) and the International Organization for Standardization (ISO). It is becoming the standard for PBX interoperability in Europe and North America.

T1 CAS

The T1/E1 small modules support CAS signaling to provide the SR2330/4134 with voice connections to the PSTN. The T1/E1 modules support CAS only in T1 mode. (R2 signaling is supported in E1 mode.) The SR2330/4134 converts the CAS signals into SIP signaling and vice versa. When configured for T1 CAS, each interface supports 24 timeslots for voice traffic.

With CAS signaling, the least significant bit of information in a T1 signal is robbed from the channels that carry voice and is used to transmit signaling information. This is sometimes called in-band signaling. CAS is a method of signaling within each traffic channel rather than having a dedicated signaling channel (like ISDN). The signaling for a particular traffic circuit is permanently associated with that circuit.

CAS signaling also processes the receipt of Dialed Number Identification Service (DNIS) and Automatic Number Identification (ANI) information, which is used to support authentication and other functions. The biggest disadvantage of CAS signaling is the use of user bandwidth to perform signaling functions.

The SR2330/4134 supports the following CAS signaling formats: E&M Wink Start, E&M Immediate Start, and E&M Delay Dial.

E1 R2

In E1 mode, the T1/E1 small modules support R2 signaling to provide the SR2330/4134 with voice connections to the PSTN. When configured for E1 R2, each interface supports 30 timeslots for voice traffic.

R2 signaling is a channel associated signaling (CAS) system developed in the 1960s that is still in use today in Europe, Latin America, Australia, and Asia. R2 signaling exists in several country versions or variants in an international version called Consultative Committee for International Telegraph and Telephone (CCITT-R2). The R2 signaling specifications are contained in International Telecommunication Union Telecommunication Standardization Sector (ITU-T) Recommendations Q.400 through Q.490. E1 R2 signaling is an international signaling standard that is common to channelized E1 networks. Although R2 signaling is defined in the ITU-T Q.400-Q.490 recommendations, there are many variations in how R2 is implemented. (Various countries have chosen to implement R2 differently.) The SR2330/4134 currently supports E1 R2 digital line signaling and compelled interregister signaling, as well as the Argentina, Brazil, China, Korea, Mexico and Singapore E1 R2 country variants.

E1 R2 backward digit configuration

Beginning with Release 10.2, you can use the character ‘R’ in the configuration of backward-digits to indicate a variable number of digits.

Incoming Call – variable DNIS

The Secure Router accepts a variable number of digits in the Dialed Number Identification Service (DNIS) number or dialed number. For a variable number of digits to be accepted, the backward-digits must be configured as 1R31, where **R** indicates a variable number of digits.

For example, if a peer device uses the following dial plan, the Secure Router backward digit must be configured as **1R31**.

- International Call—12 digits
- National Call—10 digits
- Local—7 digits
- Extension—4 digits

If the peer device is configured to send an **F** to indicate the end of DNIS, the Secure Router sends a **3** (Group A backward signal) after receiving the **F**, indicating that the number is complete.

If the peer device configured not to send an **F** to indicate the end of DNIS, the Secure Router sends a **3** (Group A backward signal) after an interdigit timeout (default 6 seconds), indicating that the number is complete.

In previous releases, the Secure Router accepted only a fixed number of digits in the DNIS number. For example if the backward digit was 11131 then the Secure Router could accept a call with only 4 digits in the DNIS number.

Incoming Call – request ANI

You can configure the Secure Router to request Automatic Number Identification (ANI), or caller identification. For this request, you must configure the Secure Router backward-digit as **1R61R31**. The second **R** in the backward-digit configuration indicates a variable number of digits in ANI.

For example:

- If the peer is configured to send 4, 7, or 10 digits for ANI, the Secure Router backward-digit can be configured as **1R61R31**.
- If the peer is configured to send a fixed 4 digits for ANI, the Secure Router backward-digit can be configured as **1R61111131**. In this case, after receiving 4 digits, the Secure Router sends a **3** to indicate that ANI is complete. If the ANI includes more than four digits the Secure Router ignores the extra digits.

The following table shows backward digit configurations for various combinations of DNIS and ANI numbers, and digits.

Call Direction	DNIS	ANI	Backward digit configuration
Incoming	4 (Fixed)	None	11131
Incoming	4 (Fixed)	4 (Fixed)	1113111131
Incoming	Variable	4 (Fixed)	1R6111131
Incoming	Variable	Variable	1R61R31

Call Direction	DNIS	ANI	Backward digit configuration
Incoming	4 (Fixed)	Variable	11161R31
Incoming	5 (Fixed)	None	111131
Incoming	5 (Fixed)	5 (Fixed)	111131111131
Incoming	Variable	5 (Fixed)	1R61111131
Incoming	Variable	Variable	1R61R31
Incoming	5 (Fixed)	Variable	111161R31

Inter-Digit Timeout Configuration

The default value of the EI R2 inter-digit timeout value is set to 6 seconds.

Voice Carrier medium module

The Voice Carrier medium module can house up to four small FXS or FXO modules. With this module, you can expand the number of available small slots on the SR4134. You can install FXS or FXO modules only in the Voice Carrier module.

The SR2330 does not support the Voice Carrier medium module.

With the Voice Carrier Medium module, the SR4134 Media Gateway can accommodate a total of up to 64 FXS/FXO ports. However, only 32 total FXS/FXO ports are tested and qualified for this release.

Mediation Server module

The Mediation Server module is a SR4134 medium module that hosts Microsoft Mediation Server software. There are two versions of this module. One uses a 32bit Intel CPU and another uses 64bit Intel CPU. The 32bit module runs Microsoft Windows Server 2003 and hosts OCS R1 version of Microsoft Mediation Server. The 64bit module runs Microsoft Windows Server 2008 and hosts OCS R2 version of Microsoft Mediation Server. The Mediation Server software performs the necessary signaling and media transcoding for calls between the OCS network and the Media Gateway.

The SR2330 does not support the Mediation Server module.

The Mediation Server translates SIP and PSTN calls to OC client connections and translates calls from OC clients to the Media Gateway for routing to the PSTN or SIP network.

The main functions of the Mediation Server are as follows:

- Translating SIP over TCP (on the gateway side) to SIP over Mutual TLS (on the Enterprise Voice side).
- Encrypting and decrypting SRTP on the OCS side.
- Translating media streams between the OCS and the Media Gateway.

The Mediation server module supports traffic up to the equivalent of one T1 or 24 simultaneous calls.

Voice feature summary

The following table lists the SR2330/4134 Media Gateway supported features.

Table 4: Supported voice features

Feature	Description
T1/E1 ISDN PRI Digital Trunks	• NI2 – North America • DMS 100 – Genband • 5ESS – AT&T • Euro ISDN • NTT – Japan • QSIG
ISDN BRI S/T Digital Trunks	• NI – North America • DMS 100 – Genband • 5ESS – AT&T • Euro ISDN • User side support only • BRI S/T is only supported in point-to-point mode
E1 R2	• ITU-T Q.400-Q.490 • Digital line signaling • Compelled interregister signaling • Variants: Argentina, Brazil, China, Korea, Mexico, and Singapore
FXS/DID	• Caller ID in FXS mode • Answer and Disconnect Supervision in DID mode

Feature	Description
	• Signaling Formats: - FXS- Loop Start and Ground Start - DID - Immediate, Delay Dial, Wink Start • Address Signaling Formats: - In-band DTMF • Countries: - Canada - USA - Brazil Homologation for other countries in progress
FXO/CAMA	• Answer and Disconnect Supervision with loop-start signaling • Analog CAMA trunk • Caller ID support • Signaling Formats - Loop Start and Ground Start - ANI signaling for CAMA E911 • Address Signaling Formats - In-band DTMF - MF for CAMA E911 • Countries - USA - Canada - Mexico Homologation for other countries in progress
T1 CAS	• Signaling formats - E&M Wink Start - E&M Immediate Start - E&M Delay Dial • Countries - USA - Canada

Feature	Description
DSP	• Codecs - G.711 A-law and u-law - G.726 - 16, 24, 32 Kb/s - G.723.1 - 5.3, 6.3 Kb/s - G.729AB • Features - G.168 Echo Canceller - max tail length of up to 128ms - Voice Activity Detection - Comfort Noise Generation - Out-of-band DTMF • SR2330 Max Channel Density - G.711 Channels (10ms) - 64 - G.711 Channels (20ms) - 64 - G.726 Channels (10ms) - 24 - G.726 Channels (20ms) - 32 - G.729A Channels - 32 - G.723.1 Channels - 24 - T38 Channels - 32 • SR4134 Max Channel Density - G.711 Channels (10ms) - 96 - G.711 Channels (20ms) - 128 - G.726 Channels (10/20ms) - 64 - G.729A Channels - 64 - G.723.1 Channels - 64 - T38 Channels - 32
Fax & Modem	• FAX and Modem Pass-Through • T.38 Fax Relay – 2.4 kbps, 4.8 kbps, 7.2 kbps, 9.6 kbps, 12.0 kbps, 14.4 kbps
Call Features	Call Hold/Resume
SIP	• RFC 3261 • Support for UDP and TCP transport
Management	CLI

Limitations

The SR2330/4134 Media Gateway has the following limitations:

- The SR2330/4134 Voice Carrier Medium module only supports FXS/FXO cards.
- With the Voice Carrier Medium module, the SR2330/4134 Media Gateway can accommodate a total of up to 64 FXS/FXO ports. However, only 32 total FXS/FXO ports are tested and qualified for release 10.3.
- The SR2330/4134 Media Gateway supports only the user side for ISDN BRI.
- The SR2330/4134 Media Gateway supports no local call features on voice interfaces.
- The SR2330/4134 Media Gateway does not support multiple identical media types in the SDP. For example, if the SR2330/4134 receives an offer SDP containing 2 audio m-lines, it processes only the first audio stream in the offer and rejects all other similar media.
- The SR2330/4134 Media Gateway does not support Secure Real-time Transport Protocol (SRTP).
- The SR2330/4134 Media Gateway does not support SIP over Transport Layer Security (TLS).
- Overlap sending is not supported. Only Enbloc Sending is supported.
- Overlap Receiving is supported on switch types ETSI and QSIG for PRI interfaces and on switch type ETSI for BRI interfaces.
- The SR2330/4134 supports QSIG as a switch type for voice bundles on PRI interfaces only.
- The SR2330/4134 does not support supplementary services for QSIG.
- With QSIG, the SR2330/4134 does not send a FAC REJ reply to a supplementary service request from the far end.
- ISDN voice does not support TEI mode point-to-multipoint.

Standards compliance

The SR2330/4134 SIP Media Gateway supports the following standards.

SIP compliance

The SR2330/4134 supports the following SIP standards:

- RFC 3261 - SIP: Session Initiation Protocol
- RFC 3262 - Reliability of Provisional Responses in the SIP
- RFC 3264 - An Offer/Answer Model with the SDP
- RFC 2833 - RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals
- RFC 3550 - RTP: A Transport Protocol for Real-Time Applications
- RFC 3311 - The Session Initiation Protocol (SIP) UPDATE Method

ISDN compliance

The Q.930/Q.931 software supports the following standards:

- Q.930 - ISDN User-Network Interface Layer 3 Specification, 1988, CCITT.
- Q.931 - ISDN User-Network Interface Layer 3 Specification for Basic Call Control, 1988, CCITT.
- TR 41449 ATT ISDN Primary Rate Interface Specification - July 1989, AT&T.
- 235-900-321 ISDN Basic Rate Interface Specification - September 1990, AT&T.
- NIS S208-5, Basic Rate User-Network Interface Specification - Issue 1.1, 1990, Northern Telecomm.
- NIS A211-1, ISDN Primary Rate User-Network Interface Specification - Document Release 2.01, 1997, Northern Telecomm.
- SR-NWT-001959, Generic Guidelines for ISDN Terminal Equipment on Basic Access Interfaces - June 1991, Bellcore.
- NIS A211-4, ISDN Primary Rate User-Network Interface Specification Release 8.0, 1995, Northern Telecomm.
- TR-NWT-000866, ISDN Message Service Generic Requirements, Issue 1991, Bellcore.
- TR-NWT-001268, ISDN Primary Rate Interface Call Control Switching Signaling Generic Requirements for Class II Equipment, Issue 1, 1991, Bellcore.
- GR-892-CORE, Switching System Operations Generic Requirements Issue 2, December 1995, Revision 1, November 1997, Bellcore.
- NIS A233-2 NT-NI Primary Rate Interface Specification, Document Release 2.01, May 1997.

- CPE Requirements for MCI ISDN Primary Rate Interface 014-0018-MCI Telecommunications Corporation, January 11, 1996, Rev 4.0.
- ETS 300 403: Integrated Services Digital Network (ISDN); Digital Subscriber Signaling System No. one (DSS1) protocol; Signaling network layer for circuit-mode basic call control
- ETS 300 267-1: Integrated Services Digital Network (ISDN); Telephony 7 kHz and video telephony teleservices; Digital Subscriber Signaling System No. one (DSS1) protocol; Part 1: Protocol specification
- ETS 300 171/172 – QSIG Basic Call
- ETS 300 125 – QSIG Layer 2
- ETS 300 012 – QSIG Layer 1

T1 CAS compliance

The SR2330/4134 supports the following T1 CAS standards:

- TIA/EIA-464B Requirements for Private Branch Exchange (PBX) Switching Equipment

E1 R2 compliance

The SR2330/4134 supports the following E1 R2 standards:

- Specifications of Signalling System R2 ITU-T Q.400-Q.490 (11/1988)

FXS/FXO compliance

The SR2330/4134 supports the following FXS/FXO standards:

- FCC Class A device, CE
- TIA-968A Technical Requirements for Connection of Terminal Equipment to the Telephone Network

Chapter 4: SR2330/4134 Media Gateway configuration

Configure the Media Gateway to provide communication between a SIP network and the PSTN.

Prerequisites for SR2330/4134 Media Gateway configuration

- SR4134 must be running minimum Release 10.1 software.
- SR2330 must be running minimum release 10.2 software.
- Internal PVM or PVIM module must be installed in the SR2330/4134.
- Voice-capable modules (FXS, FXO, ISDN BRI, or T1/E1 [ISDN PRI, QSIG, T1 CAS, or E1 R2]) must be installed in the SR2330/4134.
- To route calls to the SIP network, a SIP server must be installed and running in the SR2330/4134 network.

SR2330/4134 Media Gateway tasks

The following work flow shows you the sequence of tasks you perform to configure the SR2330/4134 Media Gateway. This work flow applies to all supported SIP servers.

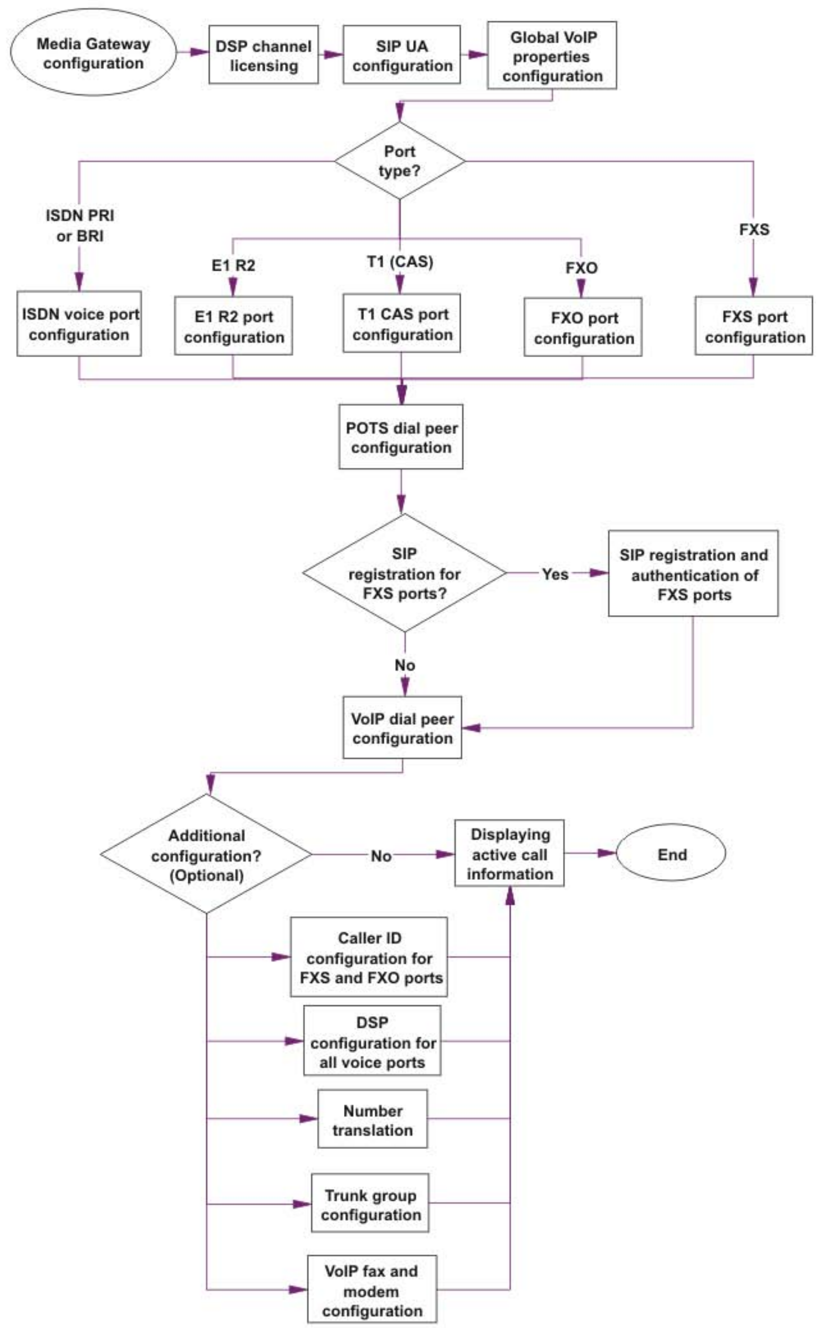


Figure 17: SR2330/4134 Media Gateway tasks

Chapter 5: SIP UA configuration

Configure the SIP UA to specify the target SIP server used for routing and to configure related SIP properties.

In addition to the configuration details in this section, see also SIP registration and authentication of FXS ports on page 141 to configure SIP registration of FXS ports.

SIP UA configuration procedures

The following task flow shows you the sequence of procedures you perform to configure the SIP UA.

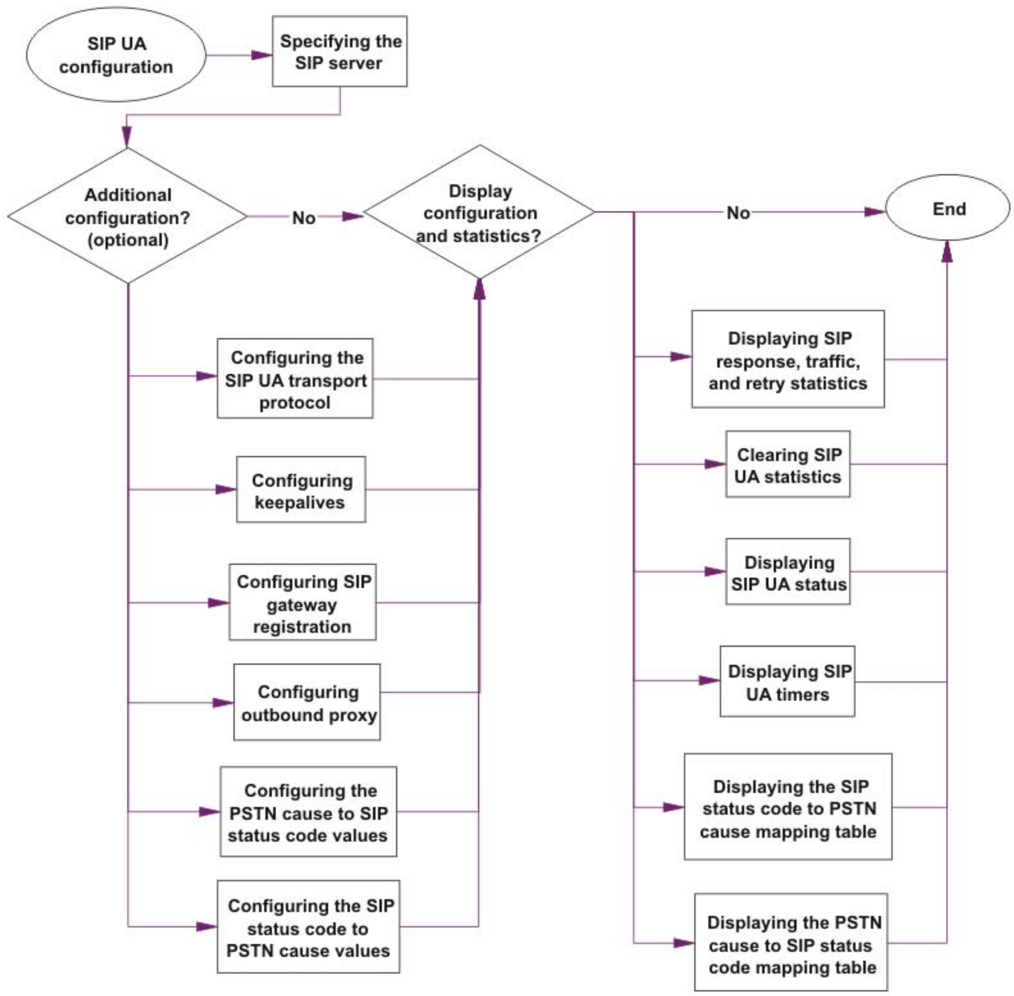


Figure 18: SIP UA configuration procedures

Specifying the SIP server

Use this procedure to configure a network address for the Session Initiation Protocol (SIP) server interface.

Procedure steps

1. To enter configuration mode, enter:

   `configure terminal`

2. To select SIP UA configuration, enter:

```
sip-ua
```

3. To specify the SIP server network address, enter:

```
[no] sip-server {dns:<host-name> | ipv4:<ip-addr>[:port-
num]}
```

Table 5: Variable definitions

Variable	Value
dns:<host-name>	Specifies a valid DNS host name for the global SIP server in the following format: name.gateway.xyz.
ipv4:<ip-addr>	Sets the global SIP server interface to an IP address. A valid IP address takes the following format: xxx.xxx.xxx.xxx.
[:port-num]	Specifies the port number to use.
[no]	Removes the configured network address.

Specifying a Secondary or Tertiary SIP server

Use this procedure to configure the network address of a secondary Session Initiation Protocol (SIP) server interface.

This command is used for the host part of SIP URIs, and it specifies the common SIP domain to be used when multiple keepalive servers are required.

Note:

If SSM is configured as a Tertiary SIP server and if both the Primary and Secondary SIP servers go DOWN, the SR/AG fails over to the Tertiary SIP server (SSM).

If you configure SSM as a Tertiary SIP server, Avaya recommends configuring the default-gateway in SSM with the gateway's bind IP, port and transport. This enables SSM to forward calls for users that are not registered with SSM to the gateway.

Procedure steps

1. To enter configuration mode, enter:

```
configure terminal
```

2. To select SIP UA configuration, enter:

```
sip-ua
```

3. To specify the SIP server network address, enter:

```
sip-server {dns:<host-name> | ipv4:<ip-addr>[:port-num]}
[secondary | tertiary]
```

Table 6: Variable definitions

Variable	Value
{dns:<host-name> | ipv4:<ip-addr>[:port-num]} [secondary | tertiary]	• dns:<host-name>: Specifies a valid DNS host name of the SIP service provider in the following format: `name.gateway.xyz`. • ipv4:<ip-addr>: Sets the global SIP domain to an IP address. A valid IP address takes the following format: `xxx.xxx.xxx.xxx`. • [:port-num] : Specifies the port number to use. • secondary: Specifies that this is a secondary SIP server. • tertiary: Specifies that this is a tertiary SIP server.
[no]	Removes the configured domain address.

Example

This example uses `sr.avaya.com` as the host name in the command, **`sip-domain dns:sr.avaya.com`**, and the following SIP servers:

- Primary SIP Server (SM1) – 47.152.227.168
- Secondary SIP Server (SM2) – 192.168.129.26
- Tertiary SIP Server (SSM) – 192.168.129.9

```
interface ethernet 0/3
    ip address 192.168.129.9 255.255.255.192
    exit ethernet
voice service voip
    sip
        bind all ipv4:192.168.129.9:5070
        rel1xx disable
        exit sip
    fax rate-management transferredTCF
    codec 1 g711ulaw 160
    no comfort-noise-negotiate
    user-param phone
    ssm
        bind ip ipv4:192.168.129.9
        enable
        sip-server
            domain dns:sr.avaya.com
            exit sip-server
       default-gateway ipv4:192.168.129.9:5070 transport tcp  <---Gateway  bind ip,
port and transport
        exit ssm
    exit voip
```

```
sip-ua
    sip-domain dns:sr.avaya.com
    sip-server ipv4:47.152.227.168:5060  <--- Primary SIP Server (SM1)
    sip-server ipv4:192.168.129.26:5060 secondary  <--- Secondary SIP Server (SM2)
    sip-server ipv4:192.168.129.9:5060 tertiary  <--- Tertiary SIP Server (SSM)
    transport tcp
    registrar ipv4:47.152.227.168 expires 3600  <--- Primary SIP Server (SM1)
    keepalive target sip-server
    keepalive target sip-server secondary
    keepalive target sip-server tertiary
        exit sip-ua
```

Configuring the SIP UA transport protocol

Use this procedure to specify whether the SIP signaling messages on inbound calls are received through the SIP TCP socket or UDP socket. By default, both TCP and UDP are enabled.

If the transport parameter changes, POTS lines that are currently registered with the SIP server reregister using the new transport parameter.

Procedure steps

1. To enter configuration mode, enter:

   `configure terminal`
2. To select SIP UA configuration, enter:

   `sip-ua`
3. To configure the SIP UA transport protocol, enter:

   `[no] transport {tcp | udp}`

Table 7: Variable definitions

Variable	Value
tcp	Specifies that the SIP dial peer uses the TCP transport layer protocol.
udp	Specifies that the SIP dial peer uses the UDP transport layer protocol.
[no]	Resets the transport protocol to the default value: UDP, TCP.

Configuring keepalives

The SR2330/4134 can monitor the state of SIP servers specified under **`sip-ua sip-server`** using a keep-alive mechanism by sending SIP OPTIONS at configurable regular intervals. The SR2330/4134 monitors both primary and, if configured, secondary SIP servers. For call routing, the Media Gateway points to the primary server as the active server, assuming the primary server is up and reachable. Otherwise, the Media Gateway points to the secondary server as the active server. If both primary and secondary servers are down or otherwise unreachable, then the active server goes into a fail-safe state and in this mode no active server is returned to the routing logic. In this case, the routing logic can check for a matching POTS dial peer. If no POTS dial peer matches, then calls are rejected with a "503 – Service Unavailable" equivalent response on the PSTN side.

When both the servers are down, prefix-based calls continue to function.

If the bound SIP interface (specified by **`voice service voip sip bind`**) goes down, then the keepalive mechanism is disabled until the link comes up. As soon as the link comes up, then the keepalive mechanism is reenabled according to the configuration.

You can turn the keepalive OPTIONS messages on or off and indicate to which SIP server they are sent. You can also configure the transport to be used to send the OPTIONS messages (UDP is default).

The following options are available for monitoring servers:

- None of the servers is monitored (no **`keepalive target`** command is executed).
- Only primary server is monitored (**`keepalive target`** command is executed only for the primary server).
- Primary and secondary servers, or primary and tertiary servers are monitored (**`keepalive target`** command is executed for both primary and secondary, or both primary and tertiary).
- Primary, secondary, and tertiary servers are monitored (**`keepalive target`** command is executed for primary, secondary, and tertiary).

Use this procedure to configure keepalives for the SIP server.

Procedure steps

1. To enter configuration mode, enter:

   ```
   configure terminal
   ```

2. To specify SIP UA configuration, enter:

   ```
   sip-ua
   ```

3. To specify the SIP server (if not already specified), enter:

   ```
   sip-server {dns:<host-name> | ipv4:<ip-addr>[:port-num]}
   [secondary | tertiary]
   ```

4. To configure the keepalive target, enter:

   ```
   keepalive target sip-server [secondary | tertiary]
   ```

5. To configure the keepalive timer, enter:

   ```
   keepalive timer <timer>
   ```

6. To configure the keepalive retries, enter:

```
keepalive retry <retries>
```

7. To configure the keepalive trigger, enter:

```
keepalive trigger <trigger>
```

Table 8: Variable definitions

Variable	Value
{sip-server {dns:<host-name> | ipv4:<ip-addr>[:port-num]} [secondary | tertiary]	• dns:<host-name>: specifies a valid DNS host name for the global SIP server in the following format: name.gateway.xyz. • ipv4:<ip-addr>: sets the global SIP server interface to an IP address. A valid IP address takes the following format: xxx.xxx.xxx.xxx. • [:port-num]: specifies the port number to use. • secondary: specifies that this is a secondary SIP server. • tertiary: specifies that this is a tertiary SIP server.
target sip-server [secondary | tertiary]	Specifies to use the configured SIP server as the target for keepalives. If the secondary option is included, both the primary and secondary servers are monitored. If the tertiary option is included, then the primary, secondary, and tertiary servers are all monitored.
<timer>	Specifies the interval at which keep-alive OPTIONS are sent to the server.
<retries>	Specifies the retry count before the server status is set to down. Range is 1-10. Default is 3.
<trigger>	Specifies a trigger count before the server status is set to up. Range is 1-10. Default is 1.

