

Interworking of Secure RTP calls for SIP and H.323

The Session Initiation Protocol (SIP) support for the Secure Real-time Transport Protocol (SRTP) is an extension of the Real-time Transport Protocol (RTP) Audio/Video Profile (AVP) and ensures the integrity of RTP and Real-Time Control Protocol (RTCP) packets that provide authentication, encryption, and the integrity of media packets between SIP endpoints.

SIP support for SRTP was introduced in Cisco IOS Release 12.4(15)T. In this and later releases, you can configure the handling of secure RTP calls on both a global level and on an individual dial peer basis on Cisco IOS voice gateways. You can also configure the gateway (or dial peer) either to fall back to (nonsecure) RTP or to reject (fail) the call for cases where an endpoint does not support SRTP.

The option to allow negotiation between SRTP and RTP endpoints was added for Cisco IOS Release 12.4(20)T and later releases, as was interoperability of SIP support for SRTP on Cisco IOS voice gateways with Cisco Unified Communications Manager. In Cisco IOS Release 12.4(22)T and later releases, you can also configure SIP support for SRTP on Cisco Unified Border Elements (Cisco UBEs).

Prerequisites

The following are prerequisites for the Interworking of Secure RTP calls for SIP and H.323 feature:

- Establish a working IP network and configure VoIP.



Note

For information about configuring VoIP, see [Enhancements to the Session Initiation Protocol for VoIP on Cisco Access Platforms](#).

- Ensure that the gateway has voice functionality configured for SIP.
- Ensure that your Cisco router has adequate memory.
- As necessary, configure the router to use Greenwich Mean Time (GMT). SIP requires that all times be sent in GMT. SIP INVITE messages are sent in GMT. However, the default for routers is to use Coordinated Universal Time (UTC). To configure the router to use GMT, issue the clock timezone command in global configuration mode and specify GMT.

Cisco Unified Border Element

- Cisco IOS Release 12.2(20)T or a later release must be installed and running on your Cisco Unified Border Element.

Cisco Unified Border Element (Enterprise)

- Cisco IOS XE Release 3.1S or a later release must be installed and running on your Cisco ASR 1000 Series Router.

Restrictions

- The SIP gateway does not support codecs other than those listed in the table titled “SIP Codec Support by Platform and Cisco IOS Release” in the “Enhanced Codec Support for SIP Using Dynamic Payloads” section of the [Configuring SIP QoS Features](#) module.
- SIP requires that all times be sent in GMT.