



V150.1 MER Modem Relay Support for TDM to SIP Gateway

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V150.1 MER Modem Relay Support for TDM to SIP Gateway

The Cisco V.150.1 Minimum Essential Requirements feature complies with the requirements of the National Security Agency (NSA) SCIP-216 Minimum Essential Requirements (MER) for V.150.1 recommendation. This feature is added to support V.150.1 MER modem relay support on SIP gateways. This is of importance to establish secure calls as per V.150.1 MER specifications between two SIP gateways with the PSTN line side being T1/E1 CAS or PRI. Typically, specialized encryption-capable BRI/analog phones (STE) that can communicate over V.150.1 modem relay or over modem pass-through are used as endpoints.

Feature Information for Cisco V.150.1 MER Modem Relay Support for TDM to SIP Gateway

The following table provides release information about the feature or features described in this module. This table lists only the software release that introduced support for a given feature in a given software release train. Unless noted otherwise, subsequent releases of that software release train also support that feature.

Use Cisco Feature Navigator to find information about platform support and Cisco software image support. To access Cisco Feature Navigator, go to www.cisco.com/go/cfn. An account on Cisco.com is not required.

Feature Name	Releases	Feature Information
V.150.1 MER Modem Relay Support for TDM to SIP Gateway	Cisco IOS 15.5(3)M	<p>The V.150.1 MER Modem Relay Support for TDM to SIP Gateway feature is added to support V.150.1 MER modem relay support on SIP gateways with the PSTN line side being T1/E1 CAS or PRI.</p> <p>The following command was modified: modem relay sse v150mer.</p>

Information on V150.1 MER Modem Relay Support for TDM to SIP Gateway

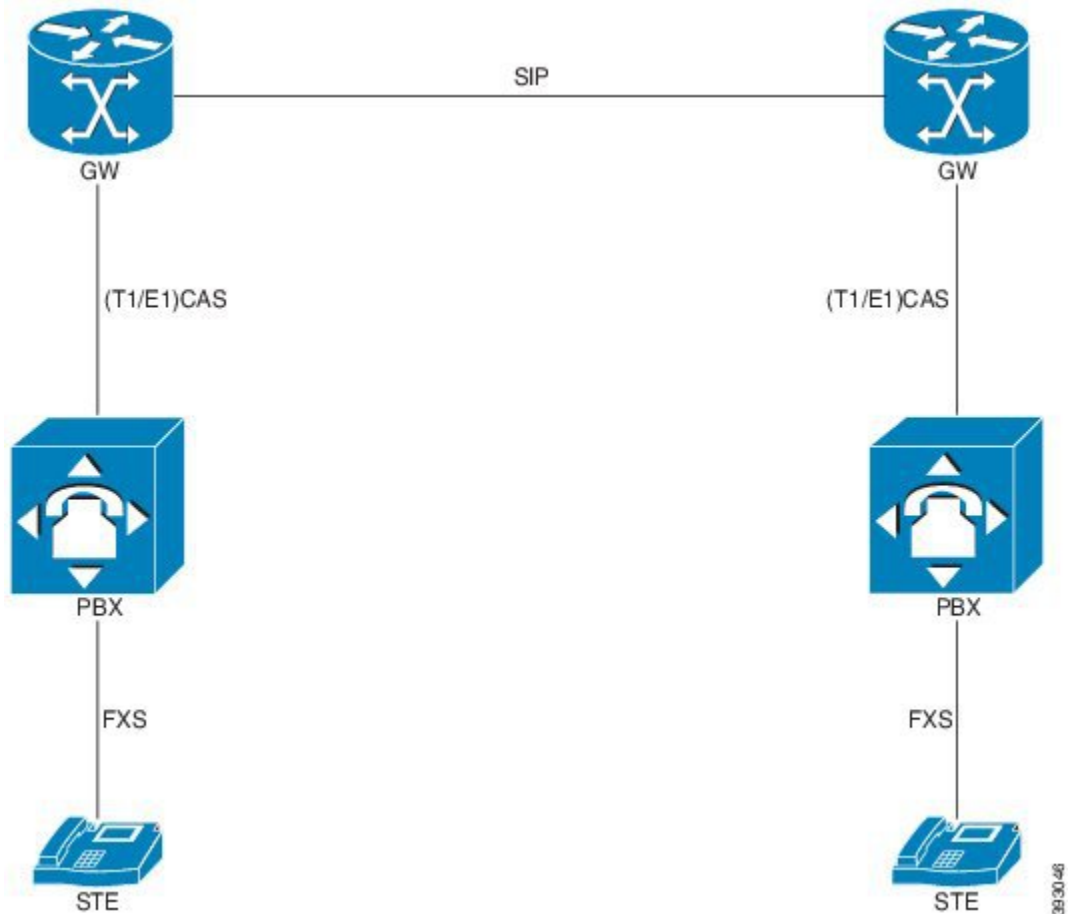
V150.1 MER Modem Relay Support for TDM to SIP Gateway

V.150.1 is an ITU standard for relaying modem or fax transmission that uses state signaling event (SSE) packets to trigger the transition from audio to modem-relay mode or fax-relay mode. The SSE headers are carried in the RTP packet. After call setup, in-band signaling through Simple Packet Relay Transport (SPRT) and SSE messages is used to transition from one state to another. SPRT packets (non-RTP packets) carry the data after the digital signal processor (DSP) transitions into modem relay mode.

In Cisco IOS Release 15.5(3)M, V150.1 MER Modem Relay support is provided on TDM to SIP gateways for placing secure calls between two SIP gateways with PSTN side being T1/E1 CAS or PRI. This feature provides only modem relay support as per the V.150.1 MER specifications.