Configuring SIP gateway registration

Use this procedure to register the gateway with the SIP server. This is useful for the SIP server to track whether the gateway is up. If both primary and secondary servers are configured under **`sip-ua sip-server`**, the active server is used for registration.

To configure the username and password, you can use the **authentication** command under **sip-ua** .

Procedure steps

1. To enter configuration mode, enter:

   `configure terminal`
2. To specify SIP UA configuration, enter:

   `sip-ua`
3. To register with the SIP server, enter:

   `register dynamic`
4. To configure a username and password for the gateway, enter:

   `[no] authentication <username> <password>`

Configuring outbound proxy

Use this procedure to configure an outbound proxy address for the Session Initiation Protocol (SIP) server interface. By default, no outbound proxy is configured.

You must configure the outbound proxy before configuring a SIP UA sip-server and registrar.

Procedure steps

1. To enter configuration mode, enter:

   `configure terminal`
2. To specify SIP UA configuration, enter:

   `sip-ua`
3. To configure the outbound proxy, enter:

   `outbound-proxy {dns:<host-name> | ipv4:<ip-addr>[:port-num]}`

Table 9: Variable definitions

Variable	Value
dns:<host-name>	Specifies a valid DNS host name for the SIP proxy in the following format: name.gateway.xyz.
ipv4:<ip-addr>	Specifies the SIP proxy interface to an IP address. A valid IP address takes the following format: xxx.xxx.xxx.xxx.

Variable	Value
[:port-num]	Specifies the port number to use.

Configuring the PSTN cause to SIP status code values

Use this procedure to modify the PSTN cause to SIP status code values.

To view the existing and default values, use the **`show sip-ua map pstn-sip`** command.

Procedure steps

1. To enter configuration mode, enter:

   `configure terminal`

2. To select SIP UA configuration, enter:

   `sip-ua`

3. To configure the PSTN cause to SIP status code values, enter:

   `[no] set pstn-cause <pstn-cause> sip-status <sip-status>`

Table 10: Variable definitions

Variable	Value
<pstn-cause>	Specifies the PSTN cause value. Valid range is 1–127.
<sip-status>	Specifies the SIP status code value. Valid range is 400–699.
[no]	Resets the values to the defaults.

Configuring the SIP status code to PSTN cause values

Use this procedure to modify the SIP status code to PSTN cause values.

To view the existing and default values, use the **`show sip-ua map sip-pstn`** command.

Procedure steps

1. To enter configuration mode, enter:

   `configure terminal`

2. To select SIP UA configuration, enter:

```
sip-ua
```

3. To configure the SIP status code to PSTN cause values, enter:

```
[no] set sip-status <sip-status> pstn-cause <pstn-cause>
```

Table 11: Variable definitions

Variable	Value
<sip-status>	Specifies the SIP status code value. Valid range is 400–699.
<pstn-cause>	Specifies the PSTN cause value. Valid range is 1–127.
[no]	Resets the values to the defaults.

Displaying SIP response, traffic, and retry statistics

Use this procedure to display response, traffic, and retry SIP statistics.

Procedure steps

To display SIP statistics, enter:

```
show sip-ua statistics
```

Clearing SIP UA statistics

Use this procedure to reset the SIP UA statistical counters.

Procedure steps

To reset the SIP UA statistical counters, enter:

```
clear sip-ua statistics
```

Displaying SIP UA status

Use this procedure to display the keepalive status of the Primary, Secondary & Tertiary servers.

Procedure steps

To display the SIP UA status, enter:

```
show sip-ua status
```

Displaying SIP UA timers

Use this procedure to display the current settings for the SIP UA timers.

Procedure steps

To display the current settings for the Session Initiation Protocol (SIP) user-agent (UA) timers, enter:

```
show sip-ua timers
```

Displaying the SIP status code to PSTN cause mapping table

Use this procedure to display the SIP status code to PSTN cause mapping table.

Procedure steps

To display the SIP status code to PSTN cause mapping table, enter:

```
show sip-ua map sip-pstn
```

Displaying the PSTN cause to SIP status code mapping table

Use this procedure to display the PSTN cause to SIP status code mapping table.

Procedure steps

To display the PSTN cause to SIP status code mapping table, enter:

```
show sip-ua map pstn-sip
```

Chapter 6: Global VoIP properties configuration

Configure the global VoIP properties to enable the flow of voice traffic on the SR2330/4134 and to configure related properties. This configuration includes binding an SR2330/4134 interface IP address as the source address for SIP traffic on the SR2330/4134.

In addition to the configuration details contained in this section, see also the following sections for related configuration:

- VoIP dial peer configuration on page 147: to create a VoIP dial peer and configure related properties.
- VoIP fax and modem configuration on page 177: to configure fax and modem properties for the VoIP dial peer.

Global VoIP properties configuration procedures

The following task flow shows you the sequence of procedures you perform to configure the global VoIP properties.

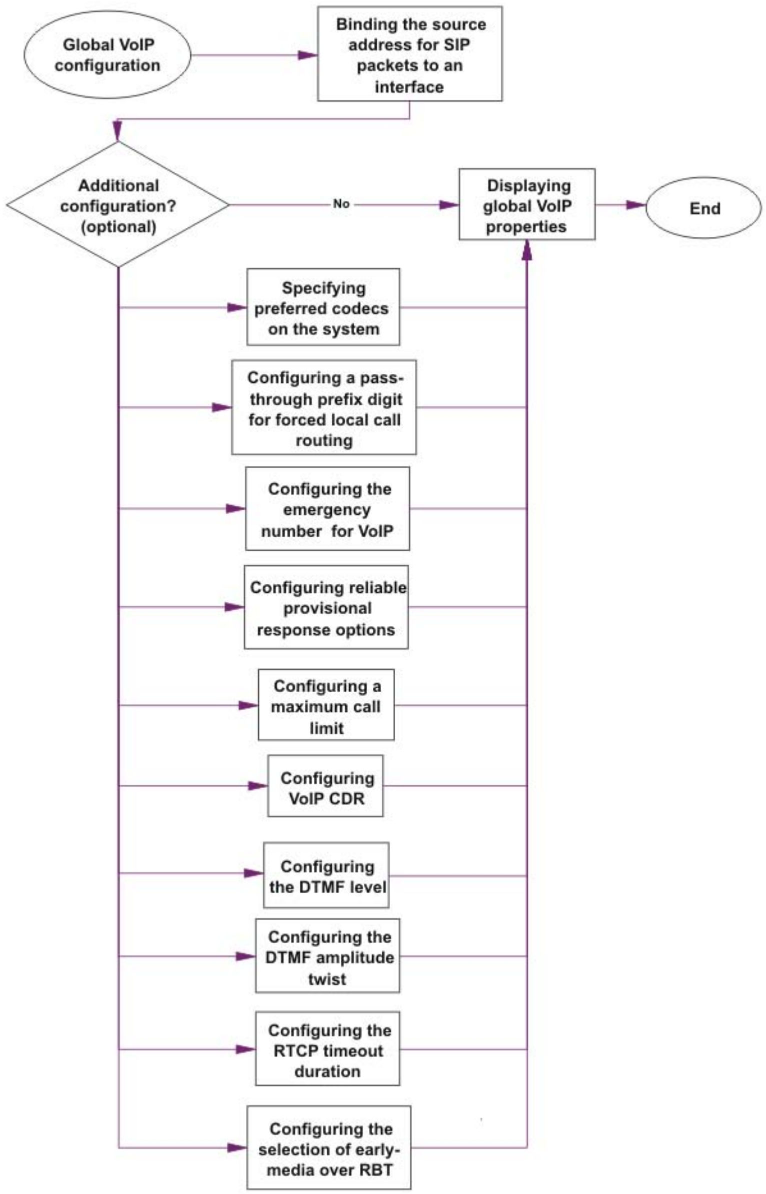


Figure 19: Global VoIP properties configuration procedures

Binding the source address for SIP packets to an interface

Use this procedure to bind the source address for SIP signaling and media packets to the IP address of a specific interface. The SR2330/4134 uses this IP address as the source address for SIP signaling. This configuration instructs the interface to listen for SIP signaling and media packets.

You must configure at least one interface to listen for SIP signaling and media traffic; otherwise, the Media Gateway cannot recognize incoming SIP traffic and is isolated from the VoIP network. You can configure one interface to receive both signaling and media traffic, or one interface to receive the signaling traffic and a second interface to receive the media traffic. The latter option allows load-sharing of the SIP traffic.

However, you can bind only one interface to receive a particular type of traffic. For example, you cannot specify multiple interfaces to load-share the SIP signaling traffic.

Important:

To change the IP address of a bound interface, you must first unbind the interface (using the **`no bind`** command) and then assign the new IP address. You can then reenter the **`bind`** command to bind to the new IP address.

If you change the interface IP address first and then enter the **`no bind`** command, the command fails because the bound IP address has changed. In this case, you must reset the IP address to the original value, and then enter the **`no bind`** command.

By default, no binding is configured.

Procedure steps

1. To enter configuration mode, enter:

   `configure terminal`

2. To select VoIP service configuration, enter:

   `voice service voip`

3. To select SIP configuration, enter:

   `sip`

4. To bind the source address for SIP packets to the IP address of a specific interface, enter:

   `[no] bind {control | media | all} <source-ip>`

Table 12: Variable definitions

Variable	Value
control	Binds only SIP signaling packets.
media	Binds only media packets.
all	Binds SIP signaling and media packets. The source address (the address that shows where the SIP request came from) of the signaling and media packets is set to the IP address of the specified interface.
<source-ip>	Specifies an address as the source address of SIP packets.

Variable	Value
[no]	Removes the binding.

Specifying preferred codecs on the system

Use this procedure to specify a list of preferred codecs to use on the system.

By default, if you do not specify preferred codecs to use, the SR2330/4134 uses G.711 u-law as preference 1 and G.711 a-law as preference 2. When you specify any preferred codecs, your configuration overrides both of these settings.

Procedure steps

1. To enter configuration mode, enter:

   ```
   configure terminal
   ```

2. To select VoIP service configuration, enter:

   ```
   voice service voip
   ```

3. To specify a list of preferred codecs to use on the system, enter:

   ```
   codec <preference> {g711alaw | g711ulaw | g729r8 | g723r53 |
   g723r63| g726r16| g726r24| g726r32} [<bytes>]
   ```

Table 13: Variable definitions

Variable	Value
<preference>	Specifies the order of preference for the specified codec, with 1 as the most preferred and 8 as the least preferred.
{g711alaw | g711ulaw | g729r8 | g723r53 | g723r63| g726r16| g726r24| g726r32}	Specifies the codec. Values are as follows: • g711alaw: G.711 a-law 64 000 bps • g711ulaw: G.711 u-law 64 000 bps • g723r63: G.723.1 6300 bps • g723r53: G.723.1 5300 bps • g726r16: G.726 16 000 bps • g726r24: G.726 24 000 bps • g726r32: G.726 32 000 bps • g729r8: G.729 ANNEX-A & B 8000 bps
[<bytes>]	Specifies the size of the voice frame in bytes, from 10 to 240.

Variable	Value
[no]	Removes the specified codec from the list.

Procedure job aid

The following table provides a comparison of the capabilities of each supported codec.

Table 14: Codec properties

CLI option	CLI comment	Voice data bytes per frame	ptime
g711alaw	G.711 a-law 64 000 bps	80, 160 (default), or 240	10, 20, or 30
g711ulaw	G.711 u-law 64 000 bps	80, 160 (default), or 240	10, 20, or 30
g723r53	G.723.1 5300 bps	20 (default), or 40	30, or 60
g723r63	G.723.1 6300 bps	24 (default), or 48	30, or 60
g726r16	G.726 16 000 bps	20, 40 (default), 60, 80, 100, 120	10, 20, 30, 40, 50, or 60
g726r24	G.726 24 000 bps	30, 60 (default), 90, 120, 150, 180	10, 20, 30, 40, 50, or 60
g726r32	G.726 32 000 bps	40, 80 (default), 120, 160, 200, or 240	10, 20, 30, 40, 50, or 60
g729r8	G.729 8000 bps	10, 20 (default), 30, 40, 50, 60, 70, or 80	10, 20, 30, 40, 50, 60, 70, or 80

Configuring a pass-through prefix digit for forced local call routing

Use this procedure to specify a pass-through prefix digit. A user can enter the configured prefix digit to force calls to be routed locally to the PSTN through directly connected PSTN interfaces.

By default, pass-through digit functionality is disabled.

Procedure steps

1. To enter configuration mode, enter:

   ```
   configure terminal
   ```

2. To select VoIP service configuration, enter:

```
voice service voip
```

3. To configure a pass-through prefix digit, enter:

```
[no] pass-through-call-prefix <prefix-digit>
```

Table 15: Variable definitions

Variable	Value
<prefix-digit>	Specifies a single prefix digit in the range 0-9.
[no]	Resets the pass-through digit functionality to the default state: disabled.

Configuring the emergency number

Use this procedure to configure the SR2330/4134 emergency number.

When the emergency number is dialed, the SR2330/4134 first attempts to match the dialed digits to a local POTS dial peer. If the SR2330/4134 can match the dialed digits to a POTS dial peer, then the call is routed directly to the associated trunk without signaling the central SIP server. However, if no matching POTS dial peer is found, then the SR2330/4134 forwards the call to the SIP server for routing.

To enable emergency call routing, you must configure a POTS dial peer for the required digits and specify those same digits as the emergency number as described in the following procedure. Otherwise, the call is treated as a standard call and is forwarded to the SIP server for routing.

By default, no emergency number is specified.

Procedure steps

1. To enter configuration mode, enter:

```
configure terminal
```

2. To select VoIP service configuration, enter:

```
voice service voip
```

3. To configure the emergency number, enter:

```
[no] emergency-number <emergency-number>
```

Table 16: Variable definitions

Variable	Value
<emergency-number>	Specifies the emergency number. The maximum string size is 6 characters.
[no]	Removes the configured emergency number.

Configuring reliable provisional response options

Use this procedure to configure reliable delivery of provisional responses.

By default, the SR2330/4134 sends reliable provisional responses to the remote SIP endpoint. You can use this procedure to disable this functionality.

Procedure steps

1. To enter configuration mode, enter:

   `configure terminal`

2. To select VoIP service configuration, enter:

   `voice service voip`

3. To select SIP configuration, enter:

   `sip`

4. To configure reliable provisional response options, enter:

   `[no] rel1xx disable`

Table 17: Variable definitions

Variable	Value
disable	Disables support of reliable provisional responses.
[no]	Enables support of reliable provisional responses.

Configuring call restriction options

Use this procedure to configure incoming SIP call restriction on gateway, to accept calls from a configured SIP Server or session targets only.

By default gateway accepts SIP requests from any IP address.

Procedure steps

1. To enter configuration mode, enter:

   `configure terminal`

2. To select VoIP service configuration, enter:

   `voice service voip`

3. To select SIP configuration, enter:

   `sip`

4. To configure call restrict options, enter:

   `[no] call-restrict incoming`

Table 18: Variable definitions

Variable	Value
incoming	Enables call restriction to receive calls from a configured SIP server or session targets only.
[no]	Disables call restriction

Configuring RTP source port validation

Use this procedure to disable or enable RTP source port validation.

By default, RTP source port validation is enabled.

When negotiating a SIP session, each SIP peer generates a Session Decription Protocol (SDP) offer/answer message that specifies the media streams and codecs it wishes to use, as well as the desired IP addresses and ports to use to receive the media.

When the session is established and the SR2330/4134 receives the RTP stream from the peer, by default, the SR2330/4134 always verifies that the port from which the stream is sent matches the advertised port in the SDP offer/answer. This provides an additional layer of protection for the VoIP calls.

However, the SDP offer/answer message specifies only the receiving port, not the sending port. While peers typically send and receive the RTP stream on the same port, they are not required to do so. As a result, if the peer sends the RTP stream from a port that is different from the advertised receiving port, the SR2330/4134 interprets this as an RTP source port mismatch and does not forward the media to the connected TDM ports. This results in no speech path for the TDM ports.

To identify whether RTP port validation is the source of a speech path issue, you can perform a packet capture and analyze whether the peer is sending and receiving the RTP stream from the same port.

If the packet capture shows different sending and receiving RTP source ports, disable RTP source validation.

Procedure steps

1. To enter configuration mode, enter:

   `configure terminal`

2. To select VoIP service configuration, enter:

   `voice service voip`

3. To configure RTP source port validation, enter:

   `[no] rtp port-validation`

Table 19: Variable definitions

Variable	Value
no rtp port-validation	Disables the RTP source port validation of the peer.
rtp port-validation	Enables the RTP source port validation of the peer.

Configuring the DTMF level

Use this procedure to configure the DTMF level.

Procedure steps

1. To enter configuration mode, enter:

   `configure terminal`

2. To select voice DSP configuration, enter:

```
voice dsp
```

3. To adjust the DTMF level, enter:

```
dtmf-level <dtmf-level>
```

Table 20: Variable definitions

Variable	Value
<dtmf-level>	Specifies the DTMF level adjustment in 0.1dB steps (enter as step-in-dB*10). Range is -60 - 70.

Configuring the DTMF amplitude twist

Use this procedure to configure the DTMF amplitude twist.

Procedure steps

1. To enter configuration mode, enter:

```
configure terminal
```

2. To select voice DSP configuration, enter:

```
voice dsp
```

3. To adjust the DTMF amplitude twist, enter:

```
dtmf-twist <dtmf-twist>
```

Table 21: Variable definitions

Variable	Value
<dtmf-twist>	Specifies the amplitude twist value in 0.1dB steps (enter as step-in-dB*10). Range is between -30 and 30. Twist = mpOfSecFreq - AmpOfPriFreq -3.3dBm

Configuring the RTCP timeout duration

When the call is on hold and no RTCP is received for a configured duration, the call is disconnected. Use this procedure to set the duration before the call is disconnected.

Procedure steps

1. To enter configuration mode, enter:

   `configure terminal`

2. To select voice DSP configuration, enter:

   `voice dsp`

3. To adjust the DTMF level, enter:

   `no-rtcp-timeout <no-rtcp-timeout>`

Table 22: Variable definitions

Variable	Value
<no-rtcp-timeout>	Specifies the value of the timeout in seconds (default 180).

Configuring the selection of early-media over RBT

Use this procedure to enable early media over ring back tone (RBT).

Procedure steps

1. To enter configuration mode, enter:

   `configure terminal`

2. To select VoIP service configuration, enter:

   `voice service voip`

3. To enable early media over RBT, enter:

   `[no] monitor-early-media`

Table 23: Variable definitions

Variable	Value
[no]	Disables early media over RBT.

Configuring VoIP CDR

The SR2330/4134 can keep track of successful and failed calls using Call Detailed Record (CDR).

The SR2330/4134 has a syslog infrastructure for logging other call events. The SR2330/4134 provides CDR storage using the same mechanism. Each field of the CDR is collected during the call and after a call is completed, the record is dispatched to the syslog server. All successful and failed calls are captured. As expected, failed calls may not have all data available. If data for a field is not available, the field displays “n/a”.

The following table describes the supported CDR fields. The following fields are expected to be available for all successful calls: Call Id, Start Time, Direction, Disconnect Time, Disconnect reason.

Table 24: CDR field definitions

CDR field	Definition
Call Id	Provides a unique identifier, used internally (0-3199).
Start Time	Specifies the time the call originates.
Orig Called Number	Specifies the original called number. The SR2330/4134 may translate this number using the translation profile feature or digit manipulation (or both) in the dial-peer before sending it out to the other endpoint.
Orig Calling Number	Specifies the original calling number. The SR2330/4134 may translate this number using the translation profile feature or digit manipulation (or both) in the dial-peer before sending it out to the other endpoint .
Direction	Specifies the direction of the call as either incoming or outgoing. The direction is in relation to the TDM endpoint; that is, incoming corresponds to a TDM-to-SIP call and outgoing corresponds to a SIP-to-TDM call.
Setup time	Specifies the time at which the call is setup; that is, the call control engine is made aware of this by the corresponding stack.
Connect time	Specifies the time at which the two parties connect the call. Two-way voice path must be established at this point. However, there can be some exceptions where the media mode is explicitly set to otherwise and two-way media is not flowing although both parties are connected.
Disconnect time	Specifies the time at which the call is disconnected by either the TDM endpoint or the SIP endpoint.
Disconnect reason	Specifies the reason for the disconnect.

The format of the time is YYYY-MM-DD hh:mm:ss.

Each CDR record begins with a syslog preamble that is separated from the CDR with a colon (:). The CDR information is stored in a comma-delimited fashion starting with the call ID information. The call ID information is useful for debugging purpose by the field engineers rather than the end user (customer).

The following is an example of a CDR record with a syslog preamble:

```
Mar 25 15:13:19 192.168.9.234
Mar 25 20:09:20  SR1 VOIP-CDR notice:
Call Id 992,2008-03-25 20:08:59,6135559999,n/a,incoming,2008-03-25
20:09:05,2008-03-25 20:09:08,2008-03-25 20:09:20,Normal
```

The CDR administrator can easily import the CDR csv format to a spreadsheet and remove the first field (syslog preamble) without losing any useful information.

The SR2330/4134 being a VoIP gateway, it always uses a two call-leg model for hairpin calls. Therefore, a TDM-to-TDM hairpin call results in two CDR entries: one for the TDM-to-SIP call leg another for the SIP-to-TDM call leg.

Use the following procedure to configure VoIP CDR.

Procedure steps

1. To enter configuration mode, enter:

   `configure terminal`

2. To select system logging configuration, enter:

   `system logging syslog`

3. To configure VoIP CDR, enter:

   `module voip-cdr sys9 sys10 sys11 sys12 sys13 sys14 local0 local1 local2 local3 local4 local5 local6 local7`

Table 25: Variable definitions

Variable	Value
sys9	indicates system use
sys10	indicates system use
sys11	indicates system use
sys12	indicates system use
sys13	indicates system use
sys14	indicates system use
local0	locally defined messages
local1	locally defined messages
local2	locally defined messages
local3	locally defined messages
local4	locally defined messages

Variable	Value
local5	locally defined messages
local6	locally defined messages
local7	locally defined messages

Displaying global VoIP properties

Use this procedure to display global VoIP properties.

Procedure steps

To display global VoIP properties, enter:

```
show voice service voip
```

Chapter 7: ISDN voice port configuration

Configure the ISDN port to enable voice communication with a PSTN CO using ISDN.

ISDN voice port configuration procedures

The following task flow shows you the sequence of procedures you perform to configure an ISDN port.

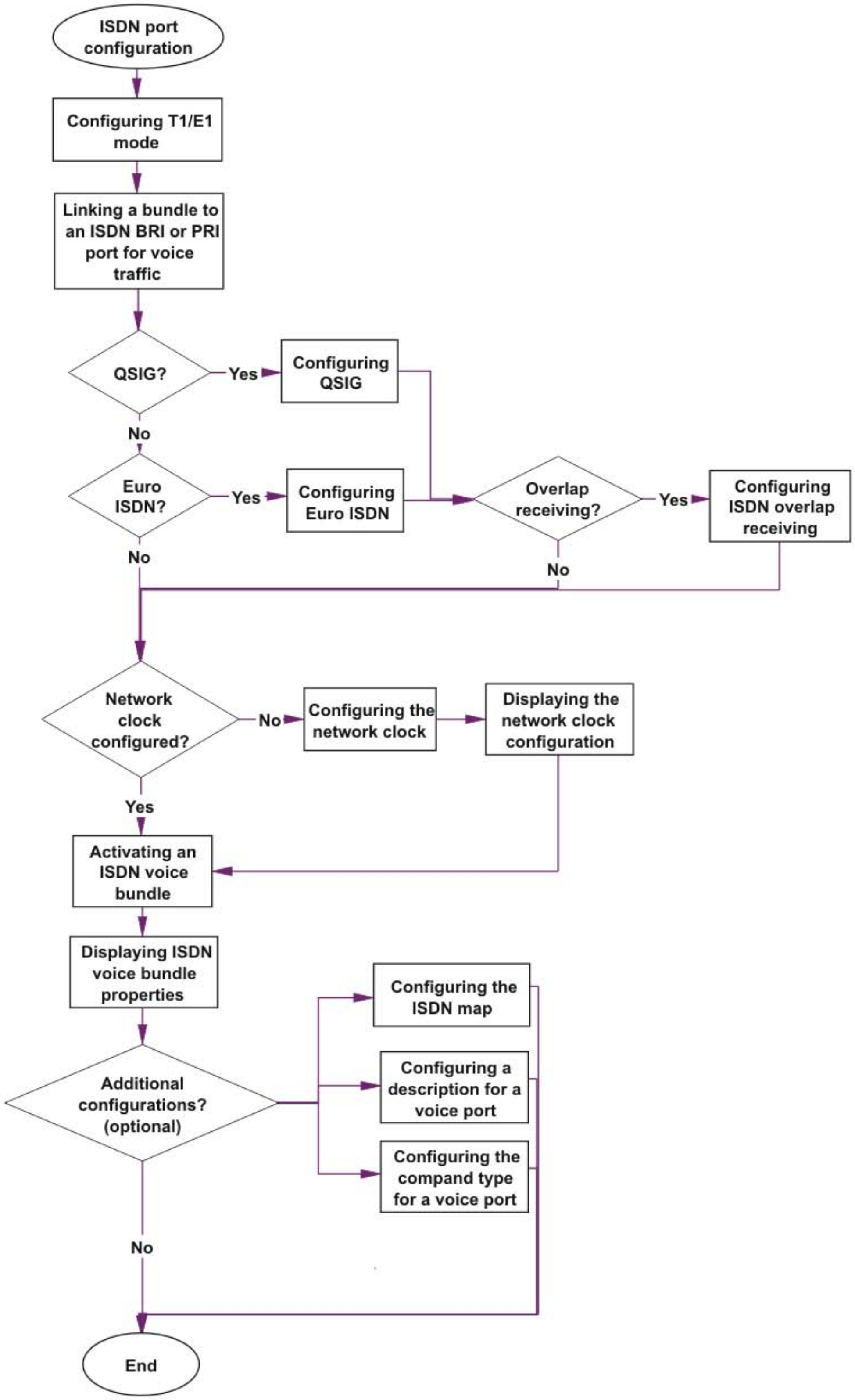


Figure 20: ISDN voice port configuration procedures

Configuring T1/E1 mode

Use the following procedure to configure the mode on the T1/E1 module to support T1 or E1 links. You must reboot the SR2330/4134 in order for the configuration to take effect.

Procedure steps

1. To enter configuration mode, enter:

   ```
   configure terminal
   ```

2. To select the module and carrier type, enter:

   ```
   system carrier-type <slot> {t1 | e1}
   ```

3. To exit the configuration mode, enter:

   ```
   exit
   ```

4. To reboot the SR2330/4134, enter:

   ```
   reboot
   ```

Linking a bundle to an ISDN BRI or PRI port for voice traffic

Use the following procedure to link an interface bundle to an ISDN port for voice. On T1 ports, ISDN PRI and T1 CAS are mutually exclusive. On E1 ports, ISDN PRI and E1 R2 are mutually exclusive.

The SR2330/4134 supports ISDN PRI (on T1/E1 ports) and BRI signaling for voice.

Procedure steps

1. To enter configuration mode, enter:

   ```
   configure terminal
   ```

2. To create a logical interface bundle, enter:

   ```
   interface bundle <bundle-name>
   ```

3. To link the bundle to an ISDN port for voice, enter:

   ```
   link {pri_t1 <slot/port[<:timeslots>]> | pri_e1 <slot/port
   [<:timeslots>]> | bri <slot/port:links>} voice
   ```

4. To specify ISDN configuration, enter:

   ```
   isdn
   ```

5. To specify the switch type, enter:

   `switch-type <switch-type>`

6. To specify the side, enter:

   `side <side-type>`

Table 26: Variable definitions

Variable	Value
pri_e1 <slot/port [<:timeslots>]>	Specifies an E1 link and timeslots for ISDN PRI.
pri_t1 <slot/port [<:timeslots>]>	Specifies T1 link and timeslots for ISDN PRI.
bri <slot/port:links>	Specifies ISDN BRI link. You can specify 1 (64Kb/s) or 2 (128 Kb/s) links.
<switch-type>	Valid switch-type values for voice are as follows: • basic-ni: National ISDN Switch type • basic-dms: Genband DMS-100 switch type • basic-5ess: AT&T basic rate switch type (default for BRI) • primary-ni2: National ISDN 2 primary rate switch type • primary-dms100: DMS100 primary rate switch type • primary-5ess: AT&T primary switch type (default) • basic-euro: EURO ISDN for BRI • primary-euro: EURO ISDN for PRI • primary-ntt: NTT (Japan) primary rate switch type
<side-type>	Valid options are: • USR: User side • NET: Network side (not supported for BRI)

Example of linking a bundle to an ISDN BRI or PRI port for voice traffic

The following example shows the configuration of a T1 link for ISDN PRI voice:

```
interface bundle isdn_pri
link pri_t1 4/1 voice
isdn
switch-type primary-5ess
side USR
activate
```

Configuring QSIG

Use the following procedure to configure QSIG on a T1 PRI or E1 PRI port.

