SRTP-SRTP Pass-Through

Overview

SRTP-SRTP pass-through feature allows pass-through of encrypted media from one call-leg to the other.

Cisco Unified Border Element (CUBE) supports SIP calls between endpoints using Transport Layer Security (TLS) for SIP signaling encryption and Secure Real-Time Protocol (SRTP) to provide RTP media encryption. However, these two encryption mechanisms may not be deployed simultaneously, depending on the required call flow invoked on the associated configuration.

The following are conditions of the SRTP Passthrough feature:

- SRTP Passthrough must be configured on both legs of the call. If the target adjacency does not support SRTP Passthrough, then the call is rejected by error message 415 (Unsupported Media Type).
- "m= .. RTP/SAVP .." and a="crypto:..." fields coming in on an Invite from one adjacency are passed on in an Invite to the target adjacency.
- "m= ...RTP/SAVP..." is a required field in the Invite to trigger SRTP Passthrough behavior in the CUBE.

Pass-Through of Unsupported Crypto Suites

Note Effective from Cisco IOS XE Everest Release 16.5.1b, CUBE supports AEAD_AES_128_GCM and AEAD_AES_256_GCM crypto-suites. For more information, see SRTP-SRTP Interworking.

CUBE supports transparent passthrough of all (supported and unsupported) crypto suites.

CUBE has the ability to pass across crypto attributes (containing any unsupported crypto suites) as well as media packets (encrypted with unsupported crypto suites).

If SRTP pass-thru feature is enabled, media interworking will not be supported. Ensure that you have symmetric configuration on both the incoming and outgoing dial-peers to avoid media-related issues.

Feature Information

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.

Table 1: Feature Information for SRTP-SRTP Pass-Through

Feature Name	Releases	Feature Information
Support for SRTP-SRTP Basic calls	Baseline functionality	This feature introduced support for basic SRTP-SRTP pass-through calls.

Configure Pass-Through of Unsupported Crypto Suites for a Specific Dial Peer

SUMMARY STEPS

1. **enable**
2. configure terminal
3. **dial-peer voice** *tag* **voip**
4. **destination-pattern** *string*
5. **session protocol sipv2**
6. **sessiontarget ipv4:** *destination-address*
7. **incoming called-number** *string*
8. **srtp pass-thru**
9. **codec** *codec*
10. **end**
11. **dial-peer voice** *tag* **voip**
12. Repeat Steps 4, 5, 6, and 7 to configure a second dial peer.
13. **srtp pass-thru**
14. **codec** *codec*
15. **exit**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Device> enable`	Enables privileged EXEC mode. • Enter your password if prompted.

	Command or Action	Purpose
Step 2	configure terminal **Example:** `Device# configure terminal`	Enters global configuration mode.

Step 3	**dial-peer voice** *tag* **voip**
	Example:
	`Device(config)# dial-peer voice 201 voip`
		Defines a particular dial peer, to specify the method of voice encapsulation, and enters dial peer voice configuration mode.
		• In the example, the following parameters are set:
		  • Dial peer 201 is defined.
		  • VoIP is shown as the method of encapsulation.

Step 4	**destination-pattern** *string* **Example:** `Device(config-dial-peer)# destination-pattern 5550111`	Specifies either the prefix or the full E.164 telephone number to be used for a dial peer string. • In the example, 5550111 is specified as the pattern for the telephone number.
Step 5	**session protocol sipv2** **Example:** `Device(config-dial-peer)# session protocol sipv2`	Specifies a session protocol for calls between local and remote routers using the packet network. • In the example, the **sipv2** keyword is configured so that the dial peer uses the IETF SIP.
Step 6	**sessiontarget ipv4:** *destination-address* **Example:** `Device(config-dial-peer)# session target ipv4:10.13.25.102`	Designates a network-specific address to receive calls from a VoIP or VoIPv6 dial peer. • In the example, the IP address of the dial peer to receive calls is configured as 10.13.25.102.
Step 7	**incoming called-number** *string* **Example:** `Device(config-dial-peer)# incoming called-number 5550111`	Specifies a digit string that can be matched by an incoming call to associate the call with a dial peer. • In the example, 5550111 is specified as the pattern for the E.164 or private dialing plan telephone number.
Step 8	**srtp pass-thru** **Example:** `Device(config-dial-peer)# srtp pass-thru`	Enables transparent passthrough of all crypto suites for a specific dial peer.
Step 9	**codec** *codec* **Example:** `Device(config-dial-peer)# codec g711ulaw`	Specifies the voice coder rate of speech for the dial peer. • In the example, G.711 mu-law at 64,000 bps, is specified as the voice coder rate for speech.

	Command or Action	Purpose
Step 10	**end** **Example:** `Device(config-dial-peer)#end`	Exits dial peer voice configuration mode.

Step 11	**dial-peer voice** *tag* **voip**
	Example:
	`Device(config)# dial-peer voice 200 voip`
		Defines a particular dial peer, to specify the method of voice encapsulation, and enters dial peer voice configuration mode.
		• In the example, the following parameters are set:
		  • Dial peer 200 is defined.
		  • VoIP is shown as the method of encapsulation.

Step 12	Repeat Steps 4, 5, 6, and 7 to configure a second dial peer.	--
Step 13	**srtp pass-thru** **Example:** `Device(config-dial-peer)# srtp pass-thru`	Enables transparent passthrough of all crypto suites for a specific dial peer.
Step 14	**codec** *codec* **Example:** `Device(config-dial-peer)# codec g711ulaw`	Specifies the voice coder rate of speech for the dial peer. • In the example, G.711 mu-law at 64,000 bps, is specified as the voice coder rate for speech.
Step 15	**exit** **Example:** `Device(config-dial-peer)# exit`	Exits dial peer voice configuration mode.

Configure Pass-Through of Unsupported Crypto Suites Globally

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **srtp pass-thru**
5. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable**	Enables privileged EXEC mode.

	Command or Action	Purpose
	Example: `Device> enable`	• Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Device# configure terminal`	Enters global configuration mode.
Step 3	**voice service voip** **Example:** `Device(config)# voice service voip`	Enters VoIP voice-service configuration mode.
Step 4	**srtp pass-thru** **Example:** `Device(config-dial-peer)# srtp pass-thru`	Enables transparent passthrough of all crypto suites globally.
Step 5	**end** **Example:** `Device(config-dial-peer)#end`	Exits dial peer voice configuration mode.

Configuration Examples for SRTP-SRTP Pass-Through

Example for SRTP=SRTP Pass-Through

```
enable
configure terminal
dial-peer voice 201 voip
destination-pattern 5550111
session protocol sipv2
session target ipv4:10.13.25.102
incoming called-number 5550111
srtp
codec g711ulaw
end

dial-peer voice 200 voip
destination-pattern 5550111
session protocol sipv2
session target ipv4:10.13.25.101
incoming called-number 5550111
srtp
codec g711ulaw
end
```

Example for Pass-Through of Unsupported Crypto Suites for a specific dial peer

```
enable
configure terminal
dial-peer voice 201 voip
destination-pattern 5550111
session protocol sipv2
session target ipv4:10.13.25.102
incoming called-number 5550111
srtp pass-thru
codec g711ulaw
end

dial-peer voice 200 voip
destination-pattern 5550111
session protocol sipv2
session target ipv4:10.13.25.101
incoming called-number 5550111
srtp pass-thru
codec g711ulaw
end
```

Example for Pass-Through of Unsupported Crypto Suites Globally

```
enable
configure terminal
voice service voip
srtp pass-thru
end
```