Topology

Figure 1: Topology—V150.1 MER Modem Relay Support for TDM to SIP Gateway



Prerequisites for Cisco V.150.1 MER Modem Relay Support for TDM to SIP Gateway

- ISR-G2 routers must be installed on your network containing Cisco VWIC-xMFT-T1/E1 card for T1 PRI or CAS lines
- PVDM3 cards containing Dsp type: sp2600
- DTMF relay must be configured with RTP-NTE under dial-peer

Restrictions for Cisco V.150.1 MER Modem Relay Support for TDM to SIP Gateway

- V150.1 MER modem relay is supported only for TDM to SIP call flows
- V150.1 MER Modem relay is not supported with initial call being SRTP
- No-Audio codec and V150.1 MER compliance T.38 fax relay protocols are not supported
- VBD mode and other modulations like V.90 (apart from V.34 and V.32) are not supported
- SPRT with payload number 120 and SSE with payload number 118 values are fixed
- Audio to V150.1 MER modem relay transition is supported only for the following codecs: G.711, G.729, and G723
- Both NSE and SSE based modem relay should not be configured together

How to Configure V.150.1 MER Modem Relay Support for TDM to SIP Gateway

Configuring V.150.1 MER Modem Relay Support for TDM to SIP Gateway

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. Configure SSE MER V.150.1 using the following commands:
 - **modem relay sse v150mer** in the dial-peer configuration mode
 - **modem relay sse v150mer** in the global VoIP configuration mode
4. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	enable	Enables privileged EXEC mode.
Step 2	configure terminal	Enters global configuration mode.
Step 3	Configure SSE MER V.150.1 using the following commands: <ul style="list-style-type: none"> • modem relay sse v150mer in the dial-peer configuration mode 	

	Command or Action	Purpose
	<ul style="list-style-type: none"> • modem relay sse v150mer in the global VoIP configuration mode <p>Example:</p> <p>In dial-peer configuration mode</p> <pre>! Configuring SSE V.150.1 MER for one dial peer only Device (config)# dial-peer voice 20 voip Device (config-dial-peer)# modem relay sse v150mer Device (config-dial-peer)# end</pre> <p>Example:</p> <p>In global VoIP SIP configuration mode</p> <pre>! Configuring SSE MER V.1501.1 globally Device (config)# voice service voip Device (config-voi-serv)# modem relay sse v150mer Device (config-serv-serv)# end</pre>	
Step 4	end	Exits to privileged EXEC mode.

Verifying and Troubleshooting V.150.1 MER Modem Relay Support for TDM to SIP Gateway

To verify and troubleshoot the configuration of the Cisco V.150.1 MER Modem Relay Support for TDM to SIP feature, use the following **show** commands. The **show** commands provide information about the calls.

1. **show call active voice {brief/compact}**-This command displays details about the VoIP call legs on the voice gateway, and also the type of codec and modem relay (V.150.1 or NSE) used. Initially, when the call is audio, the command displays details about the audio codec. Once the modem call is established across gateway, the codec is displayed as modem relay.

Example

```
Router# show call active voice compact
<callID> A/O FAX T<sec> Codec type Peer Address IP R<ip>:<udp>
Total call-legs: 2
      38 ANS T20 modem-relay VOIP P 9.44.48.116:16422
      39 ORG T20 modem-relay TELE P2357
```

Example

```
Router# show call active voice | sec Modem
Modem Relay Mode = signaling-assisted
Modem Relay Local Rx Speed=31200 bps
Modem Relay Local Tx Speed=31200 bps
Modem Relay Remote Rx Speed=31200 bps
Modem Relay Remote Tx Speed=31200 bps
Modem Relay Phy Layer Protocol=v34
Modem Relay Ec Layer Protocol=v14
Modem RelayType=sse-v150-mer
```

2. **show modem relay statistics {sprt/all/v42}**-This command provides information on the various statistics for modem relay. The SPRT packets will increase for all the V.150.1 MER modem relay calls. Hence, this command is used to make sure that the call is using the V.150.1 MER modem relay method.

Example

```
Router# show modem relay statistics sprt
ID:11E4
SPRT Layer Statistics
  sprt_info_frames_rcvd=0 sprt_xid_frames_rcvd=0
  sprt_tc0_explicit_acks_rcvd=0 sprt_tcl_explicit_acks_rcvd=0
-----
-----
  sprt_info_tframes_failed_to_consume=0
  sprt_info_bytes_rcvd=1637 sprt_info_bytes_sent=47
  sprt_pkts_dropped_intf_busy=0 sprt_min_rexmit_timeout=500
  sprt_max_rexmit_timeout=500

Total Modem Relay Call Legs = 1
```

Troubleshooting V.150.1 MER Modem Relay Support for TDM to SIP Gateway

Use the following **debug** commands to troubleshoot V.150.1 MER modem relay support for TDM to SIP gateway:

- debug ccsip message
- debug ccsip error
- debug voip ccapi inout
- debug voip vtsp all
- debug vpm signal
- debug modem relay events
- debug modem relay errors

Use the following advanced **debug** commands to troubleshoot V.150.1 MER modem relay support for TDM to SIP gateway thoroughly:

- debug voip ccapi inout
- debug vpm signal
- debug voip vtsp all
- debug voip hpi all
- debug modem relay errors
- debug modem relay events
- debug modem relay packetizer
- debug voice dsm all
- debug voice dsmp all

- debug voip rtp session named-event
- debug voip dspapi all
- debug voip ccapi all
- debug ccsip verbose