Although QSIG is supported on T1 PRI ports, it is typically configured on E1 PRI ports. QSIG is not supported on BRI interfaces.

Procedure steps

1. To enter configuration mode, enter:

   `configure terminal`
2. To configure an interface bundle, enter:

   `interface bundle <bundle-name>`
3. To link the bundle to an ISDN port for voice, enter:

   `link {pri_e1 <slot/port [<:timeslots>]> | pri_t1 <slot/port [<:timeslots>]>} voice`
4. To specify ISDN configuration, enter:

   `isdn`
5. To specify QSIG as the switch type, enter:

   `switch-type primary-qsig`
6. To activate the bundle, enter:

   `activate`

Configuring ISDN overlap receiving

Use the following procedure to configure overlap receiving on an ISDN voice bundle. Overlap receiving is supported on BRI, E1 ports configured for Euro ISDN, as well as E1 QSIG interfaces.

By default, ISDN ports use en bloc signaling.

Procedure steps

1. To enter configuration mode, enter:

   ```
   configure terminal
   ```

2. To configure an interface bundle, enter:

   ```
   interface bundle <bundle-name>
   ```

3. To link the bundle to an ISDN port for voice, enter:

   ```
   link {pri_e1 <slot/port [<:timeslots>]> | bri <slot/
   port:links>} voice
   ```

4. To specify ISDN configuration, enter:

   ```
   isdn
   ```

5. To specify the switch type, enter:

   ```
   switch-type {basic-euro | primary-euro | primary-qsig}
   ```

6. To specify overlap receiving, enter:

   ```
   [no] overlap-receive
   ```

7. To specify q931-timers configuration, enter:

   ```
   q931-timers
   ```

8. To configure the t302 timer, enter:

   ```
   t302 <t302-timer>
   ```

9. To exit q931 timers configuration, enter:

   ```
   exit
   ```

10. To activate the ISDN bundle, enter:

    ```
    activate
    ```

Table 27: Variable definitions

Variable	Value
[no] overlap-receive	The no form of the command changes the mode of signaling to en bloc.
t302 <t302-timer>	Specifies the t302 timer value. Range is from 3 to 15 seconds. Default: 15 seconds.

Activating an ISDN voice bundle

Use the following procedure to activate the ISDN voice port.

If you change any of the ISDN properties, you must deactivate ISDN (using the **`no activate`** command), and then reactivate ISDN for the changes to be applied.

Procedure steps

1. To enter configuration mode, enter:

   `configure terminal`

2. To specify the ISDN bundle to activate, enter:

   `interface bundle <bundle-name>`

3. To specify ISDN configuration, enter:

   `isdn`

4. To activate ISDN, enter:

   `[no] activate`

Configuring the ISDN map

Use the procedure to override the default ISDN type and plan generated by the SR2330/4134 with custom values.

Procedure steps

1. To enter configuration mode, enter:

   `configure terminal`

2. To specify bundle configuration, enter:

   `interface bundle <bundle-name>`

3. To specify ISDN configuration, enter:

   ```
   isdn
   ```

4. To configure the ISDN mapping, enter:

   ```
   map <address> <plan> <type>
   ```

Table 28: Variable definitions

Variable	Value
[no]	Reverts to the default ISDN type and plan.
<address>	Specifies either the calling number or the called number. This parameter can be a regular expression also for pattern matching. It specifies that the ISDN type and plan will be overridden for addresses that match the regular expression.
<plan>	Specifies the numbering plan: • Unknown: unknown with bit value 0000 • isdn: ISDN/telephony numbering- E.164/ E.163 with bit value 0001 • tel: telephony numbering—E.163 with bit value 0010 • data: data numbering—X.121 with bit value 0011 • telex: telex numbering—Recommendation F.69 with bit value 0100 • national: national standard numbering with bit value 1000 • private: private numbering with bit value 1001
<type>	Specifies the type: • unknown: unknown with bit value 000 • international: international number with bit value 001 • national: national number with bit value 010 • network: network specific number with bit value 011 • subscriber: subscriber number with bit value 100

Variable	Value
	• overlap: overlap sending with bit value 1001 • abbreviated: abbreviated number with bit value 110

Displaying ISDN voice bundle properties

The commands to display ISDN bundle properties include the following:

- **`show interface bundles`**: displays the bundle configurations
- **`show interface bundle <bundle-name>`**: displays a specific bundle configuration
- **`show isdn interfaces`**: displays the ISDN properties for all bundles
- **`show isdn interface <bundle-name>`**: displays the ISDN properties for a specific bundle
- **`show voice port {slot/[subslot]port:[<d-channel>] | summary}`**: displays the voice port configuration (<d-channel> = 23 for T1 PRI and 15 for E1 PRI)
- **`show module configuration {e1|t1} <slot/port>`**: displays the e1 or t1 module configuration

Chapter 8: T1 CAS port configuration

Configure the T1 CAS port to enable voice communications with the PSTN or with a PBX using Channel Associated Signaling (CAS).

In addition to the configuration details contained in this section, see also DSP configuration for all voice ports on page 161 to configure DSP related properties for the voice port.

T1 CAS port configuration procedures

The following task flow shows you the sequence of procedures you perform to configure a T1 CAS port.

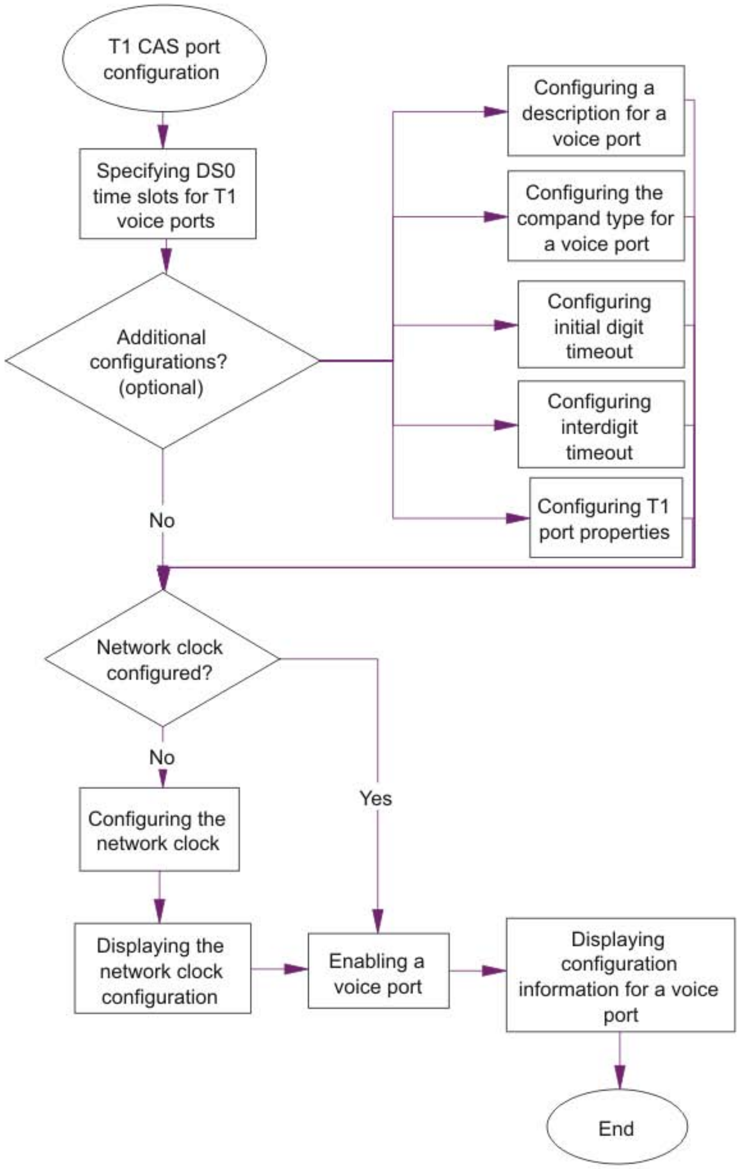


Figure 21: T1 CAS port configuration procedures

Specifying DS0 time slots for T1 voice ports

Use this procedure to specify the DS0 time slots that make up a logical voice port on a T1 module and to specify CAS as the signaling type by which the logical voice port communicates with the PBX or PSTN.

Prerequisites

- Ensure the T1/E1 module is in T1 mode. If it is not, you can set the module to T1 mode by entering the **`system carrier-type <slot> t1`** command and rebooting the SR2330/4134.

Procedure steps

1. To enter configuration mode, enter:

   `configure terminal`

2. To select a T1 module to configure, enter:

   `module t1 <slot/port>`

3. To specify the DS0 time slots and signaling on the T1 module, enter:

   `[no] cas-group timeslots <timeslot-list> {em-delay-dial | em-wink-start | em-immediate-start }`

Table 29: Variable definitions

Variable	Value
<timeslot-list>	Lists time slots in the DS0 group. Valid values include a single time-slot number, a single range of numbers, or multiple ranges of numbers separated by commas, within the range of 1 to 24 for T1. For example: • 2 • 1-15,17-24 • 1-23 • 2, 4, 6-12
{em-delay-dial | em-wink-start | em-immediate-start }	Specifies the type of signaling for the CAS group. The ear and mouth (E&M) interface allows connection for PBX trunk lines (tie lines) and telephone equipment. Valid types are as follows: • em-delay-dial: The originating endpoint sends an offhook signal and then waits for an off-hook signal followed by an on-hook signal from the destination. • em-immediate-start: E&M immediate start. • em-wink-start: The originating endpoint sends an offhook signal and waits for a wink-start from the destination.

Variable	Value
[no]	Removes the group and signaling settings. To modify an existing T1 CAS configuration, you must first remove the existing configuration using this option.

Example of specifying DS0 time slots for T1 voice ports

```
module t1 1/2
cas-group timeslots 1-24 em-delay-dial
exit
```

Configuring T1 port properties

Configuring T1 framing

Set the framing mode for a T1 link.

Procedure steps

1. To enter the configuration mode, enter:

   `configure terminal`
2. To select the T1 module port to configure, enter:

   `module t1 <slot/port>`
3. To configure framing, enter:

   `framing {esf | d4}`

Table 30: Variable definitions

Variable	Value
esf	Extended Super Frame framing format for T1 (default)
d4	Super Frame framing format for T1

Configuring T1 linecode

Set the type of line coding for a T1 link.

Procedure steps

1. To enter the configuration mode, enter:

   `configure terminal`

2. To select the T1 module port to configure, enter:

   `module t1 <slot/port>`

3. To configure the linecode, enter:

   `linecode {b8zs | ami}`

Table 31: Variable definitions

Variable	Value
ami	AMI linecode for T1
b8zs	B8Zs linecode for T1 (default).

Configuring T1 yellow alarm detection and generation

Set the yellow alarm operation on a T1 link.

Procedure steps

1. To enter the configuration mode, enter:

   `configure terminal`

2. To select the T1 module port to configure, enter:

   `module t1 <slot/port>`

3. To configure yellow alarm, enter:

   `yellow_alarm {generate | detect | gen_det | disable}`

Table 32: Variable definitions

Variable	Value
detect	Detect incoming yellow alarms from the network.
disable	Disable yellow alarm generator.
gen_det	Generate and detect yellow alarms (default state).
generate	Generate and send yellow alarms to the network.

Configuring T1 clock source

Configure the clock source to set the network timing source for a T1 link.

Procedure steps

1. To enter the configuration mode, enter:

   `configure terminal`

2. To select the T1 module port to configure, enter:

   `module t1 <slot/port>`

3. To configure the clock source, enter:

   `clock_source {network | internal | line}`

Table 33: Variable definitions

Variable	Value
network	Sets the clock source to be recovered from the incoming T1 signal (voice only).
internal	Sets the clock source to the internal clock of the SR2330/4134 (default).
line	Sets the clock source to be recovered from the incoming T1 signal (data only).

Configuring T1 alarm thresholds

Set the T1 user statistic alarm thresholds.

When user-configurable thresholds are exceeded, the SR2330/4134 generates alarms that indicate the possible deterioration of a T1 link. Refer to the following parameters to determine the specific T1 data type that needs to be configured. You can define one alarm threshold for each parameter.

Procedure steps

1. To enter the configuration mode, enter:

   `configure terminal`

2. To select the T1 module port to configure, enter:

   `module t1 <slot/port>`

3. To configure the alarm for user statistics, enter:

```
alarms thresholds user <1 - 10>
[ ses | es | uas | eev | css | oof | crc | bpv ]
<sampling-interval> <rising-threshold> <falling-threshold>
sample_type [ absolute | delta ]
```

Table 34: Variable definitions

Variable	Value
1-10	Alarm threshold number.
absolute | delta	absolute: The errored second or event count is compared directly to the specified threshold values, and the appropriate alarm type (rising or falling) is reported. delta: The errored second or event count is compared to the difference between the rising and falling thresholds, and a rising alarm is reported if the actual error count exceeds that difference. This is the default setting if you do not specify a sampling type.
ses | es | uas | eev | css | oof | crc | bpv	Specifies the threshold type: bpv: Threshold for bipolar violation crc: Threshold for cyclic redundancy check css: Threshold for controlled slip second eev: Threshold for esf error events es: Threshold for errored seconds oof: Threshold for out of frame ses: Threshold for severely errored seconds uas: Threshold for unavailable seconds
falling-threshold	Minimum number of errored seconds or events below which a falling alarm is reported. This alarm is reported if a rising alarm was previously reported and the number of errored seconds or events subsequently dropped below this minimum threshold. The falling threshold value must be less than the rising threshold value above. The range is 0 - 2147483647.
rising-threshold	Number of errored seconds or events which, if exceeded during any sampling interval, results in a rising alarm. The range is 0 - 2147483647
sampling-interval	Sampling interval, in seconds. The range is 1 - 65535.

Configuring hierarchy for T1 alarms

Enable or disable the hierarchy for displaying Receive Loss of Signal (RLOS) and Receive Loss of Frame (RLOF) on a T1 link.

Procedure steps

1. To enter the configuration mode, enter:

```
configure terminal
```

2. To select the T1 module port to configure, enter:

```
module t1 <slot/port>
```

3. To configure hierarchy, enter:

```
[no] alarms hierarchy
```

Table 35: Variable definitions

Variable	Value
no	Disables hierarchy.

Configuring CSU line mode for T1

Set the amount of T1 line build out (LBO) for the Channel Service Unit (CSU) interface.

Procedure steps

1. To enter the configuration mode, enter:

```
configure terminal
```

2. To select the T1 module port to configure, enter:

```
module t1 <slot/port>
```

3. To configure CSU line mode, enter:

```
linemode csu lbo {db_zero | db7_5 | db15 | db22_5}
```

Table 36: Variable definitions

Variable	Value
db7_5	Configure LBO for zero db (default)
db15	Configure LBO for 7.5 db
db22_5	Configure LBO for 15 db
db_zero	Configure LBO for 22.5 db

Configuring DSX line mode for T1

Set the amount of T1 signal equalization based on the cabling distance to the DSX cross-connect.

Procedure steps

1. To enter the configuration mode, enter:

   `configure terminal`
2. To select the T1 module port to configure, enter:

   `module t1 <slot/port>`
3. To configure DSX line mode, enter:

   `linemode dsx cable-length <length>`

Table 37: Variable definitions

Variable	Value
<length>	1: specifies cable length of 0-110 2: specifies cable length of 110-220 3: specifies cable length of 220-330 4: specifies cable length of 330-440 5: specifies cable length of 440-550 6: specifies cable length of 550-660

Configuring T1 circuit ID

Specify an optional circuit ID to a T1 interface.

Procedure steps

1. To enter the configuration mode, enter:

   `configure terminal`
2. To select the T1 module port to configure, enter:

   `module t1 <slot/port>`
3. To configure the T1 circuit ID, enter:

   `circuitId [cktID] <circuit-id>`

Table 38: Variable definitions

Variable	Value
circuit-id	Optional circuit ID for the T1 channel.

Configuring contact information for T1

Specify contact information for a person who can provide information regarding the T1 link.

Procedure steps

1. To enter the configuration mode, enter:

   `configure terminal`
2. To select the T1 module port to configure, enter:

   `module t1 <slot/port>`
3. To configure the contact information, enter:

   `contactInfo <contact-info>`

Table 39: Variable definitions

Variable	Value
contact-info	Person to contact for information regarding the T1 link.

Configuring description for T1

Enter a description for the T1 interface.

Procedure steps

1. To enter the configuration mode, enter:

   `configure terminal`
2. To select the T1 module port to configure, enter:

   `module t1 <slot/port>`
3. To enter the description, enter:

   `description <port-description>`

Table 40: Variable definitions

Variable	Value
port-description	Describes the T1 interface.

Configuring a name for T1

Assign a name to the T1 link.

Procedure steps

1. To enter the configuration mode, enter:

```
configure terminal
```

2. To select the T1 module port to configure, enter:

```
module t1 <slot/port>
```

3. To assign a name to the T1 link, enter:

```
name <name>
```

Table 41: Variable definitions

Variable	Value
name	Assigns a name to the T1 link (max 15 bytes).

Configuring loopback framing for T1

Set overwriting or insertion of framing for in-band loopcode.

Procedure steps

1. To enter the configuration mode, enter:

```
configure terminal
```

2. To select the T1 module port to configure, enter:

```
module t1 <slot/port>
```

3. To configure loopback framing, enter:

```
loopback_framing {overwrite | insert}
```

Table 42: Variable definitions

Variable	Value
overwrite	Overwriting of framing for in-band loopcode for T1 (default).
insert	Insertion of framing for in-band loopcode for T1.

Chapter 9: E1 R2 port configuration

R2 signaling is a channel associated signaling (CAS) system developed in the 1960s that is still in use today in Europe, Latin America, Australia, and Asia. R2 signaling exists in several country versions or variants in an international version called Consultative Committee for International Telegraph and Telephone (CCITT-R2). The R2 signaling specifications are contained in International Telecommunication Union Telecommunication Standardization Sector (ITU-T) Recommendations Q.400 through Q.490. E1 R2 signaling is an international signaling standard that is common to channelized E1 networks. Although R2 signaling is defined in the ITU-T Q.400-Q.490 recommendations, there are many variations in how R2 is implemented. (Various countries have chosen to implement R2 differently.)

E1 R2 port configuration procedures

The following task flow shows you the sequence of procedures you perform to configure an E1 R2 port.

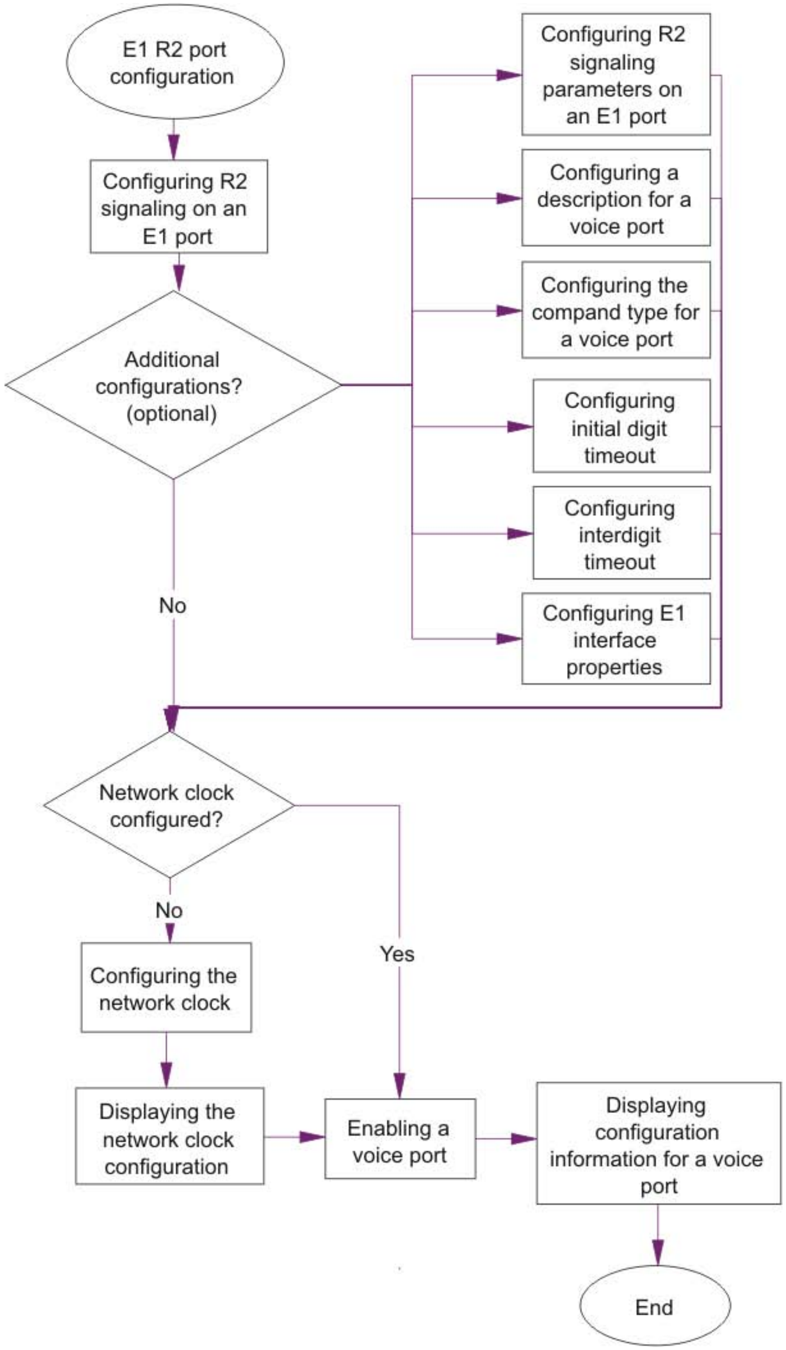


Figure 22: E1 R2 port configuration procedures

Configuring R2 signaling on an E1 port

Use the following procedure to enable R2 signaling on an E1 module.

Prerequisites

Ensure the T1/E1 module is in E1 mode. If it is not, you can set the module to T1 mode by entering the `system carrier-type <slot> e1` command and rebooting the SR2330/4134.

Procedure steps

1. To enter configuration mode, enter:

   `configure terminal`

2. To select an E1 module to configure, enter:

   `module e1 <slot/port>`

3. To specify the DS0 time slots and signaling on the E1 module, enter:

   `[no] cas-group timeslots <timeslot-list> r2-digital r2-compelled`

Table 43: Variable definitions

Variable	Value
<timeslot-list>	Lists time slots in the DS0 group. Valid values include a single time-slot number, a single range of numbers, or multiple ranges of numbers separated by commas, within the range of 1 to 31 for E1. For example: • 2 • 1-15,17-24 • 1-31 • 2, 4, 6-12
r2-digital	Specifies R2 ITU Q421 digital line signaling, which is the most common signaling configuration. The A and B bits are used for line signaling.
r2-compelled	Specifies R2 compelled Interregister Signaling. When a tone-pair is sent from the SR2330/4134 (forward signal), the tones stay on until the remote end responds (sends an ACK) with a pair of tones that signals the SR2330/4134 to turn off the tones. The tones are compelled to stay on until they are turned off.
[no]	Removes the group and signaling settings. To modify an existing E1 CAS configuration, you must first remove the existing configuration using this option.

Configuring R2 signaling parameters on an E1 port

Use the following procedure to configure the R2 signaling parameters on an E1 module.

Procedure steps

1. To enter configuration mode, enter:

   `configure terminal`
2. To select an E1 module to configure, enter:

   `module e1 <slot/port>`
3. To configure backward digits, enter:

   `cas-custom backward-digits <backward-digit-string>`
4. To configure the initial category digit, enter:

   `cas-custom category <category-string>`
5. To configure the second category digit, enter:

   `cas-custom categoryII <categoryII-string>`
6. To configure the country parameters, enter:

   `cas-custom country <country-name>`

Table 44: Variable definitions

Variable	Value
<backward-digit-string>	Specifies the string that allows the compelled signaling sequence to be modified. Digit string range: 1 – 9, A – F up to 40 digits. Default is 1116. (See Group A backward signals table.)
<category-string>	Identifies the initial category digit, which distinguishes the type of call (for example, normal subscriber, payphone, and so on) Digit string range: 1 – 9, A – F up to 2 digits. ITU default is 1 (normal subscriber). (See Group II forward signals table.)
<categoryII-string>	Identifies the second category digit, for those variants that use two category digits Digit string range: 1 – 9, A – F up to 2 digits. ITU default is 1 (normal subscriber). (See Group II forward signals table.)

Variable	Value
<country-name>	Specifies the local country, region or corporation that is used with R2 signaling. The name can be one of the following: BRAZIL, CHINA, GENERIC, KOREA, MEXICO, SINGAPORE. Default is R2_ITU GENERIC.

The following table describes the G.411 Group II forward signals.

Table 45: G. 411 Group II forward signals

Combination	Signal designation	Definition	Comments
1	II-1	Subscriber without priority	These signals are used for national working only.
2	II-2	Subscriber with priority	
3	II-3	Maintenance equipment	
4	II-4	Spare	
5	II-5	Operator	
6	II-6	Data transmission	
7	II-7	Subscriber (or operator without forward transfer facility)	These signals are used for international working.
8	II-8	Data transmission	
9	II-9	Subscriber with priority	
10	II-10	Operator with forward transfer facility	
11	II-11	These are spare signals for national use.	
12	II-12		
13	II-13		
14	II-14		
15	II-15		

The following table describes the Q.411 Group A backward signal codes.

Table 46: Q.411 Group A backward signals

Combination	Signal designation	Definition
1	A-1	Send next digit (n + 1)

Combination	Signal designation	Definition
2	A-2	Send last but one digit (n – 1)
3	A-3	Address-complete, changeover to reception of Group B signals
4	A-4	Congestion in the national network
5	A-5	Send calling party's category
6	A-6	Address-complete, charge, set-up speech conditions
7	A-7	Send last but two digit (n 2)
8	A-8	Send last but three digit (n 3)
9	A-9	Spare for national use
10	A-10	
11	A-11	Send country code indicator
12	A-12	Send language or discrimination digit
13	A-13	Send nature of circuit
14	A-14	Request for information on use of an echo suppressor (is an incoming half-echo suppressor required ?)
15	A-15	Congestion in an international exchange or at its output

The following table describes the Q.411 Group B backward signal codes.

Table 47: Q.441 Group B backward signals

Combination	Signal designation	Definition
1	B-1	Spare for national use
2	B-2	Send special information tone
3	B-3	Subscriber line busy
4	B-4	Congestion (encountered after changeover from Group A signals to Group B signals)
5	B-5	Unallocated number
6	B-6	Subscriber's line free, charge
7	B-7	Subscriber's line free, no charge
8	B-8	Subscribers line out of order
9	B-9	Spare for national use
10	B-10	

Combination	Signal designation	Definition
11	B-11	
12	B-12	
13	B-13	
14	B-14	
15	B-15	

Configuring R2 backward digits

To indicate variable number of digits a new character ‘R’ has been introduced in configuration of backward-digit. With this user can conveniently configure backward digit. ‘R’ indicates variable number of digits.

Incoming Call – variable DNIS

SR2330/4134 can now accept variable number digits in the Dialed Number Identification Service (DNIS) number or “dialed” number. For this, backward-digits need to be configured as 1R31. In prior releases, the behavior was to accept only a fixed number of digits in DNIS. For example, if backward digit is 11131 then the SR2330/4134 was able to accept a call with only 4 digits in DNIS.

'R’ indicates a variable number of digits.

For example, if the peer makes use of dial plan shown below:

International Call – 12 digits

National Call – 10 digits

Local – 7 digits

Extension – 4 digits

For above calls on the SR2330/4134, backward digit needs to configured as 1R31.

If the peer is configured to send ‘F’ indicating the end of the DNIS, then after receiving ‘F’ SR2330/4134 sends 3 (Group A backward signal) indicating that the number complete.

If the peer is configured not to send ‘F’ to indicate no more DNIS digits, then the SR2330/4134 sends 3 (Group A backward signal) indicating number complete after interdigit timeout (default 6 seconds)

SR2330/4134 can now be configured to request for Automatic Number Identification (ANI) or caller identification. Backward-digit needs to be configured as 1R61R31. The second ‘R’ in backward-digit configuration stands for variable number of digits in ANI.

1. If the peer is configured to send 4, 7, or 10 digits in ANI, then backward-digit can be configured as ‘1R61R31’ on the SR2330/4134.
2. If the peer is configured to send fixed 4 digits in ANI, then backward-digit can be configured as ‘1R61111131’. In this case after receiving 4 digits, the SR2330/4134 will send ‘3’ indication ANI is complete. If there are more digits in ANI then the SR2330/4134 will ignore rest of the digits.

The following table shows the backward digit configuration for various combinations of DNIS and ANI numbers/digits.

Call Direction	DNIS	ANI	Backward digit configuration
Incoming	4 (Fixed)	None	11131
Incoming	4 (Fixed)	4 (Fixed)	1113111131
Incoming	Variable	4 (Fixed)	1R6111131
Incoming	Variable	Variable	1R61R31
Incoming	4 (Fixed)	Variable	11161R31
Incoming	5 (Fixed)	None	111131
Incoming	5 (Fixed)	5 (Fixed)	111131111131
Incoming	Variable	5 (Fixed)	1R61111131
Incoming	Variable	Variable	1R61R31
Incoming	5 (Fixed)	Variable	111161R31

Inter-Digit Timeout Configuration

The default value of inter-digit timeout value is set to 6 seconds. The parameter is configurable for E1 R2 ports and can be configured using CLI ‘timeout interdigit’ under voice-port.

Configuring E1 interface properties

Configuring E1 framing

Set the framing mode for the E1 link.

Procedure steps

1. To enter the configuration mode, enter:

```
configure terminal
```

2. To select the E1 module port to configure, enter:

```
module e1 <slot/port>
```

3. To configure framing, enter:

```
framing {crc | non_crc | disable}
```

Table 48: Variable definitions

Variable	Value
crc	Specifies CRC framing format for E1 (default).
non_crc	Specifies non-CRC framing format for E1
disable	Disables the E1 framer.

Configuring E1 linecode

Set the type of line coding for the E1 link.

Procedure steps

1. To enter the configuration mode, enter:

```
configure terminal
```

2. To select the E1 module port to configure, enter:

```
module e1 <slot/port>
```

3. To configure the linecode, enter:

```
linecode {hdb3 | ami}
```

Table 49: Variable definitions

Variable	Value
ami	AMI linecode for E1
hdb3	HDB3 linecode for E1 (default).

Configuring E1 yellow alarm detection and generation

Configure the yellow alarm operation on the E1 link.

Procedure steps

1. To enter the configuration mode, enter:

```
configure terminal
```

2. To select the E1 module port to configure, enter:

```
module e1 <slot/port>
```

3. To configure yellow alarm, enter:

```
yellow_alarm {generate | detect | gen_det | disable}
```

Table 50: Variable definitions

Variable	Value
detect	Detect incoming yellow alarms from the network.
disable	Disable yellow alarm generator.
gen_det	Generate and detect yellow alarms (default state).
generate	Generate and send yellow alarms to the network.

Configuring E1 clock source

Configure the clock source to set the network timing source for an E1 link.

Procedure steps

1. To enter the configuration mode, enter:

```
configure terminal
```

2. To select the E1 module port to configure, enter:

```
module e1 <slot/port>
```

3. To configure the clock source, enter:

```
clock_source {internal | line | network}
```

Table 51: Variable definitions

Variable	Value
internal	Sets the clock source to the internal clock of the SR2330/4134 (default).
line	Sets the clock source to be recovered from the incoming E1 signal (loop timing).
network	Sets the clock to be recovered from the incoming E1 signal (voice only). This is set by default when an E1 interface is configured with cas-group.

Configuring E1 alarms

When thresholds are exceeded, the system generates alarms that indicate the possible deterioration of an E1 link. You can define one alarm threshold for each available parameter.

Procedure steps

1. To enter the configuration mode, enter:

   `configure terminal`

2. To select the E1 module port to configure, enter:

   `module e1 <slot/port>`

3. To configure the alarm for user statistics, enter:

   `alarms thresholds user <1 - 10>`

   `[ ses | es | uas | css | oof | crc | bpv ]`

   `<sampling-interval> <rising-threshold> <falling-threshold>`

   `sample_type [absolute | delta]`

Table 52: Variable definitions

Variable	Value
1-10	Alarm threshold number.
absolute	The errored second or event count is compared directly to the specified threshold values, and the appropriate alarm type (rising or falling) is reported.
ses | es | uas | css | oof | crc | bpv	bpv: Threshold for bipolar violation crc: Threshold for cyclic redundancy check css: Threshold for controlled slip second es: Threshold for errored seconds oof: Threshold for out of frame ses: Threshold for severely errored seconds uas: Threshold for unavailable seconds
delta	The errored second or event count is compared to the difference between the rising and falling thresholds, and a rising alarm is reported if the actual error count exceeds that difference. This is the default setting if you do not specify a sampling type.
falling-threshold	Minimum number of errored seconds or events below which a falling alarm is reported. This alarm is reported if a rising alarm was previously reported and the number of errored seconds or events subsequently dropped below this minimum threshold. The falling threshold value must be less

Variable	Value
	than the rising threshold value above. The range is 0 - 2147483647.
rising-threshold	Number of errored seconds or events which, if exceeded during any sampling interval, results in a rising alarm. The range is 0 - 2147483647
sampling-interval	Sampling interval, in seconds. The range is 1 - 65535.

Configuring hierarchy for E1 alarms

Enable or disable the hierarchy for displaying Receive Loss of Signal (RLOS) and Receive Loss of Frame (RLOF) on an E1 link.

Procedure steps

1. To enter the configuration mode, enter:

   `configure terminal`

2. To select the E1 module port to configure, enter:

   `module e1 <slot/port>`

3. To configure hierarchy, enter:

   `[no] alarms hierarchy`

Table 53: Variable definitions

Variable	Value
no	Disables hierarchy.

Configuring line mode for E1

Configure the line mode for the E1 interface.

Procedure steps

1. To enter the configuration mode, enter:

   `configure terminal`

2. To select the E1 module port to configure, enter:

   `module e1 <slot/port>`

3. To configure line mode, enter:

```
linemode {long_haul | short_haul}
```

Table 54: Variable definitions

Variable	Value
long_haul	Long haul type for linemode configuration. Operations up to 6 dB. (default).
short_haul	Short haul type for linemode configuration. Operations up to 43 dB.

Configuring E1 circuit ID

Specify an optional circuit ID for the E1 interface.

Procedure steps

1. To enter the configuration mode, enter:

   `configure terminal`

2. To select the E1 module port to configure, enter:

   `module e1 <slot/port>`

3. To configure the E1 circuit ID, enter:

   `circuitId <circuit-id>`

Table 55: Variable definitions

Variable	Value
circuit-id	Optional circuit ID for the E1 channel.

Configuring contact information for E1

Specify contact information for a person who can provide details regarding the E1 link.

Procedure steps

1. To enter the configuration mode, enter:

   `configure terminal`

2. To select the E1 module port to configure, enter:

   `module e1 <slot/port>`

3. To configure the contact information, enter:

   `contactInfo <contact-info>`

Table 56: Variable definitions

Variable	Value
contact-info	Person to contact for information regarding the E1 link.

Configuring description for E1

Enter a description for the E1 interface.

Procedure steps

1. To enter the configuration mode, enter:

   `configure terminal`

2. To select the E1 module port to configure, enter:

   `module e1 <slot/port>`

3. To enter the description, enter:

   `description <port-description>`

Table 57: Variable definitions

Variable	Value
port-description	Describes the E1 interface.

Configuring a name for E1

Assign a name to the E1 link.

Procedure steps

1. To enter the configuration mode, enter:

   `configure terminal`

2. To select the E1 module port to configure, enter:

   `module e1 <slot/port>`

3. To assign a name to the E1 link, enter:

   `name <name>`

Table 58: Variable definitions

Variable	Value
<name>	Assigns a name to the E1 link (max 16 bytes).

E1 configuration example

The following shows an E1 example configuration.

```
module e1 3/1
  cas-group timeslots 1-20 r2-digital r2-compelled
  cas-custom
    country mexico
    backward-digits 11161R31
    exit cas-custom
  exit e1
```

Chapter 10: FXO port configuration

Configure the Foreign Exchange Office (FXO) port to connect the SR2330/4134 to a TDM PBX system or to provide off-premise PSTN connections to a central office (CO).

Important:

With FXO ports, supervisory disconnect is always enabled. Unlike FXS ports, you cannot disable supervisory disconnect on FXO ports.

In addition to the configuration details contained in this section, see also the following sections for related configuration:

- Caller ID configuration for FXS and FXO ports on page 155: to configure caller ID for the voice port.
- DSP configuration for all voice ports on page 161: to configure DSP related properties for the voice port.

FXO port configuration procedures

The following task flow shows you the sequence of procedures you perform to configure an FXO port.

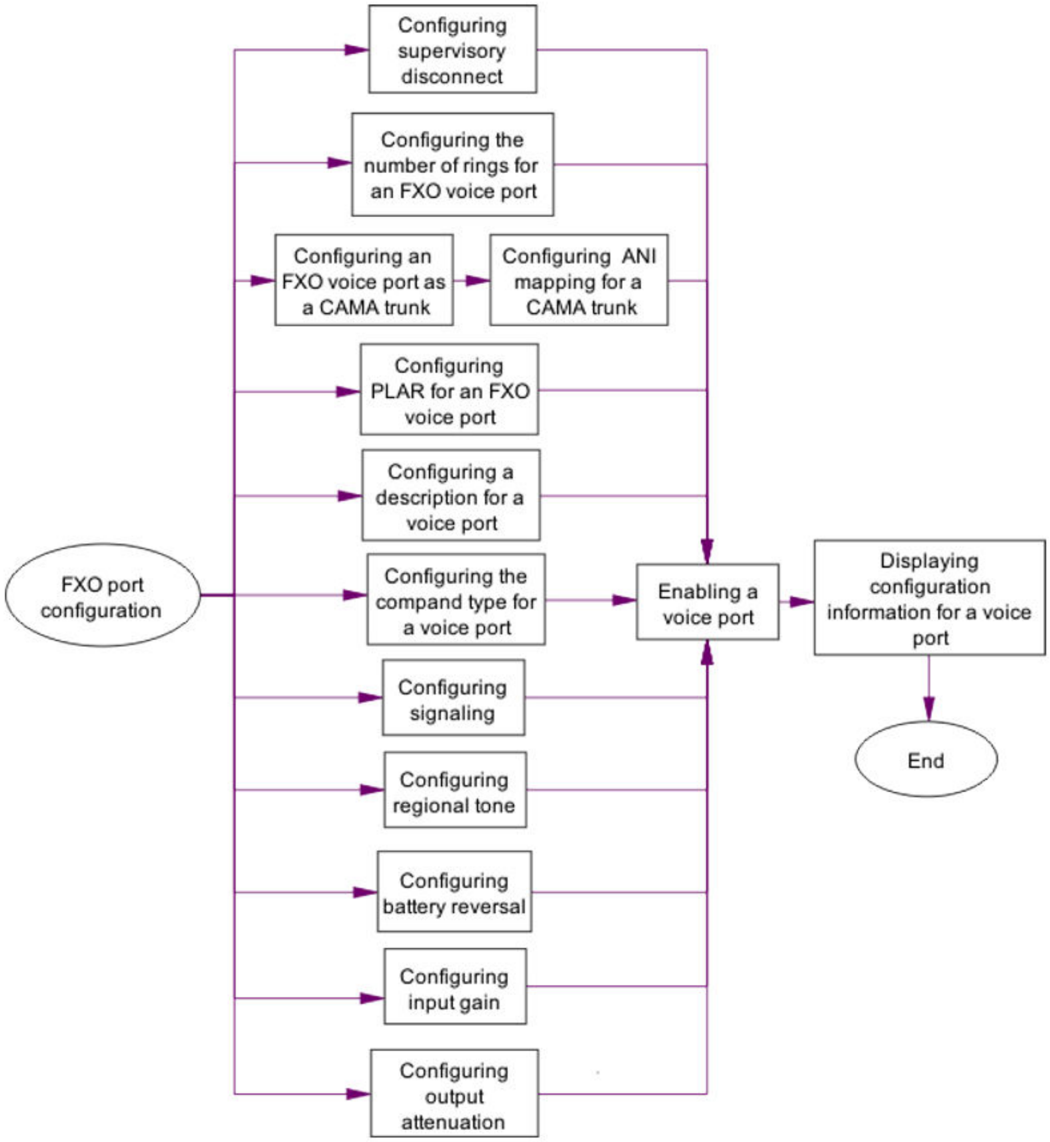


Figure 23: FXO port configuration procedures

Configuring the number of rings for an FXO voice port

Use this procedure to specify the number of rings for a specified FXO voice port.

By default, the value is 1.

Procedure steps

1. To enter configuration mode, enter:

```
configure terminal
```

2. To select a voice port, enter:

```
voice-port <slot/[<subslot/]port>
```

3. To specify the number of rings for the FXO port, enter:

```
[no] ring-number <ring-number>
```

Table 59: Variable definitions

Variable	Value
<ring-number>	Specifies the number of rings to detect before answering the call. Valid range is 1–10.
[no]	Configures the ring number to the default value: 1.

Configuring an FXO voice port as a CAMA trunk

Use this procedure to configure an FXO port for 911 calls.

By default, there is no CAMA configuration on any ports.

No two service areas in the existing North American telephony infrastructure that support E911 calls have identical service implementations. Many factors that drive the design of emergency call handling are matters of local policy and are therefore outside the scope of this document.

Local policy determines which Automatic Number Identification (ANI) format you can use for the specified Public Safety Answering Point (PSAP) location. The supported four types of ANI transmittal schemes are based on the actual number of digits transmitted toward the E911 tandem. In each instance, the actual calling number is proceeded with a key pulse (KP) followed by an information (I) field or a Numbering Plan Digit (NPD), which is then followed by the ANI calling number, and finally by a start pulse (ST), STP, ST2P, or ST3P, depending on the trunk group type in the PSTN and the traffic mix carried. The information field is one or two digits, depending on how the circuit was ordered originally. For one-digit information fields, a value of 0 indicates that the calling number is available. A value of 1 indicates that the calling number is not available. A value of 2 indicates an ANI failure.

For a complete list of values for two-digit information fields, see SR-2275: Telcordia Notes on the Networks at www.telcordia.com.

Procedure steps

1. To enter configuration mode, enter:

```
configure terminal
```

2. To select a voice port, enter:

```
voice-port <slot/[<subslot/]port>
```

3. To configure an FXO port for 911 calls, enter:

```
[no] signal cama {kp-0-nxx-xxxx-st | kp-0-npa-nxx-xxxx-st |
kp-2-st | kp-npd-nxx-xxxx-st}
```

Table 60: Variable definitions

Variable	Value
kp-0-nxx-xxxx-st	Specifies 7-digit ANI transmission. The calling phone number is transmitted, while the Numbering Plan Area (NPA) is implied by the trunk group and not transmitted.
kp–2–st	Specifies kp–2–st transmission, which is used if the PBX cannot out-pulse the ANI. If the ANI received by the SR2330/4134 is not as per configured values, kp–2–st is transmitted. For example, if the voice port is configured for out-pulsing a ten-digit ANI and the 911 call it receives has a seven-digit calling party number, the SR2330/4134 transmits kp–2–st.
kp-0-npa-nxx-xxxx-st	Specifies 10-digit transmission. The E.164 number is fully transmitted.
kp-npd-nxx-xxxx-st	Specifies 8-digit ANI transmission. The I field consists of a single-digit NPD-to-NPA mapping. The NPD table is preprogrammed in the sending and receiving equipment (on each end of the MF trunk); for example: 0= 415, 1=510, 2=650, 3=916 05551234 = (415) 555-1234, 15551234 = (510) 555-1234, and so on. When the calling party number of 415-555-0122 places a 911 call, and the SR2330/4134 has an NPD (0)-to-NPA (415) mapping, the NPA signaling format is received by the selective router at the central office (CO). NPD range is from 0 to 3.
[no]	Removes the configured 911 transmission selection.

Configuring ANI mapping for a CAMA trunk

Use this procedure to preprogram the Numbering Plan Area (NPA) or area code into a single multifrequency (MF) digit.

By default, no ANI mapping is configured.

Procedure steps

1. To enter configuration mode, enter:

   `configure terminal`

2. To select a voice port, enter:

   `voice-port <slot/[<subslot/]port>`

3. To preprogram the NPA or area code into a single MF digit, enter:

   `[no] ani mapping <npd-value> <npa-number>`

Table 61: Variable definitions

Variable	Value
<npd-value>	Specifies the value of the Numbering Plan Digit (NPD). Valid range is from 0 to 3.
<npa-number>	Specifies the number (area code) of the NPA. Valid range is from 100 to 999.
[no]	Disables Automatic Number Identification (ANI) mapping.

Configuring PLAR for an FXO voice port

The Private Line Automatic Ringdown (PLAR) autodialing mechanism permanently associates a local voice interface with a far-end voice interface, allowing call completion to a specific telephone number or PBX without dialing. When the calling telephone goes off-hook, a predefined network dial peer is automatically matched, which sets up a call to the destination telephone or PBX.

By default, PLAR is disabled.

By using the **connection plar** command, you can enhance your voice network to offer a number of useful features including the following:

- Providing an off-premises extension (OPX) from a PBX, thus simulating direct connections between FXS port users on a voice gateway and the PBX.
- Providing dial-tone from a remote PBX to offer toll-bypass functionality. Instead of relying on the gateways in your voice network to provide dial-tone, you can employ PLAR to enable remote sites to behave as though they have a direct connection to a PBX.
- Eliminating the need for user dialing, because both endpoints for the VoIP call are statically configured.

Connection PLAR behavior does not dedicate bandwidth to a call unless one or the other of the privately associated endpoints goes off-hook.

Take the following items into consideration when you plan to configure connection PLAR behavior on your VoIP network:

- Because connection PLAR is a switched VoIP call (similar to a switched virtual circuit), calls are set up and torn down as needed—bandwidth is taken up only when a call is initiated.
- Connection PLAR can operate between any type of signaling endpoint—FXO and FXS —and between any combination of analog and digital interfaces.
- Connection PLAR does not collect digits from the connected telephony device, so you can configure connection PLAR without any subsequent changes to your dial plan.
- You can enable connection PLAR on one or both of the statically configured endpoints, allowing you to use one-way or two-way connection PLAR.

Use the following procedure to configure a PLAR connection.

Procedure steps

1. To enter configuration mode, enter:

   ```
   configure terminal
   ```

2. To select a voice port, enter:

   ```
   voice-port <slot/[<subslot/]port
   ```

3. To configure PLAR, enter:

   ```
   [no] connection plar <dest-number>
   ```

Table 62: Variable definitions

Variable	Value
<dest-number>	Specifies the destination telephone number. Valid entries are any series of digits that specify the E.164 telephone number.
[no]	Removes the specified destination number.

FXO configuration example

The following is an example FXO port configuration.

```
voice-port 1/1
  signal loop-start
  ring-number 1
  caller-id enable 1
  station number 12345678
  no shutdown
  connection plar 1000
  exit voice-port
```

Chapter 11: FXS port configuration

Configure an FXS port to connect an analog telephone to the SR2330/4134.

In addition to the configuration details contained in this section, see also the following sections for related configuration:

- Caller ID configuration for FXS and FXO ports on page 155: to configure caller ID for the voice port.
- DSP configuration for all voice ports on page 161: to configure DSP related properties for the voice port.

FXS port configuration procedures

The following task flow shows you the sequence of procedures you perform to configure an FXS port.

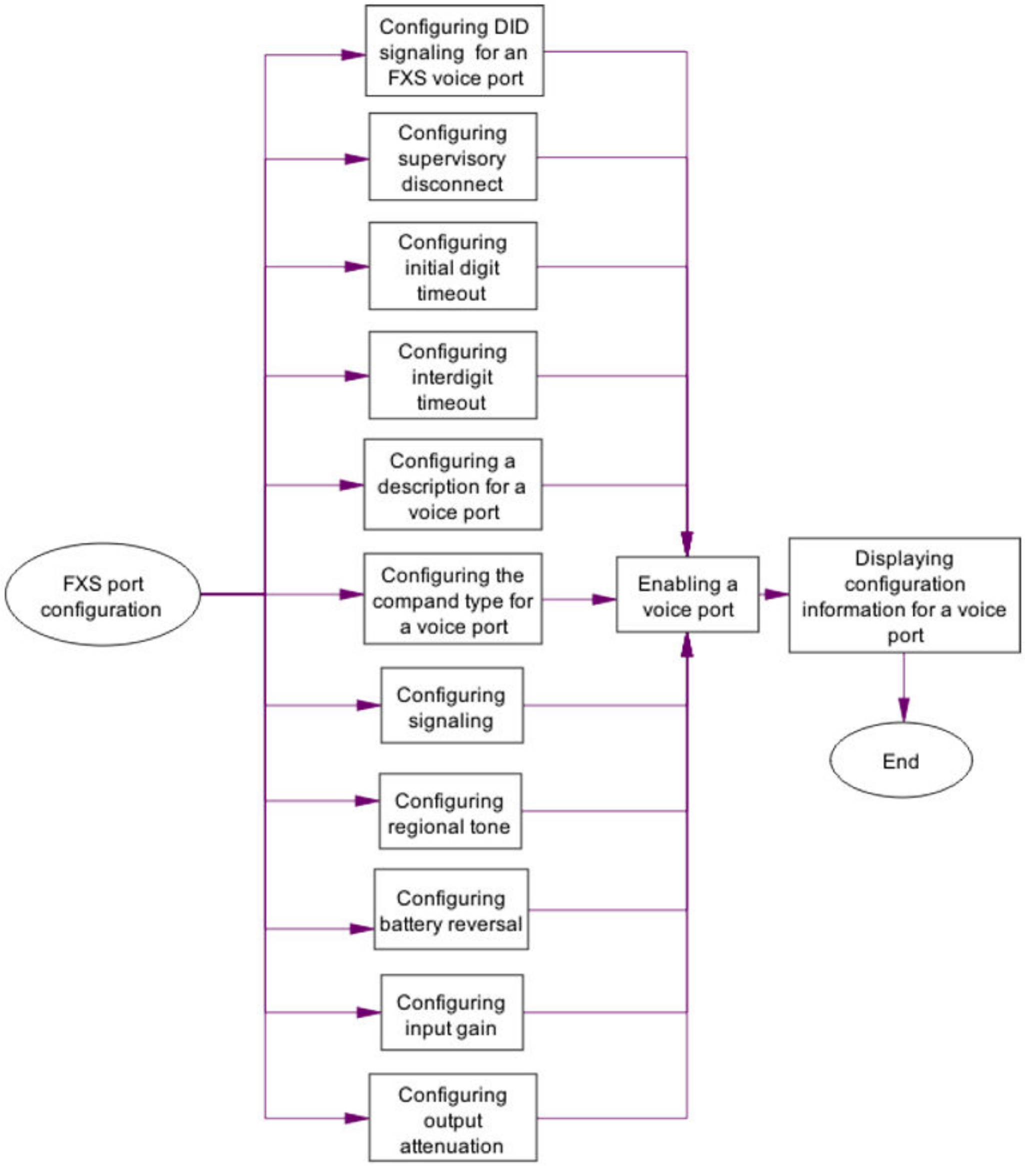


Figure 24: FXS port configuration procedures

Configuring DID signaling for an FXS voice port

Use this procedure to enable direct inward dialing (DID) on an FXS voice port. By default, DID is disabled and the voice port uses loop-start signaling.

Procedure steps

1. To enter configuration mode, enter:

   ```
   configure terminal
   ```

2. To select a voice port, enter:

   ```
   voice-port <slot/[<subslot/]port>
   ```

3. To enable DID on an FXS voice port, enter:

   ```
   [no] signal did {immediate-start | wink-start | delay-dial}
   ```

Table 63: Variable definitions

Variable	Value
immediate-start	Enables immediate-start signaling on the FXS DID voice port.
wink-start	Enables wink-start signaling on the FXS DID voice port.
delay-dial	Enables delay-dial signaling on the FXS DID voice port.
[no]	Disables DID and resets the port to loop-start signaling.

Configuring supervisory disconnect

Use this procedure to enable supervisory disconnect signaling on an FXS port.

Supervisory disconnect signaling is a power denial from the switch that lasts at least 350 ms. When the system detects this condition, it interprets this as a disconnect indication from the switch and clears the call.

When disconnect supervision is enabled in conjunction with answer supervision, disconnect supervision is detected by battery reversal.

Supervisory disconnect signaling is enabled by default.

Procedure steps

1. To enter configuration mode, enter:

   ```
   configure terminal
   ```

2. To select a voice port, enter:

   ```
   voice-port <slot/[<subslot/]port>
   ```

3. To enable a supervisory disconnect signal on the FXO port, enter:

```
[no] supervisory-disconnect
```

Table 64: Variable definitions

Variable	Value
[no]	Disables the supervisory disconnect signaling.

FXS configuration example

The following shows an FXS example configuration.

```
voice-port 1/1
  signal loop-start
  station name fxs1
  station number 31001
  no shutdown
  timeouts interdigit 3
  exit voice-port
```

Chapter 12: POTS dial peer configuration

Configure POTS dial peers to specify the connection parameters for calls made to and from POTS endpoints.

In addition to the configuration details contained in this section, see also the following sections for related configuration:

- SIP registration and authentication of FXS ports on page 141: to configure SIP registration and authentication of the POTS dial peer.
- Trunk group configuration on page 171: to add a dial peer to a trunk group.
- Number translation on page 167: to configure number translation for the dial peer.

POTS dial peer configuration procedures

The following task flow shows you the sequence of procedures you perform to configure a POTS dial peer.

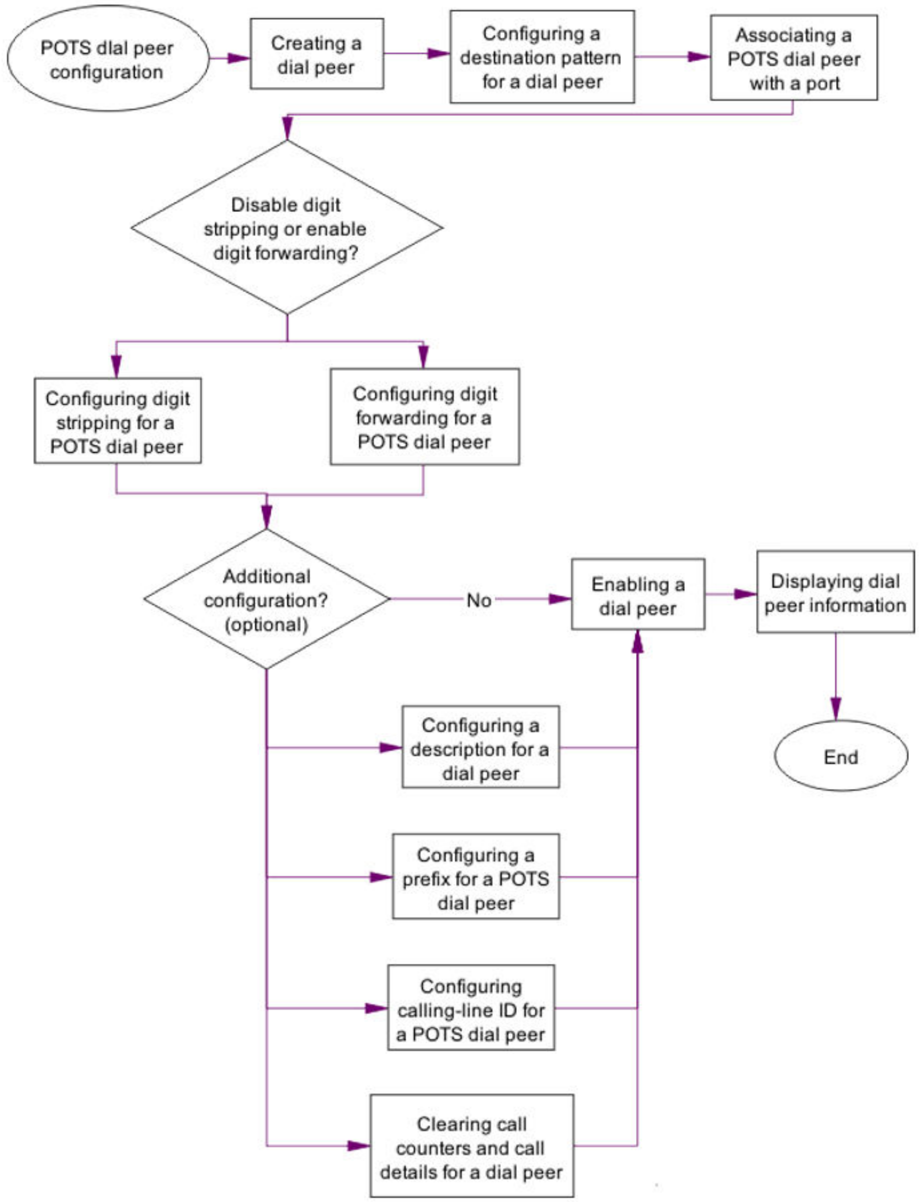


Figure 25: POTS dial peer configuration procedures

Associating a POTS dial peer with a port

Use this procedure to associate a POTS dial peer with a specific voice port.

The POTS dial peer is used for:

- calls that come from a voice port to select an incoming dial peer
- calls that come from the VoIP network to match a port with the selected outgoing dial peer

By default, no port is associated with the dial peer.

Procedure steps

1. To enter configuration mode, enter:

   ```
   configure terminal
   ```

2. To select a POTS dial peer, enter:

   ```
   dial-peer voice pots <tag>
   ```

3. To associate the POTS dial peer with a port, enter:

   ```
   [no] port <slot/[<subslot/]port>
   ```

Table 65: Variable definitions

Variable	Value
<slot/[<subslot/]port>	Port number of the voice card to configure.
[no]	Cancels the dial-peer association with the port.

Configuring a prefix for a POTS dial peer

Use this procedure to specify a prefix to add to the dialed digits of a dial peer.

When an outgoing call is initiated to this dial peer, the system sends the prefix string value to the telephony interface first, before the telephone number associated with the dial peer. By default, no string is specified.

If you want to configure different prefixes for dialed numbers on the same interface, you must configure different dial peers. This command applies only to plain old telephone service (POTS) dial peers.

Procedure steps

1. To enter configuration mode, enter:

   ```
   configure terminal
   ```

2. To select a POTS dial peer, enter:

   ```
   dial-peer voice pots <tag>
   ```

3. To specify the prefix of the dialed digits for the POTS dial peer, enter:

```
[no] prefix <prefix>
```

Table 66: Variable definitions

Variable	Value
<prefix>	Integers that represent the prefix of the telephone number associated with the specified dial peer. Valid range is a string of up to 10 digits.
[no]	Resets the prefix to the default value: no string is specified.

Configuring digit stripping for a POTS dial peer

Use this procedure to enable digit stripping on a POTS dial-peer call leg. To disable digit stripping on the dial-peer call leg, use the no form of this command.

By default, digit stripping is enabled.

This procedure applies to POTS dial peers only. When the system receives a called number and matches it to a POTS dial peer, it strips the matched digits and forwards the remaining digits to the voice interface.

Important:

As digit-stripping is enabled by default on the SR2330/4134, you must either disable digit stripping, or enable digit forwarding on the POTS dial peer to be able to connect incoming calls to the associated port.

Procedure steps

1. To enter configuration mode, enter:

   ```
   configure terminal
   ```

2. To select a POTS dial peer, enter:

   ```
   dial-peer voice pots <tag>
   ```

3. To enable digit stripping on a POTS dial peer call leg, enter:

   ```
   [no] digit-strip
   ```

Configuring digit forwarding for a POTS dial peer

Use this procedure to specify which digits to forward for voice calls.

By default, the SR2330/4134 forwards any dialed digits that do not match the destination pattern.

This procedure applies to POTS dial peers only. Forwarded digits are always right justified so that extra leading digits are stripped. The destination pattern includes both explicit digits and wildcards (if present). Use the default form of this command if a nondefault digit-forwarding scheme was entered previously and you wish to restore the default.

Important:

As digit-stripping is enabled by default on the SR2330/4134, you must either disable digit stripping, or enable digit forwarding on the POTS dial peer to be able to connect incoming calls to the associated port.

Procedure steps

1. To enter configuration mode, enter:

   `configure terminal`

2. To select a POTS dial peer, enter:

   `dial-peer voice pots <tag>`

3. To specify which digits to forward for voice calls on the POTS dial peer, enter:

   `[no] forward-digits {<num-digits> | all | extra | default}`

Table 67: Variable definitions

Variable	Value
<num-digits>	Specifies the number of digits to forward. If the number of digits is greater than the length of a destination phone number, the system uses the length of the destination number. Valid range is from 0 to 32. Setting the value to 0 is equivalent to entering the **`no forward-digits`** command.
all	Forwards all digits. If you enter all, the system uses the full length of the destination pattern.
extra	If the length of the dialed digit string is greater than the length of the dial-peer destination pattern, the system forwards the extra right-justified digits. However, if the dial-peer destination pattern is of variable length ending with T (for example, T, 123T, or 123...T), the system does not forward the extra digits.
default	Restores the default setting: the system forwards any dialed digits that do not match the destination pattern.

Variable	Value
[no]	Specifies that the system does not forward any digits that do not match the destination-pattern.

Examples of configuring digit forwarding for a POTS dial peer

In the following example, all digits in the destination pattern of a POTS dial peer are forwarded:

```
dial-peer voice pots 1
destination-pattern 8...
forward-digits all
```

In the following example, four of the digits in the destination pattern of a POTS dial peer are forwarded:

```
dial-peer voice pots 1
destination-pattern 555....
forward-digits 4
```

In the following example, the extra right-justified digits that exceed the length of the destination pattern of a POTS dial peer are forwarded:

```
dial-peer voice pots 1
destination-pattern 555....
forward-digits extra
```

Configuring calling-line ID for a POTS dial peer

Use this procedure to configure calling-line ID (CLID) information for a POTS dial peer.

The calling-line ID configured for POTS dial peers is applicable only for POTS-to-SIP calls. The configuration specifies the calling party number that is entered in the From field of the SIP Invite message.

If you configure a CLID network-number, the configured number appears in the outgoing SIP Invite message, replacing the incoming calling party number and name. If you choose the **`restrict`** option, calls from that dial peer are routed to the SIP network as anonymous (the calling party number is added under the P-Asserted-Identity Header in accordance with RFC 3325). If you configure a substitute display name, the configured display name is used only when the display name is unavailable from the incoming call.

You can configure the calling-line ID on POTS dial peers for any port type.

By default, no calling-line ID information is configured.

Procedure steps

1. To enter configuration mode, enter:

   ```
   configure terminal
   ```

2. To select a POTS dial peer, enter:

   ```
   dial-peer voice pots <tag>
   ```

3. To configure calling-line ID information for the POTS dial peer, enter:

   ```
   [no] clid {network-number <network-number> | restrict |
   substitute <name> }
   ```

Table 68: Variable definitions

Variable	Value
network-number <network-number>	Specifies the calling party network number in the CLID.
restrict	Configures caller ID restriction.
substitute <name>	Specifies a substitute display name when the display name is unavailable.
[no]	Removes the configured CLID information.

Chapter 13: SIP registration and authentication of FXS ports

Configure SIP registration and authentication of FXS ports to allow an FXS port to register and authenticate with a SIP server.

SIP registration and authentication procedures

The following task flow shows you the sequence of procedures you perform to configure the FXS port registration and authentication with a SIP server.

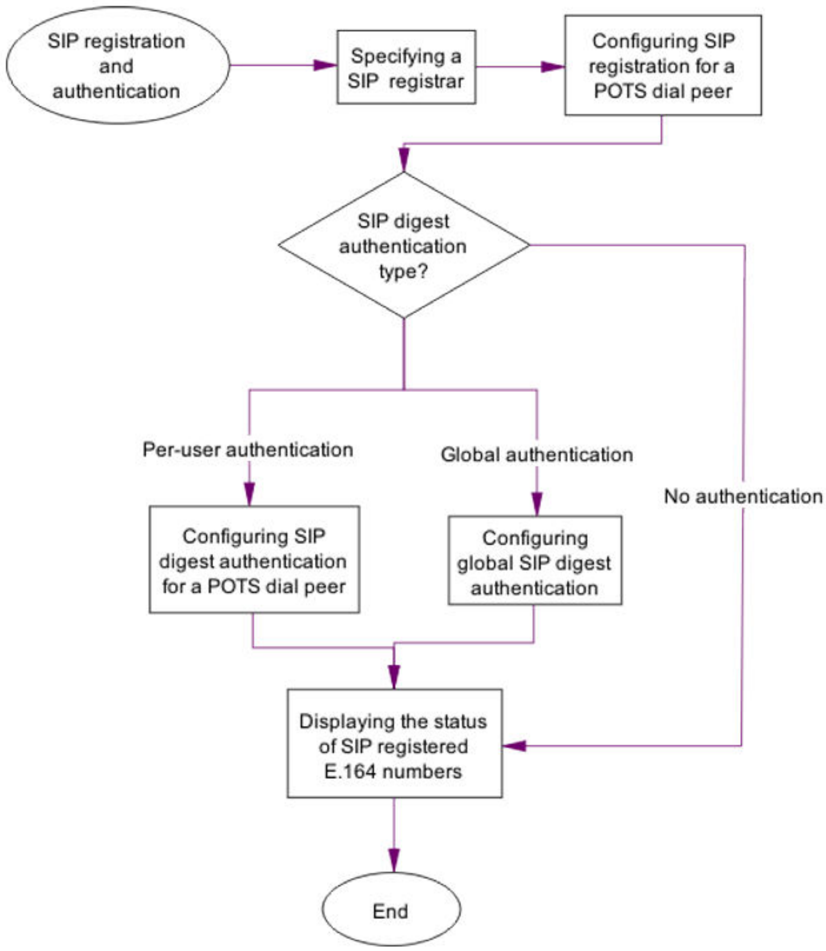


Figure 26: SIP registration and authentication procedures

Specifying a SIP registrar

Use this procedure to specify a SIP registrar server to which the SR2330/4134 can register E.164 numbers on behalf of FXS ports.

By default, registration is disabled.

The requirement to enable this registration is service-provider and SIP-server dependent.

Procedure steps

1. To enter configuration mode, enter:

   `configure terminal`

2. To select SIP UA configuration, enter:

```
sip-ua
```

3. To specify the SIP UA registrar, enter:

```
[no] registrar {dns:<host-name> | ipv4:<ip-addr>[:port-num]}
[expires <registration-time>]
```

Table 69: Variable definitions

Variable	Value
dns:<host-name>	Specifies the DNS host name of the SIP registrar server.
ipv4:<ip-addr>	Specifies the IP address of the SIP registrar server.
[:port-num]	Specifies the SIP registrar server port number to use (default is port 5060).
[expires <registration-time>]	Specifies the default registration time, in seconds. Valid range is from 60 to 65535 (default 3600).
[no]	Disables registration of E.164 numbers.

Configuring SIP registration for a POTS dial peer

Use this procedure to trigger SIP registration of the destination-pattern number of a POTS dial peer. This allows you to register the associated FXS ports with the SIP registrar.

By default, registration is disabled.

To enable registration, you must define the dial-peer destination pattern to match the number that is configured on the SIP server. If the destination pattern does not match the SIP server number, the dial peer cannot register.

In addition, to enable registration, the selected dial peer must be enabled.

To disable registration, use the no form of this command.

Procedure steps

1. To enter configuration mode, enter:

```
configure terminal
```

2. To select a POTS dial peer, enter:

```
dial-peer voice pots <tag>
```

3. To enable SIP registration for the POTS dial peer, enter:

```
[no] register e164
```

Configuring global SIP digest authentication

If the SIP server uses a global authentication value for all users, use this procedure to enter global SIP digest authentication information.

By default, no authentication is configured.

The SIP digest authenticates users against their public address, user ID, and password. When users are registered and authenticated, they can dial other registered users in the system. The requirement for SIP digest authentication is service-provider (and SIP-server) dependent.

Procedure steps

1. To enter configuration mode, enter:

   ```
   configure terminal
   ```

2. To select SIP UA configuration, enter:

   ```
   sip-ua
   ```

3. To configure SIP digest authentication, enter:

   ```
   [no] authentication <username> <password>
   ```

Table 70: Variable definitions

Variable	Value
<username>	Specifies the username of the user to authenticate. Maximum length is 32 characters.
<password>	Specifies the password for authentication. Maximum length is 32 characters.
[no]	Disables SIP digest authentication.

Configuring SIP digest authentication for a POTS dial peer

If the SIP server uses a unique authentication value for each user, use this procedure to enter the SIP digest authentication information for the POTS dial peer. This configuration overrides the global configuration of SIP digest authentication.

By default, no authentication is configured.

The SIP digest authenticates the user against the public address, user ID, and password. When users are registered and authenticated, they can dial other registered users in the system.

Procedure steps

1. To enter configuration mode, enter:

   ```
   configure terminal
   ```

2. To select a POTS dial peer, enter:

   ```
   dial-peer voice pots <tag>
   ```

3. To configure SIP digest authentication information, enter:

   ```
   [no] authentication <username> <password>
   ```

Table 71: Variable definitions

Variable	Value
<username>	Specifies the username of the user to authenticate. Maximum length is 32 characters.
<password>	Specifies a password for authentication. Maximum length is 32 characters.
[no]	Disables SIP digest authentication.

Displaying the status of SIP-registered E.164 numbers

Use this procedure to display the status of E.164 numbers that the SIP Media Gateway has registered with an external SIP registrar.

Procedure steps

To display the status of registered E.164 numbers, enter:

```
show sip-ua register status
```

Chapter 14: VoIP dial peer configuration

Configure the VoIP dial peer to specify the connection parameters of calls made to and from VoIP endpoints.

VoIP dial peer configuration procedures

The following task flow shows you the sequence of procedures you perform to configure a VoIP dial peer.

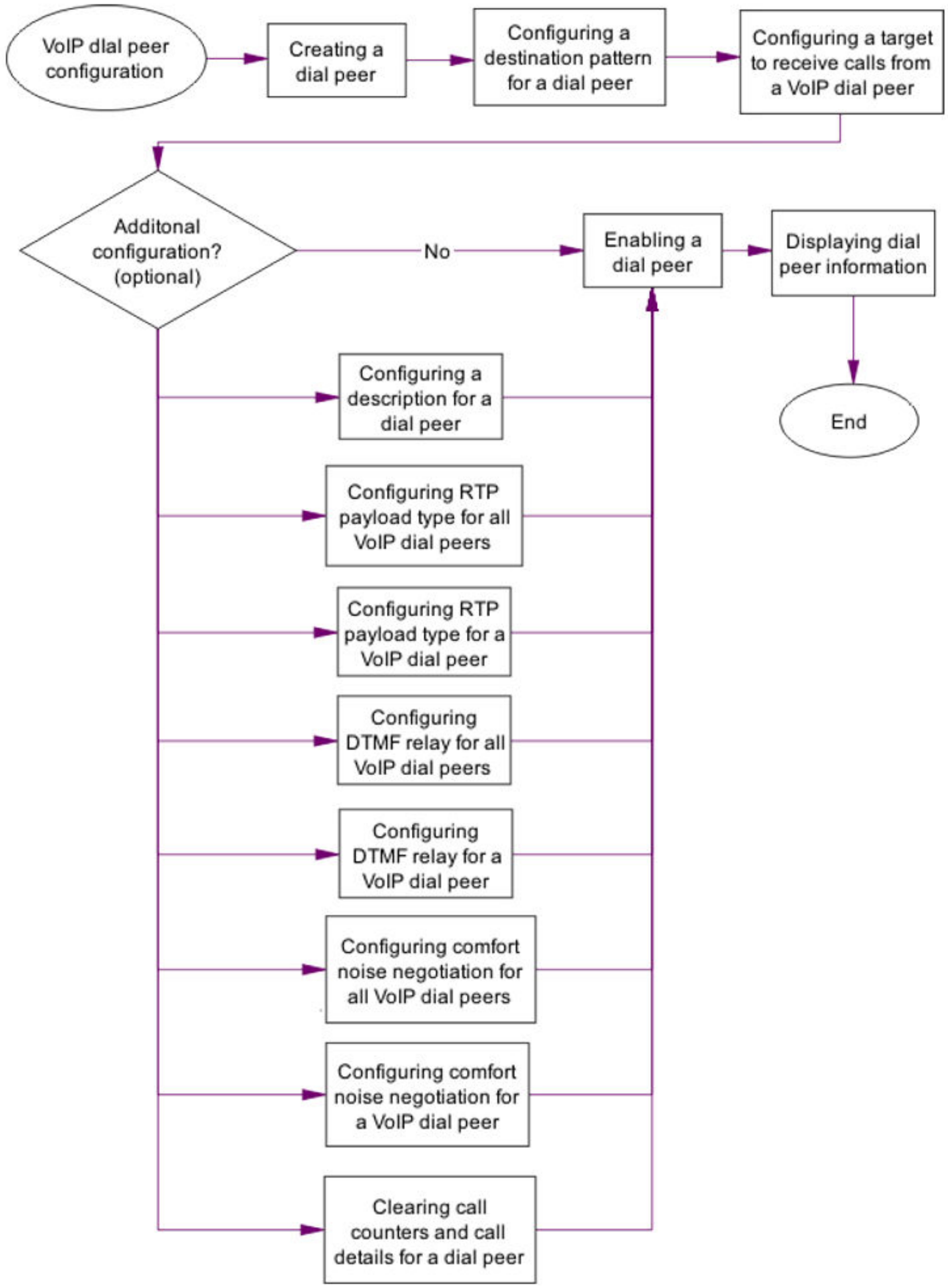


Figure 27: VoIP dial peer configuration procedures

Configuring a target to receive calls from a VoIP dial peer

Use this procedure to designate a network-specific address to receive calls from a VoIP dial peer. The session target is typically the SIP server.

By default, no session target is configured.

Procedure steps

1. To enter configuration mode, enter:

   `configure terminal`

2. To select a VoIP dial peer, enter:

   `dial-peer voice voip <tag>`

3. To configure a network target to receive calls from the VoIP dial peer, enter:

   `[no] session target {<ipv4-dest-address> | sip-server}`

Table 72: Variable definitions

Variable	Value
<ipv4-dest-address>	Specifies the destination IP address to receive calls.
sip-server	Specifies the global destination SIP server for calls from this dial peer. If you first define the SIP server IP address using the **`sip-ua sip-server`** command, you can then enter the **`session target sip-server`** option for each dial peer, rather than repeatedly entering the full SIP server IP address for each dial peer (using **`session target <ipv4-dest-address>`**).
[no]	Resets to the default setting: no session target defined.

Configuring RTP payload type for all VoIP dial peers

Use this procedure to identify the payload type of Real-Time Transport Protocol (RTP) packets for all VoIP dial peers.

By default, no RTP payload type is configured.

Procedure steps

1. To enter configuration mode, enter:

   `configure terminal`

2. To select VoIP service configuration, enter:

   `voice service voip`

3. To identify the payload type of RTP packets, enter:

   `[no] rtp payload-type nte <payload-type>`

Table 73: Variable definitions

Variable	Value
<payload-type>	A named telephone event (NTE). Valid range is from 96 to 127 (default 101).
[no]	Removes the configured RTP payload type.

Configuring RTP payload type for a VoIP dial peer

Use this procedure to identify the payload type of Real-Time Transport Protocol (RTP) packets for a specific VoIP dial peer.

By default, no RTP payload type is configured.

Procedure steps

1. To enter configuration mode, enter:

   `configure terminal`

2. To select a VoIP dial peer, enter:

   `dial-peer voice voip <tag>`

3. To identify the payload type of RTP packets, enter:

   `[no] rtp payload-type nte <payload-type>`

Table 74: Variable definitions

Variable	Value
nte <payload-type>	A named telephone event (NTE). Valid range for payload type is from 96 to 127 (default 101).

Variable	Value
[no]	Removes the RTP payload type.

Configuring DTMF relay for all VoIP dial peers

Use this procedure to specify how the SIP gateway relays dual tone multifrequency (DTMF) tones between telephony interfaces and an IP network.

By default, DTMF tones are disabled and sent in-band.

Procedure steps

1. To enter configuration mode, enter:

   `configure terminal`

2. To select VoIP service configuration, enter:

   `voice service voip`

3. To specify how the SIP gateway relays DTMF tones for all VoIP dial peers, enter:

   `[no] dtmf-relay rtp-nte`

Table 75: Variable definitions

Variable	Value
rtp-nte	Forwards DTMF tones using Real-Time Transport Protocol (RTP) with the Named Telephone Event (NTE) payload type.
[no]	DTMF tones are disabled and sent in-band.

Configuring DTMF relay for a VoIP dial peer

Use this procedure to specify how the SIP gateway relays DTMF tones between telephony interfaces and an IP network for a specific VoIP dial peer.

By default, DTMF tones are disabled and sent in-band.

Procedure steps

1. To enter configuration mode, enter:

   `configure terminal`

2. To select a VoIP dial peer, enter:

```
dial-peer voice voip <tag>
```

3. To specify how the SIP gateway relays DTMF tones for a specific VoIP dial peer, enter:

```
[no] dtmf-relay rtp-nte
```

Table 76: Variable definitions

Variable	Value
rtp-nte	Forwards DTMF tones by using Real-Time Transport Protocol (RTP) with the Named Telephone Event (NTE) payload type.
[no]	Removes all signaling options and sends the DTMF tones as part of the audio stream.

Configuring comfort noise negotiation for all VoIP dial peers

Use this procedure to configure comfort noise negotiation on all VoIP dial peers.

By default, comfort noise negotiation is enabled.

Procedure steps

1. To enter configuration mode, enter:

```
configure terminal
```

2. To select VoIP service configuration, enter:

```
voice service voip
```

3. To configure comfort noise negotiation, enter:

```
[no] comfort-noise-negotiate
```

Table 77: Variable definitions

Variable	Value
[no]	Disables comfort noise negotiation.

Configuring comfort noise negotiation for a VoIP dial peer

Use this procedure to configure comfort noise negotiation on a VoIP dial peer.

By default, comfort noise negotiation is enabled.

Procedure steps

1. To enter configuration mode, enter:

   `configure terminal`

2. To select a VoIP dial peer, enter:

   `dial-peer voice voip <tag>`

3. To configure comfort noise negotiation, enter:

   `[no] comfort-noise-negotiate`

Table 78: Variable definitions

Variable	Value
[no]	Disables comfort noise negotiation.

Chapter 15: Caller ID configuration for FXS and FXO ports

Configure caller ID to enable the receipt and transmission of caller ID information on FXS and FXO voice ports.

Caller ID configuration procedures

The following task flow shows you the sequence of procedures you perform to configure caller ID.

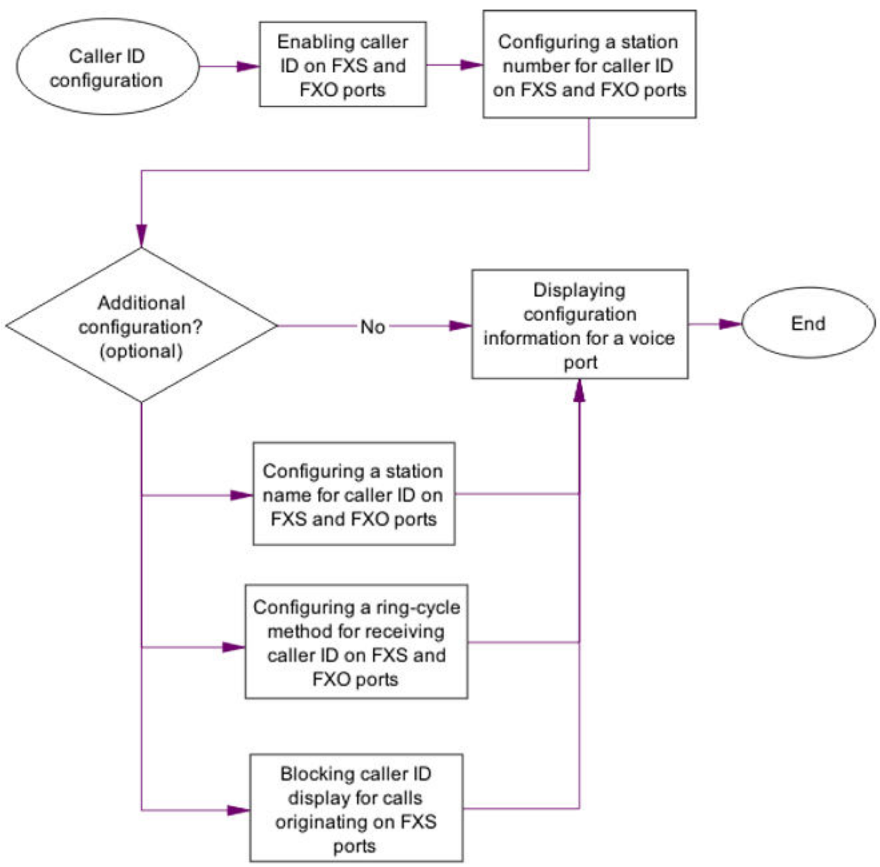


Figure 28: Caller ID configuration procedures

Enabling caller ID on FXS and FXO ports

Use this procedure to enable the system to send or receive caller ID information. Use this procedure on the sending FXS voice port or the receiving FXO voice port.

By default, caller ID is disabled.

Procedure steps

1. To enter configuration mode, enter:

   `configure terminal`

2. To select a voice port, enter:

   `voice-port <slot/[<subslot/]port>`

3. To allow the port to send or receive caller ID information, enter:

   `[no] caller-id enable [type 1]`

Table 79: Variable definitions

Variable	Value
[type 1]	Specifies the caller ID type: • 1: Type I transmits the signal when the receiving phone is on hook. Only Type I caller ID is supported.
[no]	Disables the sending and receiving of caller ID information.

Examples of enabling caller ID on FXS and FXO ports

In the following example, an FXS voice port is configured to send caller-ID information:

```
voice-port 7/2/1
station number 4082164655
caller-id enable
```

In the following example, an FXO voice port is configured to receive caller-ID information:

```
voice-port 7/3/1
caller-id enable
```

Configuring a station number for caller ID on FXS and FXO ports

Use this procedure to specify the telephone or extension number to send as caller ID information. Use this procedure on the sending FXS voice port or the FXO port through which routed caller ID calls pass.

By default, no station number is configured.

You can use this procedure on FXS voice ports that are used to originate on-net calls. The information that you enter displays on the telephone attached to the FXS port at the far end of the on-net call.

You can also use this procedure on FXO voice ports on which caller ID information is expected to be received from the Central Office (CO). This supports situations in which a call is placed from the CO to the FXO interface and continues to a far-end FXS port through an on-net call. If the FXO port receives no caller ID information from the CO, the FXO port forwards the configured station number to the far-end call recipient.

Procedure steps

1. To enter configuration mode, enter:

   ```
   configure terminal
   ```

2. To select a voice port, enter:

   ```
   voice-port <slot/[<subslot/]port>
   ```

3. To specify the number to send as caller ID information, enter:

   ```
   [no] station number <station-number>
   ```

Table 80: Variable definitions

Variable	Value
<station-number>	Specifies the station number. Valid value is a string of 1 to 15 characters.
[no]	Removes the configured station number.

Configuring a station name for caller ID on FXS and FXO ports

Use this procedure to specify the name to send as caller ID information.

By default, no station name is configured.

Procedure steps

1. To enter configuration mode, enter:

   `configure terminal`

2. To select a voice port, enter:

   `voice-port <slot/[<subslot/]port>`

3. To specify the station name to send in caller ID information, enter:

   `[no] station name <station-name>`

Table 81: Variable definitions

Variable	Value
<station-name>	Specifies the station name. Valid value is a string of 1 to 15 characters.
[no]	Removes the configured station name.

Configuring a ring-cycle method for receiving caller ID on FXS and FXO ports

Use this procedure to set the ring-cycle method for receiving caller ID information for on-hook (Type 1) caller ID at a receiving FXO or a sending FXS voice port.

This setting is determined by the Bellcore/Telcordia or ETSI standard that your telephone service provider uses for caller ID. Use it on FXO loop-start and ground-start voice ports where caller ID information arrives and on FXS voice ports that send caller ID information. This setting must match on the sending and receiving ends of the telephone line connection.

By default, caller ID alerting displays after the first ring at the receiving station.

Procedure steps

1. To enter configuration mode, enter:

```
configure terminal
```

2. To select a voice port, enter:

```
voice-port <slot/[<subslot/]port>
```

3. To set the ring-cycle method for the voice port, enter:

```
[no] caller-id alerting ring {1 | 2 }
```

Table 82: Variable definitions

Variable	Value
1	Specifies that caller ID alerting displays after the first ring at the receiving station. This is the most common setting.
2	Specifies that caller ID alerting displays after the second ring.
[no]	Sets the alerting ring to the default value: 1.

Blocking caller ID display for calls originating on FXS voice ports

Use this procedure to request the blocking of the display of caller ID information at the far end of a call for calls that originate from an FXS port. Use this procedure on the originating FXS voice port. To allow the display of caller ID information, use the no form of this command.

By default, caller ID display is not blocked.

Procedure steps

1. To enter configuration mode, enter:

```
configure terminal
```

2. To select a voice port, enter:

```
voice-port <slot/[<subslot/]port>
```

3. To request the blocking of ID display at the far end, enter:

```
[no] caller-id block
```

Chapter 16: DSP configuration for all voice ports

Configure digital signal processor (DSP) properties to specify the digital signal properties applied on voice ports.

DSP configuration procedures

The following task flow shows you the sequence of procedures you perform to configure DSP.

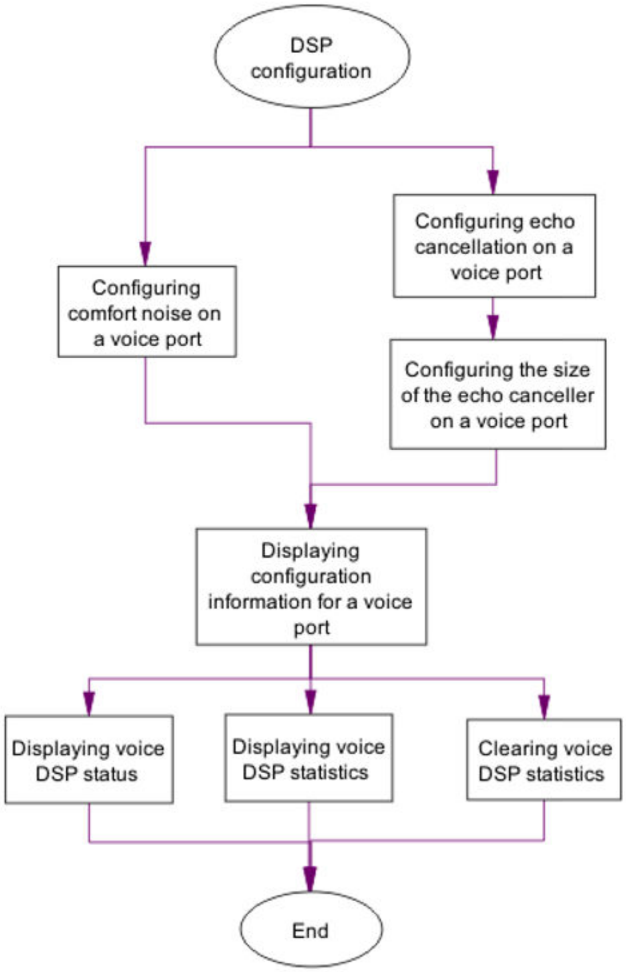


Figure 29: DSP configuration procedures

Configuring comfort noise on a voice port

Use this procedure to generate background noise to fill silent gaps during calls when voice activity detection (VAD) is activated. To provide silence when the remote party is not speaking and VAD is enabled at the remote end of the connection, use the no form of this command.

By default, no background noise is generated.

Procedure steps

1. To enter configuration mode, enter:

```
configure terminal
```

2. To select a voice port, enter:

```
voice-port <slot/[<subslot/]port[:<d-channel>]>
```

3. To generate background noise, enter:

```
[no] comfort-noise
```

Configuring echo cancellation on a voice port

Use this procedure to enable the cancellation of voice that is sent out and returned on the same interface. To disable echo cancellation, use the no form of this command.

By default, the G.168 echo canceller (EC) is enabled with the echo suppressor turned off.

Procedure steps

1. To enter configuration mode, enter:

```
configure terminal
```

2. To select a voice port, enter:

```
voice-port <slot/[<subslot/]port[:<d-channel>]>
```

3. To enable echo cancellation, enter:

```
[no] echo-cancel enable
```

Configuring the size of the echo canceller on a voice port

Use this procedure to adjust the size of the echo canceller (EC). If you enable echo-cancel, this information passes as a call parameter during media setup.

By default, the echo canceller is set to 64 ms.

Procedure steps

1. To enter configuration mode, enter:

```
configure terminal
```

2. To select a voice port, enter:

```
voice-port <slot/[<subslot/]port[:<d-channel>]>
```

3. To adjust the size of the echo canceller, enter:

```
[no] echo-cancel coverage {8 | 16 | 24 | 32 | 48 | 64 | 128}
```

Table 83: Variable definitions

Variable	Value
{8 | 16 | 24 | 32 | 48 | 64 | 128}	Specifies the echo canceller size in milliseconds (ms).
[no]	Resets this command to the default value (64 ms).

Displaying voice DSP status

Use this procedure to display the current status of DSP voice channels.

Procedure steps

To display the current status of DSP voice channels, enter:

```
show voice dsp status
```

Displaying voice DSP parameters

Use this procedure to display configured DSP related parameters.

Procedure steps

To display the configuration of DSP related parameters, enter:

```
show voice dsp configuration
```

Displaying voice DSP statistics

Use this procedure to display selective statistics of DSP voice channels.

Procedure steps

To display selective statistics of digital signal processor (DSP) voice channels, enter:

```
show voice dsp statistics
```

Clearing voice DSP statistics

Use this procedure to clear DSP statistics.

Procedure steps

To clear DSP statistics, enter:

```
clear voice dsp statistics
```

Chapter 17: Number translation

Number translation allows you to modify the telephone numbers that enter or leave the SIP Media Gateway. For example, you can add an area code to a number that must be routed to the PSTN, or remove an area code for a number that is routed to an internal company site. You can also use number translation to add or strip the plus (+) character that is sometimes used for international calls.

You can apply number translation to the incoming or outgoing call leg. As well, you can choose to modify the calling party number, the called party number, or both.

Number translation procedures

The following task flow shows you the sequence of procedures you perform to configure number translation.

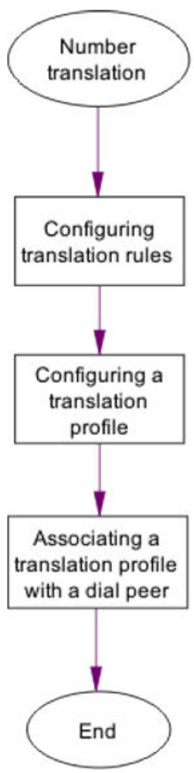


Figure 30: Number translation procedures

Configuring translation rules

Use this procedure to define translation rules for voice calls. You must first create a translation-rule group, and add translation rules to that group. Each translation-rule group can contain up to 10 different rules. The system supports a maximum of 15 translation-rule groups.

By default, no rules are configured.

Procedure steps

1. To enter configuration mode, enter:

   `configure terminal`

2. To specify a tag to identify the translation-rule group, enter:

   `voice translation-rule <trans-rule-number>`

3. To specify a translation rule, enter:

   `rule <precedence> /<match-pattern>/ /<replace-pattern>/`

Table 84: Variable definitions

Variable	Value
<trans-rule-number>	Number to identify the translation rule group. Valid range is from 1 to 2147483647.
<precedence>	Priority or precedence of the translation rule. Valid range is from 1 to 10.
/<match-pattern>/	Expression used to match incoming information. The replace-pattern can only contain digits, + (plus) and . (dot). + is only allowed at the beginning of the pattern.
/<replace-pattern>/	Expression used to replace the match pattern For the current release, the replace-pattern can only contain digits or +. + is only allowed at the beginning of the pattern.

Configuring a translation profile

Use this procedure to define a translation profile to assign to a dial peer. You can associate two translation-rule groups to each translation profile: one for calling numbers and one for called numbers. The system supports a maximum of 128 translation profiles.

By default, no translation profiles are configured.

Procedure steps

1. To enter configuration mode, enter:

```
configure terminal
```

2. To specify a name for the translation profile, enter:

```
voice translation-profile <profile-tag>
```

3. To associate a translation rule with a number, enter:

```
translate {calling|called} <trans-rule-number>
```

Table 85: Variable definitions

Variable	Value
<profile-tag>	Specifies the name of the translation profile. Maximum length of the voice translation profile is 31 alphanumeric characters.
{calling|called}	Specifies whether to associate the translation-rule group with the calling number or the called number.
<trans-rule-number>	Specifies the number of the translation rule to use in the translation profile.

Associating a translation profile with a dial peer

Use this procedure to associate a translation profile with a dial peer.

By default, the dial peer has no translation profile assigned.

Procedure steps

1. To enter configuration mode, enter:

```
configure terminal
```

2. To select a dial peer, enter:

```
dial-peer voice {pots | voip} <tag>
```

3. To associate a translation profile with a dial peer, enter:

```
translation-profile {incoming | outgoing} <profile-tag>
```

Table 86: Variable definitions

Variable	Value
{incoming | outgoing}	Specifies whether the translation profile is applied on incoming or outgoing calls.
<profile-tag>	Specifies the profile name to associate with the dial peer.

Chapter 18: Trunk group configuration

Configure trunk groups to associate a dial peer with multiple trunk ports. When you enable the trunk group, calls to the associated dial peer are connected to one of the free trunk ports in the group using round-robin selection.

To configure a trunk group, first create the trunk group and assign it a name. Then, assign each interface in the group to this trunk group. You can assign any T1 CAS, E1 R2, ISDN PRI, ISDN BRI, or FXO port to a trunk group. You cannot add FXS ports to the group.

Finally, you must assign a dial peer to the same trunk group. You cannot associate a dial peer to a voice port and a trunk group simultaneously.

The trunk group configuration only applies to POTS dial peers.

Trunk group configuration procedures

The following task flow shows you the sequence of procedures you perform to configure trunk groups.

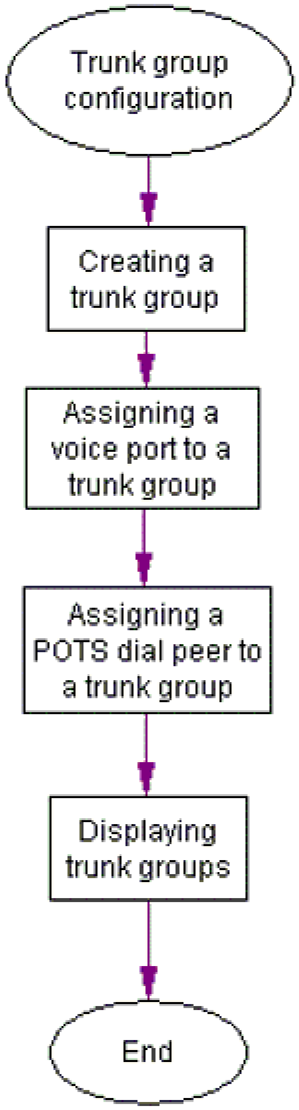


Figure 31: Trunk group configuration procedures

Creating a trunk group

Use this procedure to create a trunk group and configure it with a name.

By default, no trunk group is configured.

Procedure steps

1. To enter configuration mode, enter:

   `configure terminal`

2. To create a trunk group, enter:

   `[no] trunk group <trunk-group-name>`

Table 87: Variable definitions

Variable	Value
<trunk-group-name>	Specifies the name of the trunk group. Maximum length is 16 alphanumeric characters.
[no]	Deletes the specified trunk group.

Assigning a voice port to a trunk group

Use this procedure to assign an analog voice port to a trunk group.

Multiple voice ports can belong to the same trunk group. However, each voice port can belong to only one trunk group.

Procedure steps

1. To enter configuration mode, enter:

   `configure terminal`

2. To select a voice port, enter:

   `voice-port <slot/[<subslot/]port[:<d-channel>]>`

3. To assign an analog voice port to a trunk group, enter:

   `[no] trunk-group <trunk-group-name>`

Table 88: Variable definitions

Variable	Value
<trunk-group-name>	Specifies the name of the trunk group to which the port is assigned. Maximum length of the trunk group name is 16 alphanumeric characters.
[no]	Removes the port from the specified trunk group.

Assigning a POTS dial peer to a trunk group

Use this procedure to assign a POTS dial peer to a trunk group for trunk group label routing.

You cannot associate a POTS dial peer with a port and a trunk group simultaneously.

Procedure steps

1. To enter configuration mode, enter:

   ```
   configure terminal
   ```

2. To select a POTS dial peer, enter:

   ```
   dial-peer voice pots <tag>
   ```

3. To assign a dial peer to a trunk group , enter:

   ```
   [no] trunkgroup <trunk-group-name>
   ```

Table 89: Variable definitions

Variable	Value
<trunk-group-name>	Specifies the name of the trunk group. Maximum length is 16 alphanumeric characters.
[no]	Deletes the dial peer from the trunk group.

Displaying trunk groups

Use this procedure to display the configured trunk groups. This command displays the various trunk groups configured on the system and corresponding voice ports.

Procedure steps

To display the configured trunk groups, enter:

```
show trunk group <trunk-group-name>
```

Table 90: Variable definitions

Variable	Value
<trunk-group-name>	Displays information about the specific trunk group.

Trunk group configuration example

The following shows a trunk group example configuration.

Procedure steps

1. Create a trunk group with name ToPSTN:

   `trunk group ToPSTN`

2. Assign FXO voice ports to this trunk group:

   `voice-port 3/1`

   `trunk-group ToPSTN`

   `voice-port 3/2`

   `trunk-group ToPSTN`

3. Assign a POTS dial peer to this trunk group:

   `dial-peer voice pots 100`

   `trunkgroup ToPSTN`

4. Display the trunk group to verify the configuration:

   `show trunk group`

   ```
   Trunk group : ToPSTN
       trunk group label : ToPSTN
       ports in trunk : 3/1 3/2
   ```

Chapter 19: VoIP fax and modem configuration

Configure VoIP fax and modem properties to allow fax and modem connections to traverse the VoIP network.

Two conceptual methods exist to carry virtual real-time fax-machine-to-fax-machine communication across packet networks:

- T.38 Fax relay: Demodulates the T.30 fax from the PSTN at the sending gateway. The demodulated fax content is enveloped into a T.38 data stream, sent over the network, and remodulated into T.30 fax at the receiving end.
- Fax pass-through: Passes the modulated fax information from the PSTN in-band end-to-end over a voice speech path in an IP network. The voice speech path need to be configured with the G.711 codec. If not, the Media Gateway dynamically changes the configured voice codec to G.711 with no voice activity detection (VAD) and no echo cancellation (EC) for the duration of the fax session.

VoIP fax and modem configuration procedures

The following task flow shows you the sequence of procedures you perform to configure VoIP fax and modem.

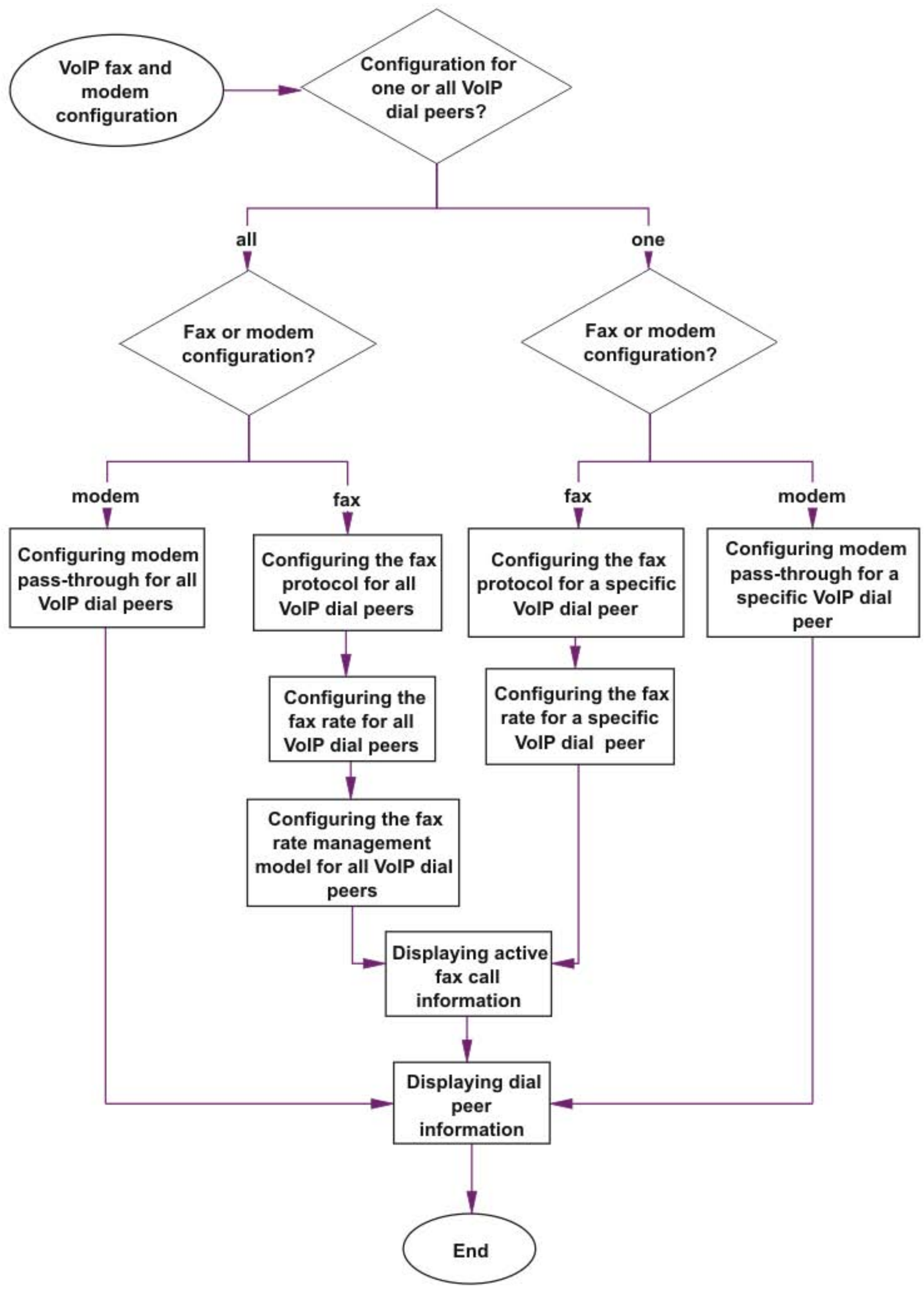


Figure 32: VoIP fax and modem configuration procedures

Configuring modem pass-through for all VoIP dial peers

Use this procedure to enable modem pass-through over VoIP for all dial peers.

By default, modem pass-through is disabled.

Procedure steps

1. To enter configuration mode, enter:

   `configure terminal`

2. To select VoIP service configuration, enter:

   `voice service voip`

3. To enable modem pass-through over VoIP for all dial peers, enter:

   `[no] modem passthrough {g711ulaw | g711alaw}`

Table 91: Variable definitions

Variable	Value
passthrough {g711ulaw | g711alaw}	Specifies which high-bandwidth codec the modem stream uses: • g711ulaw: uses the G.711 u-law codec. • g711alaw: uses the G.711 a-law codec.
[no]	Disables modem pass-through.

Configuring the fax protocol for all VoIP dial peers

Use this procedure to specify the default fax protocol to use for all VoIP dial peers.

By default, no fax protocol is configured.

If you use this command to configure fax relay options for all dial peers and you use the **`fax protocol`** command on a specific dial peer, the dial-peer configuration takes precedence over the configuration.

Procedure steps

1. To enter configuration mode, enter:

   `configure terminal`

2. To select VoIP service configuration, enter:

   `voice service voip`

3. To configure the fax protocol for all VoIP dial peers, enter:

   `[no] fax protocol {none |`

   `pass-through {g711ulaw | g711alaw} |`

```
t38 [fallback {none | pass-through-g711ulaw | pass-through-
g711alaw }] }
```

Table 92: Variable definitions

Variable	Value
none	Specifies that no fax pass-through is attempted. All special fax handling is disabled.
pass-through {g711ulaw | g711alaw}	Specifies which high-bandwidth codec the fax stream uses: • g711ulaw: uses the G.711 u-law codec. • g711alaw: uses the G.711 a-law codec.
t38	Specifies to use the ITU-T T.38 standard fax protocol for all VoIP dial peers.
[fallback {none | pass-through-g711ulaw | pass-through-g711alaw }]	With T.38, specifies the fallback mode to use to transfer a fax across a VoIP network if the T.38 fax relay cannot be successfully negotiated at the time of the fax transfer: • none: no fax pass-through is attempted. • pass-through-g711ulaw: uses the G.711 u-law codec. • pass-through-g711alaw: uses the G.711 a-law codec.
[no]	Resets to the default fax protocol value: none.

Configuring the fax rate for all VoIP dial peers

Use this procedure to establish the fax rate.

The default fax rate is 9600.

Procedure steps

1. To enter configuration mode, enter:

   ```
   configure terminal
   ```

2. To select VoIP service configuration, enter:

   ```
   voice service voip
   ```

3. To configure the fax rate for all VoIP dial peers, enter:

```
[no] fax rate {2400 | 4800 | 7200 | 9600 | 12000 | 14400|
voice}
```

Table 93: Variable definitions

Variable	Value
{2400 | 4800 | 7200 | 9600 | 12000 | 14400| voice}	Specifies the baud rate.
[no]	Resets the fax rate to the default value: 9600.

Configuring the fax rate management model

Use this procedure to configure the fax rate management model as defined in the ITU-T T.38 Recommendation. Values can be "localTCF" or "transferredTCF".

Procedure steps

1. To enter configuration mode, enter:

   ```
   configure terminal
   ```

2. To select VoIP service configuration, enter:

   ```
   voice service voip
   ```

3. To configure the fax rate management model, enter:

   ```
   fax rate-management <rate-management>
   ```

Table 94: Variable definitions

Variable	Value
<rate-management>	Valid values are localTCF or transferredTCF. Default is transferredTCF.

Configuring the fax protocol for a specific VoIP dial peer

Use this procedure to specify the fax protocol to use for a specific VoIP dial peer.

By default, the VoIP dial peer inherits the global fax protocol configuration. When you use this procedure to configure the fax protocol for a specific dial peer, this configuration takes precedence over the global configuration.

Procedure steps

1. To enter configuration mode, enter:

   `configure terminal`

2. To select a VoIP dial peer, enter:

   `dial-peer voice voip <tag>`

3. To configure the fax protocol for the specified VoIP dial peer, enter:

   `[no] fax protocol { none | system |`

   `pass-through {g711ulaw | g711alaw} |`

   `t38 [fallback {none | pass-through-g711ulaw | pass-through-g711alaw }] }`

Table 95: Variable definitions

Variable	Value
none	Specifies that no fax pass-through or T.38 fax relay is attempted. All special fax handling is disabled.
system	Uses the global configuration assigned using the **`configure voice service voip fax protocol`** command.
pass-through {g711ulaw | g711alaw}	Specifies that the fax stream uses one of the following high-bandwidth codecs: • g711ulaw: uses the G.711 u-law codec. • g711alaw: uses the G.711 a-law codec.
t38	Specifies to use ITU-T T.38 standard fax protocol for all VoIP dial peers.
[fallback {none | pass-through-g711ulaw | pass-through-g711alaw }]	With T.38, specifies the fallback mode to use to transfer a fax across a VoIP network if the T.38 fax relay cannot be successfully negotiated at the time of the fax transfer: • none: no fax pass-through is attempted. • pass-through-g711ulaw: uses the G.711 u-law codec. • pass-through-g711alaw: uses the G.711 a-law codec.
[no]	Resets to the globally defined fax protocol.

Configuring the fax rate for a specific VoIP dial peer

By default, the VoIP dial peer inherits the global fax rate configuration. When you use this procedure to configure the fax rate for a specific dial peer, this configuration takes precedence over the global configuration.

Procedure steps

1. To enter configuration mode, enter:

   ```
   configure terminal
   ```

2. To select a VoIP dial peer, enter:

   ```
   dial-peer voice voip <tag>
   ```

3. To configure the fax rate for a specific VoIP dial peer, enter:

   ```
   [no] fax rate {2400 | 4800 | 7200 | 9600 | 12000 | 14400|
   voice}
   ```

Table 96: Variable definitions

Variable	Value
{2400 | 4800 | 7200 | 9600 | 12000 | 14400| voice}	Specifies the baud rate.
[no]	Resets the fax rate to the globally defined value.

Configuring modem pass-through for a specific VoIP dial peer

Use this procedure to enable modem pass-through over VoIP for a specific dial peer.

This procedure can also be used to override the globally configured modem passthrough command.

Procedure steps

1. To enter configuration mode, enter:

   ```
   configure terminal
   ```

2. To select a VoIP dial peer, enter:

```
dial-peer voice voip <tag>
```

3. To enable modem pass-through over VoIP for the specified dial peer, enter:

```
[no] modem passthrough {codec {g711ulaw | g711alaw } |
system}
```

Table 97: Variable definitions

Variable	Value
codec {g711ulaw | g711alaw }	Specifies which high-bandwidth codec the modem stream uses: • g711ulaw: uses the G.711 u-law codec. • g711alaw: uses the G.711 a-law codec.
system	Uses the global configuration assigned using the global **modem passthrough** command.
[no]	Resets to the globally defined modem pass-through codec.

Displaying active fax call information

Use this procedure to display call information for fax calls in progress.

Procedure steps

1. To display call information for T38 fax calls in progress, enter:

```
show call active fax t38
```

2. To display call information for fax passthrough calls in progress, enter:

```
show call active voice
```

Chapter 20: Common procedures

The following sections describe common procedures that you use while configuring the SIP Media Gateway. These procedures are referenced in the previous configuration sections where applicable.

Common procedures for all voice ports:

- Selecting a voice port to configure on page 186
- Configuring a description for a voice port on page 186
- Configuring the compand type for a voice port on page 187
- Enabling a voice port on page 188
- Displaying configuration information for a voice port on page 188

Common procedures for FXS and FXO ports:

- Configuring signaling on page 189
- Configuring regional tone on page 189
- Configuring battery reversal on page 190
- Configuring input gain on page 191
- Configuring output attenuation on page 191

Common procedure for FXS and T1 CAS ports:

- Configuring initial digit timeout on page 192
- Configuring interdigit timeout on page 193

Common procedures for T1 CAS and ISDN ports:

- Configuring the network clock on page 194
- Displaying the network clock configuration on page 194

Custom tone configuration for T1/E1 PRI and CAS

- Defining dualtones for a specific country on page 195
- Creating custom tone classes on page 196
- Applying the custom tone to the voice port on page 198

Common procedures for POTS and VoIP dial peers:

- Creating a dial peer on page 199
- Configuring a destination pattern for a dial peer on page 199
- Configuring a description for a dial peer on page 201
- Enabling a dial peer on page 201

- Displaying dial peer information on page 202
- Clearing call counters and call details for a dial peer on page 202

Common procedures for all voice ports

The following sections describe the common procedures applicable to all voice ports.

Selecting a voice port to configure

Use this procedure to configure the voice port on a given card.

This command also provides access to the voice-port subtree.

Procedure steps

1. To enter configuration mode, enter:

   `configure terminal`

2. To select a voice port, enter:

   `voice-port <slot/[<subslot/]port[:<d-channel>]>`

Table 98: Variable definitions

Variable	Value
slot/[subslot/]port	Specifies the port number of the voice card to configure.
[:<d-channel>]	This option is valid only when a T1 port is configured for ISDN PRI. To configure ISDN PRI port properties, you must specify the D-channel timeslot. On a T1 port, <d-channel> = 23. On an E1 port, <d-channel> = 15.

Configuring a description for a voice port

Use this procedure to attach a text string description to the voice port. This description appears in various displays and you can use it to track the purpose or use of the voice port.

By default, no description is configured.

Procedure steps

1. To enter configuration mode, enter:

```
configure terminal
```

2. To select a voice port, enter:

```
voice-port <slot/[<subslot/]port[:<d-channel>]>
```

3. To provide a description for the voice port, enter:

```
[no] description "<description>"
```

Table 99: Variable definitions

Variable	Value
"<description>"	Specifies a description for the voice port. Valid value is a character string from 1 to 25 characters enclosed in quotation marks (" ").
[no]	Removes the voice-port description.

Configuring the compand type for a voice port

Use this procedure to specify the companding standard used to convert between analog and digital signals in pulse code modulation (PCM) systems.

The default value is u-law.

Procedure steps

1. To enter configuration mode, enter:

```
configure terminal
```

2. To select a voice port, enter:

```
voice-port <slot/[<subslot/]port[:<d-channel>]>
```

3. To specify the companding standard, enter:

```
[no] compand-type {g711ulaw | g711alaw}
```

Table 100: Variable definitions

Variable	Value
g711ulaw	Specifies the North American u-law ITU-T PCM encoding standard.
g711alaw	Specifies the European a-law ITU-T PCM encoding standard.
[no]	Resets to the default compand type value (u-law).

Enabling a voice port

Use this procedure to enable and disable specific voice ports. To put the ports in service, use the no form of this command. To take the ports offline, enter **shutdown** only.

By default, voice ports are disabled.

Procedure steps

1. To enter configuration mode, enter:
   `configure terminal`
2. To select a voice port, enter:
   `voice-port <slot/[<subslot/]port[:<d-channel>]>`
3. To enable or disable the port, enter:
   `[no] shutdown`

Displaying configuration information for a voice port

Use this procedure to display configuration information about a specific voice port. There are many CLI commands under the voice-port tree. This command displays information about all of these configured parameters.

Procedure steps

To display configuration information for a voice port, enter:

```
show voice port {<slot/[<subslot/]port[:<d-channel>]> |
[summary]}
```

Table 101: Variable definitions

Variable	Value
<slot/[<subslot/]port[:<d-channel>]>	Port number of the voice card to display. The <d-channel> value is valid only when ISDN PRI is configured on an E1/T1 port (<d-channel> = 23 for T1, 15 for E1).
summary	Displays a summary of all ports.

Common procedures for FXS and FXO ports

The following sections describe the common procedures applicable to FXS and FXO ports.

Configuring signaling

Use this procedure to specify loop-start or ground-start signaling for an FXS or FXO voice port.

By default, the FXS and FXO ports are configured for loop-start signaling.

Procedure steps

1. To enter configuration mode, enter:

   `configure terminal`

2. To select a voice port, enter:

   `voice-port <slot/[<subslot/]port>`

3. To specify the type of signaling for the voice port, enter:

   `[no] signal {loop-start | ground-start }`

Table 102: Variable definitions

Variable	Value
loop-start	Specifies the use of loop start signaling.
ground-start	Specifies the use of ground start signaling.
[no]	Reverts to the default signaling value: loop-start.

Configuring regional tone

Use this procedure to specify a regional analog voice-interface-related tone, ring, and cadence setting for an FXS or FXO voice port.

By default, regional tone is set to USA.

Procedure steps

1. To enter configuration mode, enter:

```
configure terminal
```

2. To select a voice port, enter:

```
voice-port <slot/[<subslot/]port>
```

3. To specify a regional tone, enter:

```
[no] cptone {us | ca | br}
```

Table 103: Variable definitions

Variable	Value
us	Specifies a region tone of USA.
ca	Specifies a region tone of Canada.
br	Specifies a region tone of Brazil.
[no]	Reverts the region tone value to the default: us.

Configuring battery reversal

Use this procedure to configure battery polarity reversal on an FXS or FXO port.

By default, battery reversal is enabled.

FXS ports normally reverse battery upon call connection. If an FXS port is connected to an FXO port that does not support battery reversal detection, you can disable battery reversal (using the **`no battery-reversal`** command) on the FXS port to prevent unexpected behavior.

FXO ports in loop-start mode normally disconnect calls when they detect a second battery reversal (back to normal). You can use the **`no battery-reversal`** command on FXO ports to disable this action.

If an FXO voice port in loop-start mode is connected to the PSTN and supports battery reversal, you can configure answer supervision using the **`battery-reversal answer`** command. This configures the FXO voice port to detect when a call is answered to provide correct billing information.

The answer option is supported on FXS DID ports as well.

Procedure steps

1. To enter configuration mode, enter:

```
configure terminal
```

2. To select a voice port, enter:

```
voice-port <slot/[<subslot/]port>
```

3. To configure battery polarity reversal on the port, enter:

```
[no] battery-reversal [answer]
```

Table 104: Variable definitions

Variable	Value
[answer]	Configures an FXO port to support answer supervision by detection of battery reversal.
[no]	Disables battery reversal.

Configuring input gain

Use this procedure to configure a specific input gain value or enable automatic gain control.

By default, the input gain value is 0 decibels.

Procedure steps

1. To enter configuration mode, enter:

```
configure terminal
```

2. To select a voice port, enter:

```
voice-port <slot/[<subslot/]port
```

3. To configure an input gain value, enter:

```
[no] input gain <decibels>
```

Table 105: Variable definitions

Variable	Value
<decibels>	Specifies the gain in decibels (dB) to insert at the receiver side of the interface. Valid values are -6, 0, or 6. The default is 0.
[no]	Resets to the default input gain value (0).

Configuring output attenuation

Use this procedure to configure a specific output attenuation value.

The default attenuation value is 0 decibels.

Procedure steps

1. To enter configuration mode, enter:

   `configure terminal`

2. To select a voice port, enter:

   `voice-port <slot/[<subslot/]port`

3. To configure an output attenuation value, enter:

   `[no] output attenuation <decibels>`

Table 106: Variable definitions

Variable	Value
<decibels>	Specifies the attenuation in decibels (dB) at the transmit side of the interface. Valid values are –6 or 0. The default is 0.
[no]	Resets to the default output attenuation value (0).

Common procedures for FXS, T1 CAS, and E1 R2

The following sections describe the common procedures applicable to FXS, T1 CAS, and E1 R2 ports.

Configuring initial digit timeout

Use this procedure to configure the initial digit timeout value for an FXS, T1 CAS, or E1 R2 port.

By default, initial digit timeout is set to 10 seconds.

To disable the initial digit timer, configure the timeout value to 0.

Procedure steps

1. To enter configuration mode, enter:

   `configure terminal`

2. To select a voice port, enter:

   `voice-port <slot/[<subslot/]port>`

3. To configure the initial timeout value, enter:

```
[no] timeouts initial <timeout>
```

Table 107: Variable definitions

Variable	Value
<timeout>	Specifies the number of seconds the system waits for the caller to input the first dialed digit. Valid range is from 0 to 120.
[no]	Resets to the default initial timeout value: 10 seconds.

Configuring interdigit timeout

Use this procedure to configure the interdigit timeout value for a specified FXS, T1 CAS, or E1 R2 voice port.

By default, the interdigit timeout value is 10 seconds.

The interdigit timeout specifies the number of seconds (after the caller inputs a digit) that the system waits for the caller to input a subsequent digit. The timeouts interdigit timer is activated when the caller inputs the initial digit and restarts each time the caller inputs another digit until the system identifies the destination address. If the caller exceeds the configured timeout value before the system identifies the destination address, the caller is notified through the appropriate tone and the call is terminated.

To disable the interdigit timer, set the timeout value to 0.

Procedure steps

1. To enter configuration mode, enter:

   ```
   configure terminal
   ```

2. To select a voice port, enter:

   ```
   voice-port <slot/[<subslot/]port>
   ```

3. To configure the interdigit timeout value, enter:

   ```
   [no] timeouts interdigit <timeout>
   ```

Table 108: Variable definitions

Variable	Value
<timeout>	Specifies the interdigit timeout value in seconds from 0 to 120.
[no]	Resets to the default interdigit timeout value: 10 seconds.

Common procedures for T1 CAS, E1 R2, and ISDN ports

The following sections describe the common procedures applicable to T1 CAS, E1 R2, and ISDN ports.

Configuring the network clock

Use this procedure to specify the selection priority for ports that provide timing for the network clock. You must specify a BRI or T1/E1 small module (ISDN PRI, T1 CAS, or E1 R2) configured for voice as the clock source.

Procedure steps

1. To enter configuration mode, enter:

   ```
   configure terminal
   ```

2. To set the priority for the network clock source ports, enter:

   ```
   network-clock-select {1|2} <slot/port>
   ```

Table 109: Variable definitions

Variable	Value
{1 | 2}	Specifies the priority of the clock source.
<slot/port>	Specifies the slot and port for the clock source (only T1/E1 or BRI cards in small slots).

Displaying the network clock configuration

Use this procedure to display the network clock configuration.

Procedure steps

To display the network clock configuration, enter:

```
show network-clocks
```

Custom tone configuration for T1/E1 PRI and CAS

Use the following procedures to configure custom ringback tones.

The required steps are:

1. Define dualtone classes.
2. Create the custom tone (only ringback tone is supported).
3. Apply the custom tone to the voice port (only supported on T1/E1 PRI and CAS).

Defining dualtones for a specific country

Use the following procedure to define dualtones for your specific country.

Procedure steps

1. To enter configuration mode, enter:
   `configure terminal`
2. To specify voice class configuration, enter:
   `voice class`
3. To specify the dualtone name, enter:
   `dualtone <dualtone>`
4. To add a frequency pair to the dualtone, enter:
   `freq-pair <freq-pair>`
5. To set amplitudes for the frequency pair, enter:
   `amplitude-pair <amplitude-pair>`
6. To set on/off times for the frequency pair, enter:
   `cadence-list <cadence-list>`
7. To specify the number of times to repeat the cadence pattern, enter:
   `repeat-cadence <repeat-cadence>`

Table 110: Variable definitions

Variable	Value
<dualtone>	Specifies a name for the dualtone.

Variable	Value
<freq-pair>	Specifies the order or frequency pair to play. Range: 1-4.
<amplitude-pair>	Specifies amplitude of the frequency pair. Range: 1-4.
<cadence-list>	Specifies on-off times for the fequency pair. Range: 1-4.
<repeat-cadence>	Specifies the repeat count for the cadence block. Range: 1-256. Default is 256 (infinite).

Example of defining dualtones for a specific country

The following procedure shows an example configuration for defining dualtones.

Procedure steps

1. Define dualtone 1:

```
configure terminal
voice class
dualtone tone1
freq-pair 1 425 0
amplitude-pair 1 -200 0
cadence-list 1 1000 4000
exit dualtone
```

2. Define dualtone 2:

```
dualtone tone2
freq-pair 1 440 480
amplitude-pair 1 -200 -200
cadence-list 1 2000 4000
exit dualtone
```

3. Define dualtone 3:

```
dualtone tone3
freq-pair 1 400 450
freq-pair 2 400 450
amplitude-pair 1 -200 -200
amplitude-pair 2 -200 -200
cadence-list 1 400 200
cadence-list 2 400 2000
exit dualtone
exit class
```

Creating custom tone classes

Create custom tone classes by associating a class with dualtones.

Procedure steps

1. To enter configuration mode, enter:

   `configure terminal`
2. To specify voice class configuration, enter:

   `voice class`
3. To specify the custom tone name, enter:

   `custom-cptone <custom-cptone>`
4. To specify the dualtone to use for ringback-tone, enter:

   `ringback-tone <dualtone>`

Table 111: Variable definitions

Variable	Value
<custom-cptone>	Specifies a name for the custom tone.
<dualtone>	Specifies the name of the dualtone to use.

Example of creating custom tone classes

The following procedure shows an example configuration for creating custom tone classes.

Procedure steps

1. Define a custom tone class for Germany:

```
configure terminal
voice class
custom-cptone Germany
ringback-tone dualtone tone1
exit custom-cptone
```

2. Define a custom tone class for the US:

```
custom-cptone US
ringback-tone dualtone tone2
exit custom-cptone
```

3. Define a custom tone class for Britain:

```
custom-cptone Britain
ringback-tone dualtone tone3
exit custom-cptone
exit class
```

Applying the custom tone to the voice port

After you configure the custom tone class, you can apply it to the voice port. You can add more than one class to more than one voice port.

Procedure steps

1. To enter configuration mode, enter:

   `configure terminal`

2. To specify voice port configuration, enter:

   `voice-port <slot/[<subslot/]port>`

3. To apply the custom tone to the port, enter:

   `voice-class custom-cptone <custom-cptone>`

Table 112: Variable definitions

Variable	Value
<custom-cptone>	Specifies the name of the custom tone to apply to the port.

Example of applying a custom tone to a voice port

The following procedure shows an example configuration for applying a custom tone to a voice port.

Procedure steps

Apply a custom tone to voice port 3/1:

```
configure terminal
cvoice-port 3/1
no shutdown
voice-class custom-cptone Britain
exit voice-port
```

Common procedures for POTS and VoIP dial peers

The following sections describe the common procedures applicable to POTS and VoIP dial peers.

Creating a dial peer

Use this procedure to define a dial peer and to specify the method of voice encapsulation.

Procedure steps

1. To enter configuration mode, enter:

   `configure terminal`

2. To create a dial peer, enter:

   `[no] dial-peer voice {pots | voip} <tag>`

Table 113: Variable definitions

Variable	Value
pots	Indicates a POTS peer that uses VoIP encapsulation on the IP backbone.
voip	Indicates a VoIP peer that uses voice encapsulation on the POTS network.
<tag>	Specifies a tag for the dial peer. Valid range is: 1–2147483647.
[no]	Deletes a defined dial peer. Before you can delete a dial peer, you must disable it using the **`shutdown`** command from the **`configure dial-peer voice {pots | voip}`** subtree.

Configuring a destination pattern for a dial peer

Use this procedure to specify a destination pattern for the E.164 or private dialing plan telephone number to use for a dial peer.

By default, no destination pattern is defined.

The Media Gateway uses the pattern you configure to match dialed digits to a dial peer. The dial peer is then used to complete the call. When a gateway receives voice data, it compares the called number (the full E.164 telephone number) in the packet header with the number configured as the destination pattern for the dial peer.

When you configure the destination pattern, set the string to match the local dialing conventions. There are certain areas in the world (for example, certain European countries) where valid telephone numbers can vary in length. Use the optional control character T to indicate that a particular destination pattern value is a variable-length dial string. In this case, the system does not match the dialed numbers until the interdigit timeout value has expired.

Procedure steps

1. To enter configuration mode, enter:

   `configure terminal`

2. To select a dial peer, enter:

   `dial-peer voice {pots | voip} <tag>`

3. To specify the partial or full E.164 telephone number to be used for the dial peer, enter:

   `[no] destination-pattern <destination-pattern>`

Table 114: Variable definitions

Variable	Value
<destination-pattern>	Specifies a pattern for the E.164 or private dialing plan telephone number. Valid entries are the digits 0 through 9, and the following characters: • Period (.): matches any entered digit (this character is used as a wildcard). • Percent sign (%): indicates that the preceding digit occurred zero or more times; similar to the wildcard usage. • Plus sign (+): indicates that the preceding digit occurred one or more times. • Brackets ([ ]): indicate a range. A range is a sequence of characters enclosed in the brackets; only numeric characters from 0 to 9 are allowed in the range. • T: indicates that the destination pattern value is a variable-length dial string.
[no]	Removes the configured destination pattern.

Examples of configuring a destination pattern for a dial peer

The following example shows the configuration of a destination pattern in which the preceding digit pattern is repeated multiple times:

```
configure terminal
dial-peer voice voip 200
destination- pattern 555%
```

The following example shows the configuration of a destination pattern in which the possible numeric values are between 5550109 and 5550199:

```
dial-peer voice voip 300
destination- pattern 55501[0-9]9
```

The following example shows the configuration of a destination pattern in which the possible numeric values are 5550439, 5553439, 5555439, 5557439, and 5559439:

```
dial-peer voice voip 400
destination- pattern 555[03579]439
```

Configuring a description for a dial peer

Use this procedure to attach a text string description to the connection for a dial peer. This description appears in various displays and you can use it to track the purpose or use of the dial peer.

By default, no description is configured.

Procedure steps

1. To enter configuration mode, enter:

   `configure terminal`

2. To select a dial peer, enter:

   `dial-peer voice {pots | voip} <tag>`

3. To a description to this dial peer, enter:

   `[no] description "<description>"`

Table 115: Variable definitions

Variable	Value
"<description>"	Specifies a description for the dial peer. Valid value is a character string from 1 to 25 characters in length, delimited by quotation marks (" ").
[no]	Removes the description.

Enabling a dial peer

Use this procedure to enable and disable the selected dial peer. By default, dial peers are disabled. To enable the dial peer, use the no form of this command.

Procedure steps

1. To enter configuration mode, enter:

   `configure terminal`
2. To select a dial peer, enter:

   `dial-peer voice {pots | voip} <tag>`
3. To enable or disable the dial peer, enter:

   `[no] shutdown`

Displaying dial peer information

Use this procedure to display information for voice dial peers.

Procedure steps

To display information for voice dial peers, enter:

`show dial-peer voice {<tag> | summary}`

Table 116: Variable definitions

Variable	Value
<tag>	Displays detailed information about the dial peer specified by the tag. Valid range is from 1 to 2147483647.
summary	Displays a short summary of each voice dial peer.

Clearing call counters and call details for a dial peer

Use this procedure to reset voice call counters and recent call details stored in a dial peer.

Procedure steps

To clear call counters and call details for a dial peer, enter:

`clear statistics dial-peer voice <tag>`

Table 117: Variable definitions

Variable	Value
<tag>	Clears statistics for the specified dial peer. Valid range is any integer that identifies a specific dial peer, from 1 to 2147483647.

Chapter 21: Displaying active call information

The following sections describe procedures that you use to display the status of active calls on the SR2330/4134.

- Displaying active voice call information on page 203
- Displaying SIP UA client and server information for active SIP calls on page 203

Displaying active voice call information

Use this procedure to display call information for voice calls in progress.

Procedure steps

To display call information for voice calls in progress, enter:

```
show call active voice [called-number <called-number> |
calling-number <calling-number>]
```

Table 118: Variable definitions

Variable	Value
called-number <called-number>	Displays specific called number pattern related information.
calling-number <calling-number>	Displays specific calling number pattern related information.

Displaying SIP UA client and server information for active SIP calls

Use this procedure to display active user agent client (UAC) and user agent server (UAS) information on SIP calls.

Procedure steps

To display active UAC and UAS information on SIP calls, enter:

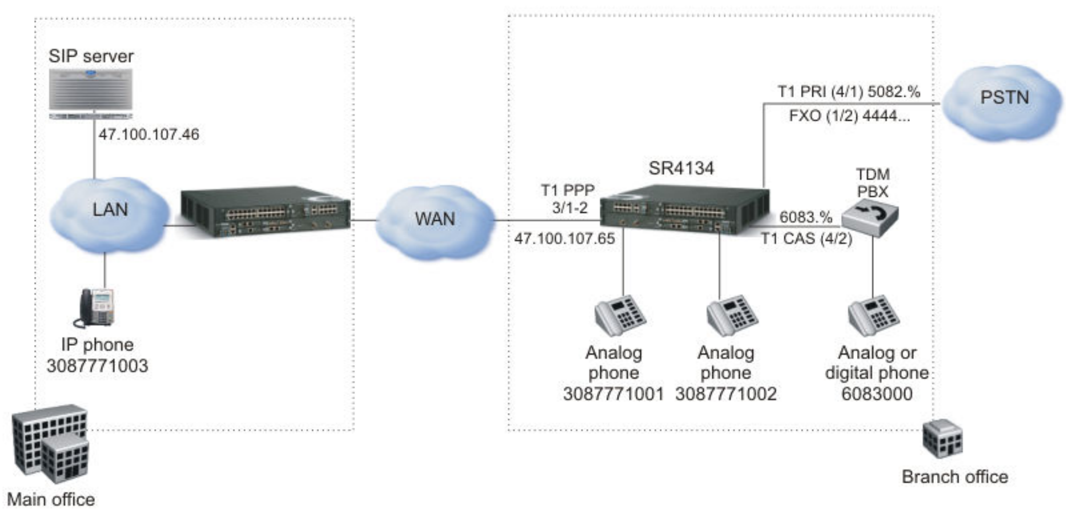
```
show sip-ua calls
```

Chapter 22: Configuration example

This chapter contains a basic Avaya Secure Router 2330/4134 Media Gateway configuration example.

Example of basic SR23300/4134 Media Gateway configuration

The following figure shows a sample SR2330/4134 Media Gateway configuration.



1. Display the DSP status:

```
show voice dsp status
```

```
SR# show voice dsp status

DSP Status
MSP Version : v6_11 Rev E - HP
T38 Version : T38DDP_VER_5_1_7
PVDM Present : Present
MSP Status : Running
RTP port validation : Enabled
Max Channel : 8
Active VOIP Channels : 0
Active FOIP Channels : 0
Additional G.711 10ms Channels: 4
Additional G.711 20ms Channels: 8
Additional G.723 Channels: 4
Additional G.726 Channels: 4
Additional G.729 Channels: 4
Additional T.38 Channels: 2
```

2. Display chassis status:

```
show chassis
```

```
SR# show chassis

Chassis Model: SR4134
Chassis Operational Status: NORMAL

Chassis Serial number: LBNNTMJV4111H34013
Chassis Rev: 8

Slot/SubSlot   Card-Type      Status        Serial#
--------------------------------------------------------
   0           MPU_A          NORMAL        LBNNTMJV7718H29016
   4           WTE_2M         NORMAL        LB*
   3           WTE_2M         NORMAL        LB*
   1           FXO_4M         NORMAL        LB*
   2           FXS_4M         NORMAL        LB*
  INT          VOIP_A         NORMAL        ---
```

In the bottom INT row, the internal VOIP_A module refers to the PVM module, which must be installed to provide voice capabilities. The slot 0, MPU_A row refers to the motherboard and the serial number displayed is the one you must provide to upgrade the DSP license, if required.

3. If required, upgrade the DSP license:

```
config term
system licenses pvm_channels 64
```

```
Enter License key:
```

```
048e7b051ecea5c41f
```

```
Requested Change is Done.

Please reboot the box for this change to take effect.
```

```
exit
reboot
```

```
Continue with reboot ? (y/n): y
```

```
y
```

```
show voice dsp status
```

```
SR# show voice dsp status


DSP Status
MSP Version : v6_11_0_5 Rev E - HP
T38 Version : T38DDP_VER_5_1_7
PVDM Present : Present
MSP Status : Running
RTP port validation : Enabled
Max Channel : 64
Active VOIP Channels : 0
Active FOIP Channels : 0
Additional G.711 10ms Channels: 32
Additional G.711 20ms Channels: 64
Additional G.723 Channels: 32
Additional G.726 Channels: 32
Additional G.729 Channels: 32
Additional T.38 Channels: 16
```

4. Configure a data interface for communication with the SIP server:

```
interface bundle wan1
link t1 3/1-2
encapsulation ppp
ip address 47.100.107.65 255.255.255.0
exit
show ip interfaces
```

```
SR/configure# show ip interfaces

wan1
Type   : PT2PT
Flags   : 0x2078243  UP, RUNNING, ATTACHED, BROADCAST, MULTICAST
Internet Address   : 47.100.107.65
Internet Netmask  : 255.255.255.0
Internet Broadcast   : 47.100.107.255
Interface RED is enabled
CBQ is enabled for outbound traffic
Peer IP Address   : 47.100.107.46
Layer2 Protocol   : PPP
Maximum Transfer Unit   : 1500 bytes
Mac Address:   00:00:1b:ba:64:c7
```

5. Specify the SIP server and registrar (for Avaya CS 1000 and OCS, specify only the SIP server):

```
sip-ua
sip-server ipv4:47.100.107.46:5060
registrar ipv4:47.100.107.46 expires 3600
exit
show sip-ua status
```

```
SR/configure# show sip-ua status

SIP User Agent Status
SIP User Agent for UDP : ENABLED
SIP User Agent for TCP : ENABLED
SIP User Agent bind status(signaling): ENABLED
SIP max-forwards : 70
SIP server : ipv4:47.100.107.46:5060
Registrar server : ipv4:47.100.107.46 expires 3600
Authentication username : n/a
Authentication password : n/a
 SDP application configuration:
 Version line (v=) required
 Session name line (s=) required
 Timespec line (t=) required
 Network types supported: IN
 Address types supported: ipv4
 Transport types supported: RTP/AVP UDP/TCP
```

6. Specify the IP address configured in step 4 as the source IP for SIP and Media (and configure a pass-through prefix):

```
voice service voip
sip
bind all ipv4:47.100.107.65:5060
exit
pass-through- call-prefix 9
exit
show voice service voip
```

```
SR/configure# show voice service voip
call limit : 128
codec preference : 1, g711ulaw, bytes 160
codec preference : 2, g711alaw, bytes 160
dtmf relay rtp nte : Disabled
emergency-number : n/a
fax rate : 9600
fax protocol : none
modem passthrough : none
pass through call prefix : 9
comfort noise negotiate : Enabled
rtp port-validation : Enabled
Configured bind IP address : all ipv4:47.100.107.65:5060
Bound IP address : all ipv4:47.100.107.65:5060
rel1xx : Enabled
```

7. Configure the PBX link (T1 CAS):

```
module t1 4/2
cas-group timeslots 1-24 em-wink-start
exit
show module configuration t1 4/2
```

```
SR/configure# show module configuration t1 4/2
T1 4/2 is ENABLED
Alarm Hierarchy: TRUE,
Yellow Alarm: GENERATE & DETECT
Framing:ESF, LineCode:B8ZS, ClockSource: NET, LineMode:CSU, LBO:0 db
FDL: ANSI Unit Protocol enabled ,ATT Unit Protocol disabled ,
CsuDsuType:  CSU , Loopback Framing (In-band): Overwrite,
CIRCUIT-ID : Not Configured ,CONTACT-INFO : Not Configured ,
DESCRIPTION : Not Configured , LINK NAME : Not Configured ,

Line Status:
   RLOS:OFF,   RAIS:OFF,   RLOF:OFF,   RRAI:OFF,   TAIS:OFF
   TRAI:OFF,  TPtrn:OFF,   Loop:OFF

Timeslot Map:
  1  2  3  4  5  6  7  8  9 10 11 12 13 14 15 16 17 18 19 20 21 22 23
24

______________________________________________________________________
|0 |0 |0 |0 |0 |0 |0 |0 |0 |0 |0 |0 |0 |0 |0 |0 |0 |0 |0 |0 |0 |0 |0 |0
|
|__|__|__|__|__|__|__|__|__|__|__|__|__|__|__|__|__|__|__|__|__|__|__|_
_|
```

```
show voice port 4/2
```

```
SR/configure# show voice port 4/2

Type of VoicePort : CAS
Operation state : UP
Administrative State : UP
Description : n/a
In Gain : 0
Output Attenuation : 0
Echo Cancellation : enabled
Echo Cancel coverage : 64
Comfort noise : disabled
Compand type : G711 U-Law
Trunk group : n/a
Analog Info Follows
Region Tone : us
Voice Card Specific Info Follows
Interdigit Duration Timing : 3
Station Name : n/a
Station Number : n/a
Initial timeout : 10
Timeslots : 1-24
Signal Type : CAS wink-start
```

8. Configure the CO digital link (T1 ISDN PRI):

```
interface bundle pri1
link pri_t1 4/1 voice
isdn
switch-type primary-ni2
activate
exit
exit
show interface bundle pri1
```

```
SR/configure# show interface bundle pri1

bundle pri1
-----------
status                              up
number of links                     1
voice-port info                     4/1:23
switch-type                         primary-ni2
side                                USR

counters for 4/1:23 since last boot/clear
    Bytes Rx                       1992    Bytes Tx
1967
    Packets Rx                      497    Packets Tx
493
    All Err Pkt Rx                    0    All Err Pkt Tx
0
    Pkt 2 Long Rx                     0    Failed Pkt Tx
0
    Overrun Rx                        0    Underrun Tx
0
    Pkt CRC Err Rx                    0
    Pkt Inv Len Rx                    0
    Abort Rx                          0
```

```
show voice port 4/1:23
```

```
SR/configure# show voice port 4/1:23

Type of VoicePort : PRI
Operation state : UP
Administrative State : UP
Description : n/a
In Gain : 0
Output Attenuation : 0
Echo Cancellation : enabled
Echo Cancel coverage : 64
Comfort noise : disabled
Compand type : G711 U-Law
Trunk group : n/a
Analog Info Follows
Region Tone : us
Voice Card Specific Info Follows
Timeslots : 1-23
Switch Type : PRI primary-ni2
```

```
debug isdn isdn-status pri1
```

```
SR# debug isdn isdn-status pri1

  SAP Id          = 0
  Sap State       = 3    DCh State    = UP
  Interface       = PRI T1
  Side            = USER
  Switch Type     = primary-ni2
  Bundle          = pri1

  BChn  BStatus Type    Id      Curr St         Curr Ev         Call
Orig
    1   Free    --      --      --              --              --
    2   Free    --      --      --              --              --
    3   Free    --      --      --              --              --
    4   Free    --      --      --              --              --
    5   Free    --      --      --              --              --
    6   Free    --      --      --              --              --
    7   Free    --      --      --              --              --
    8   Free    --      --      --              --              --
    9   Free    --      --      --              --              --
   10   Free    --      --      --              --              --
   11   Free    --      --      --              --              --
   12   Free    --      --      --              --              --
   13   Free    --      --      --              --              --
   14   Free    --      --      --              --              --
   15   Free    --      --      --              --              --
   16   Free    --      --      --              --              --
   17   Free    --      --      --              --              --
   18   Free    --      --      --              --              --
   19   Free    --      --      --              --              --
   20   Free    --      --      --              --              --
   21   Free    --      --      --              --              --
   22   Free    --      --      --              --              --
   23   Free    --      --      --              --              --
```

9. Configure the CO analog link (FXO) (PLAR is configured on the port to forward incoming FXO calls to the TDM PBX port at 6083000):

```
voice-port 1/2
signal loop-start
cptone us
ring-number 2
station number 4444003
no shutdown
connection plar 96083000
exit
show voice port 1/2
```

```
SR/configure# show voice port 1/2

Type of VoicePort : FXO
Operation state : UP
Administrative State : UP
Description : n/a
In Gain : 0
Output Attenuation : 0
Echo Cancellation : enabled
Echo Cancel coverage : 64
Comfort noise : disabled
Compand type : G711 U-Law
Trunk group : n/a
Analog Info Follows
Region Tone : us
Voice Card Specific Info Follows
Ring number : 2
Connection plar : 96083000
ANI mapping table:
not configured
Signal Type : FXO loop-start
Battery Reversal : enabled (no answer)
Caller-id Enable : Disabled
Caller-id Block : disabled
Caller-id Alerting Ring : disabled
Interdigit Duration Timing : 10
Station Name : n/a
Station Number : 4444003
```

10. Configure the analog port for phone 1:

```
voice-port 2/1
signal loop-start
cptone us
caller-id enable
caller-id alerting ring 1
station number 3087771001
no shutdown
exit
show voice port 2/1
```

```
SR/configure# show voice port 2/1

Type of VoicePort : FXS
Operation state : UP
Administrative State : UP
Description : n/a
In Gain : 0
Output Attenuation : 0
Echo Cancellation : enabled
Echo Cancel coverage : 64
Comfort noise : disabled
Compand type : G711 U-Law
Trunk group : n/a
Analog Info Follows
Region Tone : us
Voice Card Specific Info Follows
Initial timeout : 10
Supervisory disconnect : enabled
Signal Type : FXS loop-start
Battery Reversal : enabled (no answer)
Caller-id Enable : Enabled (type 1)
Caller-id Block : disabled
Caller-id Alerting Ring : enabled (type 1)
Interdigit Duration Timing : 10
Station Name : n/a
Station Number : 3087771001
```

11. Configure the analog port for phone 2:

```
voice-port 2/2
signal loop-start
cptone us
caller-id enable
caller-id alerting ring 1
station number 3087771002
no shutdown
exit
show voice port 2/2
```

```
SR/configure# show voice port 2/2

Type of VoicePort : FXS
Operation state : UP
Administrative State : UP
Description : n/a
In Gain : 0
Output Attenuation : 0
Echo Cancellation : enabled
Echo Cancel coverage : 64
Comfort noise : disabled
Compand type : G711 U-Law
Trunk group : n/a
Analog Info Follows
Region Tone : us
Voice Card Specific Info Follows
Initial timeout : 10
Supervisory disconnect : enabled
Signal Type : FXS loop-start
Battery Reversal : enabled (no answer)
Caller-id Enable : Enabled (type 1)
Caller-id Block : disabled
Caller-id Alerting Ring : enabled (type 1)
Interdigit Duration Timing : 10
Station Name : n/a
Station Number : 3087771002
```

12. Verify the status of all voice ports:

```
show voice port summary
```

```
SR/configure# show voice port summary

PORT   SIG-TYPE                 ADMIN OPER EC
====== ======================== ===== ==== ==
4/1:23 PRI primary-ni2          up    up   y
4/2    CAS wink-start           up    up   y
1/2    FXO loop-start           up    up   y
2/1    FXS loop-start           up    up   y
2/2    FXS loop-start           up    up   y
```

13. Configure the clock source:

```
network-clock-select 1 4/1
show network-clocks
```

```
SR/configure# show network-clocks

Network Clock Configuration
---------------------------
Priority | Clock Source | Clock Type | Clock State
   1     | T1 4/1       | T1         | GOOD
   3     | Mainboard    | PLL        | GOOD

Current Clock Source
--------------------
Priority | Clock Source | Clock Type | Clock Status
   1     | T1 4/1       | T1         | LOCKING
```

14. Configure the dial plan (dial peer) for the PBX link:

```
dial-peer voice pots 2
destination-pattern 6083.%
port 4/2
forward-digits all
no shutdown
exit
show dial-peer voice 2
```

```
SR/configure# show dial-peer voice 2

description : n/a
tag = 2, destination pattern : 6083.%
Admin state : Enabled, Operation state : Enabled
type : pots
prefix : n/a
port : 4/2
digit-strip : Disabled
trunkgroup : n/a
forward-digits : all
Register E.164 number : Disabled
clid : n/a
Translation Profile :
   Incoming : n/a
   Outgoing : n/a

Dial-Peer Statistics:

Successful calls  : 0
Failed calls: 0
Accepted calls: 0
Refused calls: 0
Connect Time : 0
Last Disconnect Reason : ""
```

15. Configure the dial plan (dial peer) for the CO digital link:

```
dial-peer voice pots 11
destination-pattern 5082.%
port 4/1
no digit-strip
no shutdown
exit
show dial-peer voice 11
```

```
SR/configure# show dial-peer voice 11

description : n/a
tag = 11, destination pattern : 5082.%
Admin state : Enabled, Operation state : Enabled
type : pots
prefix : n/a
port : 4/1:23
digit-strip : Disabled
trunkgroup : n/a
forward-digits : all
Register E.164 number : Disabled
clid : n/a
Translation Profile :
   Incoming : n/a
   Outgoing : n/a

Dial-Peer Statistics:

Successful calls  : 0
Failed calls: 0
Accepted calls: 0
Refused calls: 0
Connect Time : 0
Last Disconnect Reason : ""
```

16. Configure the dial plan (dial peer) for the CO analog link:

```
dial-peer voice pots 6
destination-pattern 4444...
```

```
port 1/2
no digit-strip
no shutdown
exit
show dial-peer voice 6
```

```
SR/configure# show dial-peer voice 6

description : n/a
tag = 6, destination pattern : 4444...
Admin state : Enabled, Operation state : Enabled
type : pots
prefix : n/a
port : 1/2
digit-strip : Disabled
trunkgroup : n/a
forward-digits : all
Register E.164 number : Disabled
clid : n/a
Translation Profile :
   Incoming : n/a
   Outgoing : n/a

Dial-Peer Statistics:

Successful calls  : 0
Failed calls: 0
Accepted calls: 0
Refused calls: 0
Connect Time : 0
Last Disconnect Reason : ""
```

17. Configure the dial plan (dial peer) for analog phone 1 (for CS 1000 and OCS, the authentication and register commands are not required):

```
dial-peer voice pots 3
destination-pattern 3087771001
port 2/1
no digit-strip
no shutdown
authentication 3087771001 3087771001
register e164
exit
show dial-peer voice 3
```

```
SR/configure# show dial-peer voice 3

description : n/a
tag = 3, destination pattern : 3087771001
Admin state : Enabled, Operation state : Enabled
type : pots
prefix : n/a
port : 2/1
digit-strip : Disabled
trunkgroup : n/a
forward-digits : none
Register E.164 number : Enabled
clid : n/a
Translation Profile :
   Incoming : n/a
   Outgoing : n/a

Dial-Peer Statistics:

Successful calls  : 0
Failed calls: 0
Accepted calls: 0
Refused calls: 0
Connect Time : 0
Last Disconnect Reason : ""
```

18. Configure the dial plan (dial peer) for analog phone 2 (for CS 1000 and OCS, the authentication and register commands are not required):

```
dial-peer voice pots 4
destination-pattern 3087771002
port 2/2
no digit-strip
no shutdown
authentication 3087771002 3087771002
register e164
exit
show dial-peer voice 4
```

```
SR/configure# show dial-peer voice 4

description : n/a
tag = 4, destination pattern : 3087771002
Admin state : Enabled, Operation state : Enabled
type : pots
prefix : n/a
port : 2/2
digit-strip : Disabled
trunkgroup : n/a
forward-digits : none
Register E.164 number : Enabled
clid : n/a
Translation Profile :
   Incoming : n/a
   Outgoing : n/a

Dial-Peer Statistics:

Successful calls  : 0
Failed calls: 0
Accepted calls: 0
Refused calls: 0
Connect Time : 0
Last Disconnect Reason : ""
```

19. Configure the dial plan (dial peer) for the SIP server (optional):

```
dial-peer voice voip 20
destination-pattern 3087771000[0-9]
session target ipv4:47.100.107.46:5060
no shutdown
exit
exit
show dial-peer voice 20
```

```
SR/configure# show dial-peer voice 20

description : n/a
tag = 20, destination pattern : 308777100[0-9]
Admin state : Enabled, Operation state : Enabled
type : voip
RTP payload : Disabled
DTMF Relay : Disabled
session-target : ipv4:47.100.107.46:5060
Fax : system
Fax rate : n/a
Modem : n/a
comfort noise negotiate : Enabled
Translation Profile :
   Incoming : n/a
   Outgoing : n/a

Dial-Peer Statistics:

Successful calls  : 0
Failed calls: 0
Accepted calls: 0
Refused calls: 0
Connect Time : 0
Last Disconnect Reason : ""
```

20. Verify the dial plan and port association:

```
show dial-peer voice summary
```

```
SR#  show dial-peer voice summary
                                                 PASS
TAG        TYPE ADMIN OPER PREFIX   DEST-PATTERN THRU SESSION-
TARGET/PORT
========== ==== ===== ==== ======== ============ ====
===================
20         voip up    up            308777100[0-9]
ipv4:47.100.107.46:5060
3          pots up    up            3087771001        2/1
4          pots up    up            3087771002        2/2
6          pots up    up            4444...           1/2
11         pots up    up            5082.%            4/1:23
2          pots up    up            6083.%            4/2
```

21. Verify analog port SIP registration

```
show sip-ua register status
```

```
SR# show sip-ua register status

Line      peer                           expires(sec)  registered
0         3087771001                     3600          yes
1         3087771002                     3600          yes
```

Make and verify calls between analog phones.

1. Display active calls:

```
show call active voice
```

```
SR# show call active voice

GENERIC:
Call Id=992
calling number=3087771002
called number=3087771001
Dial Peer Tag=3
Destination Pattern=3087771001
Voice Port =2/1
SessionTarget=47.100.107.65
VAD = Disabled
codecName=PCMU
CodecBytes=160
compandType=G711 U-Law
SignalingType=FXS loop-start
CallDuration= 0: 1:31
CallState=ACTIVE
CallOrigin: answer


GENERIC:
Call Id=1023
calling number=3087771002
called number=3087771001
Dial Peer Tag=20
Destination Pattern=308777100[0-9]
Voice Port =2/2
SessionTarget=47.100.107.46
VAD = Disabled
codecName=PCMU
CodecBytes=160
compandType=G711 U-Law
SignalingType=FXS loop-start
CallDuration= 0: 1:31
CallState=ACTIVE
CallOrigin: originate
```

2. Display media ports for the active calls:

```
show sip-ua calls
```

```
SR# show sip-ua calls

SIP UAC CALL INFO

Call 1
State of the call : ACTIVE
Calling Number : 3087771002
Called Number : 3087771001
Source IP Address (Sig ): 47.100.107.46
Number of Media Streams : 1
Media Stream
Negotiated Codec : PCMU Bytes: 160
Codec Payload Type : 0
Media Source IP Addr:Port: 47.100.107.65:28000
Media Dest IP Addr:Port : 47.100.107.65:28002

Call 2
State of the call : ACTIVE
Calling Number : 3087771002
Called Number : 3087771001
Source IP Address (Sig ): 47.100.107.46
Number of Media Streams : 1
Media Stream
Negotiated Codec : PCMU Bytes: 160
Codec Payload Type : 0
Media Source IP Addr:Port: 47.100.107.65:28002
Media Dest IP Addr:Port : 47.100.107.65:28000
```

3. Display DSP channels:

```
show voice dsp status
```

```
SR# show voice dsp status


DSP Status
MSP Version : v6_11_0_5 Rev E - HP
T38 Version : T38DDP_VER_5_1_7
PVDM Present : Present
MSP Status : Running
RTP port validation : Enabled
Max Channel : 64
Active VOIP Channels : 2
Active FOIP Channels : 0
Additional G.711 10ms Channels: 31
Additional G.711 20ms Channels: 62
Additional G.723 Channels: 31
Additional G.726 Channels: 31
Additional G.729 Channels: 31
Additional T.38 Channels: 15
```

4. Display DSP statistics and connections:

```
show voice dsp statistics
```

```
SR# show voice dsp statistics


DSP Statistics
count of msp rebooted : 1
msp channels created : 2
msp channels created successfully : 2
failed msp channel creations due to resources : 0
failed msp channel creations due to busy channels : 0
count of active media : 2
active media conn id : 1024
count out of sequence rtp : 0
count of bad rtp : 0
count of late rtp : 0
count of discarded rtp : 0
peak jitter in unit of ms : 0
active media conn id : 993
count out of sequence rtp : 0
count of bad rtp : 0
count of late rtp : 0
count of discarded rtp : 0
peak jitter in unit of ms : 0
```

5. Display SIP message statistics:

```
show sip-ua statistics
```

```
SR# show sip-ua statistics

SIP Response Statistics (Inbound/Outbound)
        Informational
          Trying  1/1, Ringing  0/0,  Forwarded  0/0,
          Queued  0/0, Session Progress  0/0
        Success
          OkInvite  1/1, OkBye  0/0, OkCancel  0/0, OkOptions  0/0,
          OkPrack  0/0, OkSubscribe(R)  0, OkNotify  0/0, OkInfo  0/0,
          202Accepted  0/0, OkRegister  2
        Redirection(Inbound only)
          MultipleChoice  0, MovedPermanently  0, Moved Temporarily  0,
          UserProxy  0, AlternateService  0
        Client Error:
          BadRequest  0/0, Unauthorized  11/0, Payment Required  0/0,
          Forbidden  0/0, NotFound  0/0, Not Acceptable  0/0,
          ProxyAuthReqd  0/0,Request Timeout  0/0, Conflict  0/0,
          Gone  0/0, UnsupportedMediaType  0/0, BadExtension  0/0,
          TempNotAvailable  0/0, Loop Detected  0/0, Too Many Hops
0/0,
          CallLegNonExistent  0/0, AddrIncomplete  0/0, Ambiguous  0/0,
BusyHere  0/0,
          ReqCancel  0/0, NotAcceptableMedia  0/0, Bad Event  0/0
        Server Error:
          InternalError  0/0, NotImplemented  0/0, BadGateway  0/0,
ServiceUnavail  0/0,
          GatewayTimeout  0/0, BadSipVer  0/0, PreCondfailure  0/0
        Global Failure:
          BusyEverywhere 0/0, Decline 0/0, Doesnot exist anywhere 0/0,
Not Acceptable 0/0
SIP Total Traffic Statistics(Inbound/Outbound)
          Invite  1/1, Ack  1/2, Bye  0/0,
          Cancel  0/0, Options  0/0, Prack  0/0,
          Subscribe(Outbound)  0, Notify  2/0, Refer  0/0, Info  0/0,
          Register(Outbound)  2
```

Make and verify calls between analog phones and the head office phone.

1. Trace a call between an analog phone and the head office phone:

```
debug voice cc routing
```

```
SR# debug voice cc routing

ROUTING[-----]: Call Limit is 128, active calls = 0
ROUTING[-----]: Incoming port = 2/2
ROUTING[-----]: Incoming DP found with tag = 4!!!
ROUTING[-----]: Incoming DP found dpIndex=1
ROUTING[-----]: No matching trans-profile
ROUTING[-----]: Dial peer pots matched with index 1 and clidOption = 0
ROUTING[-----]: DP clid not configured
ROUTING[-----]: No calling party number found. Check station number
configuration
ROUTING[-----]: Station number configured3087771002
ROUTING[-----]: No matching trans-profile
ROUTING[-----]: Call Limit is 128, active calls = 0
ROUTING[-----]: Receives called party number 93087771001
ROUTING[-----]: Call Id: 1023,T38 bit rate is set to
9600(gbl_tcce_config)
ROUTING[-----]: dial string i=9, prefix j=9
ROUTING[-----]: No dial-peer voip found for 93087771001
ROUTING[-----]: dial string i=9, prefix j=9
ROUTING[-----]: Matching dial peer pots found for 3087771001
ROUTING[-----]: pots match PT prefix hairpin for 93087771001
ROUTING[-----]: Called party number 3087771001
ROUTING[-----]: Called party address 47.100.107.65
ROUTING[-----]: Called party name
ROUTING[-----]: Called party port 5060
ROUTING[-----]: Set the ds0 Busy
ROUTING[-----]: Calling party number is 3087771002
ROUTING[-----]: Calling party name is
ROUTING[-----]: Calling party address is 47.100.107.65
ROUTING[-----]: No matching trans-profile
ROUTING[-----]: No matching trans-profile
ROUTING[-----]: The destination number is 3087771001, lineId = -1
ROUTING[-----]: Call Limit is 128, active calls = 1
ROUTING[-----]: Outgoing DP found with tag = 3!!!
ROUTING[-----]: Destination is Port
ROUTING[-----]: Port Id = 32
ROUTING[-----]: Line id = -1, Go for channel hunt
ROUTING[-----]: Outgoing port = 2/1
ROUTING[-----]: Route Mgr set ds0 idle = 1
ROUTING[-----]: Route Mgr populates called party number = 3087771001
ROUTING[-----]: Route Mgr receives called party name =
ROUTING[-----]: TCCE finds line id = 0
ROUTING[-----]: Returning CC 200
ROUTING[-----]: No matching trans-profile
ROUTING[-----]: No matching trans-profile
ROUTING[-----]: Call Id: 9600, T38 bit rate is set to
910910036(gbl_tcce_config)
```

2. Trace a call between analog phones using prefix “9”:

```
debug voice cc routing
```

```
SR# debug voice cc routing

ROUTING[-----]: Call Limit is 128, active calls = 0
ROUTING[-----]: Incoming port = 2/2
ROUTING[-----]: Incoming DP found with tag = 4!!!
ROUTING[-----]: Incoming DP found dpIndex=1
ROUTING[-----]: No matching trans-profile
ROUTING[-----]: Dial peer pots matched with index 1 and clidOption = 0
ROUTING[-----]: DP clid not configured
ROUTING[-----]: No calling party number found. Check station number
configuration
ROUTING[-----]: Station number configured3087771002
ROUTING[-----]: No matching trans-profile
ROUTING[-----]: Call Limit is 128, active calls = 0
ROUTING[-----]: Receives called party number 3087771003
ROUTING[-----]: Call Id: 1023,T38 bit rate is set to
9600(gbl_tcce_config)
ROUTING[-----]: dial string i=3, prefix j=9
ROUTING[-----]: Matching dial peer voip found for 3087771003
ROUTING[-----]: Session-target found for 3087771003
ROUTING[-----]: No matching trans-profile
ROUTING[-----]: No matching trans-profile
ROUTING[-----]: Call Id: 1023,T38 bit rate is set to
9600(dp_tcce_config)
ROUTING[-----]: voip match with session target
ROUTING[-----]: Called party number 3087771003
ROUTING[-----]: Called party address 47.100.107.46
ROUTING[-----]: Called party name
ROUTING[-----]: Called party port 5060
ROUTING[-----]: Set the ds0 Busy
ROUTING[-----]: Calling party number is 3087771002
ROUTING[-----]: Calling party name is
ROUTING[-----]: Calling party address is 47.100.107.46
```

3. Trace SIP messages for call establishment between an analog phone and the head office phone:

```
debug ccsip messages
```

```
SR# debug ccsip messages

 [TCCE-TRACE]:1203000790.510 Msg-OUT           Call-OUT          SIP
Interface
    SIP Message PDU
INVITE sip:3087771003@47.100.107.46 SIP/2.0
Via: SIP/2.0/UDP 47.100.107.65:5060;branch=z9hG4bK824118656-52
Max-Forwards: 70
Supported: timer,100rel,replaces
From: <sip:3087771002@47.100.107.46>;tag=SR4134_824118656-53
To: <sip:3087771003@47.100.107.46>
Call-ID: 824118656-51
CSeq: 1 INVITE
Min-SE: 90
Session-Expires: 1800;refresher=uas
Contact: <sip:3087771002@47.100.107.65>
P-Asserted-Identity:  <sip:3087771002@47.100.107.46>
Content-Type: application/sdp
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,OPTIONS,PRACK,INFO
Content-Length: 201

v=0
o=SR4134 12345 787 IN IP4 47.100.107.65
s=Session
c=IN IP4 47.100.107.65
t=0 0
m=audio 28002 RTP/AVP 0 8
a=rtcp:28003
a=ecan:fb on -
a=ptime:20
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000

[TCCE-TRACE]:1203000790.530 Msg-IN            Call-OUT          SIP
Interface
    SIP Message PDU
SIP/2.0 401 UnAuthorized
Via: SIP/2.0/UDP 47.100.107.65:5060;branch=z9hG4bK824118656-52
CSeq: 1 INVITE
Call-ID: 824118656-51
From: <sip:3087771002@47.100.107.46>;tag=SR4134_824118656-53
To: <sip:3087771003@47.100.107.46>;tag=1551681419691433
WWW-Authenticate: Digest
realm="Registered_Subscribers",domain="sip:sylantro.nortel.com",nonce
="ddf731ebf745099ba6bf5af32aad03f6",opaque="",stale=FALSE,algorithm=MD5
Content-Length: 0
```

```
[TCCE-TRACE]:1203000790.530 Msg-OUT            Call-OUT          SIP
Interface
    SIP Message PDU
ACK sip:3087771003@47.100.107.46 SIP/2.0
Via: SIP/2.0/UDP 47.100.107.65:5060;branch=z9hG4bK824118656-52
Max-Forwards: 70
From: <sip:3087771002@47.100.107.46>;tag=SR4134_824118656-53
To: <sip:3087771003@47.100.107.46>;tag=1551681419691433
Call-ID: 824118656-51
CSeq: 1 ACK
Contact: <sip:3087771002@47.100.107.65>
Content-Length: 0


[TCCE-TRACE]:1203000790.530 Msg-OUT            Call-OUT          SIP
Interface
    SIP Message PDU
INVITE sip:3087771003@47.100.107.46 SIP/2.0
Via: SIP/2.0/UDP 47.100.107.65:5060;branch=z9hG4bK3644249472-54
Max-Forwards: 70
Supported: timer,100rel,replaces
From: <sip:3087771002@47.100.107.46>;tag=SR4134_824118656-53
To: <sip:3087771003@47.100.107.46>
Call-ID: 824118656-51
CSeq: 2 INVITE
Min-SE: 90
Session-Expires: 1800;refresher=uas
Contact: <sip:3087771002@47.100.107.65>
Content-Type: application/sdp
Authorization: Digest
username="3087771002",realm="Registered_Subscribers",nonce="ddf731ebf74
5
099ba6bf5af32aad03f6",uri="sip:3087771003@47.100.107.46",response="eaff
039ca9f85459557ecebc259
1d2e4",algorithm=MD5,opaque=""
P-Asserted-Identity:  <sip:3087771002@47.100.107.46>
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,OPTIONS,PRACK,INFO
Content-Length: 201
```

```
v=0
o=SR4134 12345 787 IN IP4 47.100.107.65
s=Session
c=IN IP4 47.100.107.65
t=0 0
m=audio 28002 RTP/AVP 0 8
a=rtcp:28003
a=ecan:fb on -
a=ptime:20
a=rtpmap:8 PCMA/8000
a=rtpmap:0 Púïä
[TCCE-TRACE]:1203000790.570 Msg-IN            Call-OUT          SIP
Interface
    SIP Message PDU
SIP/2.0 100 Trying
Via: SIP/2.0/UDP 47.100.107.65:5060;branch=z9hG4bK3644249472-54
CSeq: 2 INVITE
Call-ID: 824118656-51
From: <sip:3087771002@47.100.107.46>;tag=SR4134_824118656-53
To: <sip:3087771003@47.100.107.46>
Content-Length: 0


[TCCE-TRACE]:1203000790.830 Msg-IN            Call-OUT          SIP
Interface
    SIP Message PDU
SIP/2.0 180 Ringing
RSeq:  1
Via: SIP/2.0/UDP 47.100.107.65:5060;branch=z9hG4bK3644249472-54
CSeq: 2 INVITE
Call-ID: 824118656-51
From: <sip:3087771002@47.100.107.46>;tag=SR4134_824118656-53
To: <sip:3087771003@47.100.107.46>;tag=d7ad1506-1dd1-11b2-8292-
b03162323164+d7ad1506
Contact: <sip:3087771003@47.100.107.46:5075;transport=udp>
Require:  100rel
P-Asserted-Identity:  "phone1,polycom"<sip:1003@sylantro.nortel.com>
Content-Length: 0
```

```
[TCCE-TRACE]:1203000790.840 Msg-OUT          Call-OUT          SIP
Interface
    SIP Message PDU
PRACK sip:3087771003@47.100.107.46:5075;transport=udp SIP/2.0
Via: SIP/2.0/UDP 47.100.107.65:5060;branch=z9hG4bK111636864-55
Max-Forwards: 70
Authorization: Digest
username="3087771002",realm="Registered_Subscribers",nonce="ddf731ebf74
5
099ba6bf5af32aad03f6",uri="sip:3087771003@47.100.107.46",response="eaff
039ca9f85459557ecebc259
1d2e4",algorithm=MD5,opaque=""
Supported: replaces
From: <sip:3087771002@47.100.107.46>;tag=SR4134_824118656-53
To: <sip:3087771003@47.100.107.46>;tag=d7ad1506-1dd1-11b2-8292-
b03162323164+d7ad1506
Call-ID: 824118656-51
CSeq: 3 PRACK
RAck: 1 2 INVITE
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,OPTIONS,PRACK,INFO
Content-Length: 0

[TCCE-TRACE]:1203000790.850 Msg-IN           Call-OUT          SIP
Interface
    SIP Message PDU
SIP/2.0 200 OK
Via: SIP/2.0/UDP 47.100.107.65:5060;branch=z9hG4bK111636864-55
CSeq: 3 PRACK
Call-ID: 824118656-51
From: <sip:3087771002@47.100.107.46>;tag=SR4134_824118656-53
To: <sip:3087771003@47.100.107.46>;tag=d7ad1506-1dd1-11b2-8292-
b03162323164+d7ad1506
Content-Length: 0
```

```
[TCCE-TRACE]:1203000796.700 Msg-IN           Call-OUT          SIP
Interface
    SIP Message PDU
SIP/2.0 200 OK
Via: SIP/2.0/UDP 47.100.107.65:5060;branch=z9hG4bK3644249472-54
CSeq: 2 INVITE
Call-ID: 824118656-51
From: <sip:3087771002@47.100.107.46>;tag=SR4134_824118656-53
To: <sip:3087771003@47.100.107.46>;tag=d7ad1506-1dd1-11b2-8292-
b03162323164+d7ad1506
Contact: <sip:3087771003@47.100.107.46:5075;transport=udp>
Content-Type: application/sdp
Supported: timer
Session-Expires: 1800;Refresher=uas
Require:  timer
P-Asserted-Identity:  "phone1,polycom"<sip:1003@sylantro.nortel.com>
Allow: INVITE,BYE,ACK,CANCEL,PRACK,REFER,OPTIONS,REGISTER,NOTIFY
Content-Length: 148

v=0
o=- 434985978 434985978 IN IP4 47.100.107.46
s=Polycom IP Phone
c=IN IP4 47.100.107.66
t=0 0
m=audio 2226 RTP/AVP 0
a=rtpmap:0 PCMU/8000

[TCCE-TRACE]:1203000796.700 Msg-OUT          Call-OUT          SIP
Interface
    SIP Message PDU
ACK sip:3087771003@47.100.107.46:5075;transport=udp SIP/2.0
Via: SIP/2.0/UDP 47.100.107.65:5060;branch=z9hG4bK1851557632-56
Max-Forwards: 70
Authorization: Digest
username="3087771002",realm="Registered_Subscribers",nonce="ddf731ebf74
5
099ba6bf5af32aad03f6",uri="sip:3087771003@47.100.107.46",response="eaff
039ca9f85459557ecebc259
1d2e4",algorithm=MD5,opaque=""
From: <sip:3087771002@47.100.107.46>;tag=SR4134_824118656-53
To: <sip:3087771003@47.100.107.46>;tag=d7ad1506-1dd1-11b2-8292-
b03162323164+d7ad1506
Call-ID: 824118656-51
CSeq: 2 ACK
Contact: <sip:3087771002@47.100.107.65>
Content-Length: 0
```

4. Trace SIP messages for call disconnect between an analog phone and the head office phone:

```
debug ccsip messages
```

```
SR# debug ccsip messages

[TCCE-TRACE]:1203000806.10 Msg-IN           Call-OUT          SIP
Interface
   SIP Message PDU
BYE sip:3087771002@47.100.107.65 SIP/2.0
From: <sip:3087771003@47.100.107.46>;tag=d7ad1506-1dd1-11b2-8292-
b03162323164+d7ad1506
To: <sip:3087771002@47.100.107.46>;tag=SR4134_824118656-53
Call-ID: 824118656-51
CSeq: 1 BYE
Max-Forwards: 70
Supported: timer
Via: SIP/2.0/UDP 47.100.107.46:5075;branch=z9hG4bK1848684867177792
Content-Length: 0

[TCCE-TRACE]:1203000806.10 Msg-OUT          Call-OUT          SIP
Interface
   SIP Message PDU
SIP/2.0 200 OK
Via: SIP/2.0/UDP 47.100.107.46:5075;branch=z9hG4bK1848684867177792
From: <sip:3087771003@47.100.107.46>;tag=d7ad1506-1dd1-11b2-8292-
b03162323164+d7ad1506
To: <sip:3087771002@47.100.107.46>;tag=SR4134_824118656-53
Call-ID: 824118656-51
CSeq: 1 BYE
Contact: <sip:3087771002@47.100.107.65>
Supported: replaces,timer
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,OPTIONS,PRACK,INFO
Content-Length: 0
```