Configure Multiple Trunks Using Tenants

Overview

The CUBE Tenant feature allows you to configure SIP trunks individually using parameters that were previously only available globally, or with individual dial-peers. Tenants act as a configuration template for dial-peers, which allow you to customize the global configuration to suit the requirements for each trunk. Dial-peers associated with a tenant automatically receive all of its configuration, making trunk configuration simple and consistent. If necessary, specific configurations may be overridden at the dial-peer level, allowing maximum flexibility.

When bound to an interface configured with a VRF, the tenant feature may also be used to configure trunks for multiple customers, each with their own characteristics on the same platform.

The **voice class tenant** <*tag*> command allows sip-specific attributes to be configured for each trunk. The command **voice class tenant** <*tag*> can then be used to apply the tenant configuration to individual dial-peers. Refer to "Table 1: Multi-Tenant Configuration List, on page 2" for information on the complete list of configurations present under the **voice class tenant** <*tag*>.

If tenants are configured under dial-peer, then configurations are applied in the following order of preference.

- Dial-peer configuration
- Tenant configuration
- Global configuration

That is, if the value of the attribute under dial-peer configuration is system, then the value is taken from the tenant configuration. And, if the value under the tenant configuration is also system, then the global configuration is used.

If there are no tenants configured under dial-peer, then the configurations are applied using the default behavior in the following order:

- Dial-peer configuration
- Global configuration

The following table lists the various configurations present under **voice class tenant** <*tag*>. For more information on specific configurations, see the Voice and Video command reference guide lists.

Note Attributes that are not available under **voice class tenant** <*tag*> use the default behavior—With preference of dial-peer followed by the global configuration.

Table 1: Multi-Tenant Configuration List

Command	Description
aaa	SIP-UA AAA related configuration
anat	Allow alternative network address types IPv4 and IPv6
asserted-id	Configure SIP UA privacy identity settings
associate	Associate a RCB for outgoing calls
asymmetric	Configure global SIP asymmetric payload support
authentication	Digest Authentication Configuration
bandwidth	Allow SIP SDP bandwidth-related options
bind	SIP bind command
block	Block 18X response to INVITE
call-route	Configure call routing options
conn-reuse	Reuse the sip registration tcp connection for the end-point behind a Firewall
connection-reuse	Use listener port for sending requests over UDP
contact-passing	302 contact to be passed through for CFWD
content	Content carried as part of SIP message
copy-list	Configure list of entities to be sent to peer leg
credentials	User credentials for registration
disable-early-media	Disable early-media cut through
dns -a-override	Skip DNS A/AAAA query when SRV query timesout
dscp -profile	DSCP Profile global config
early-media	Configure method to handle early-media Update Request
early-offer	Configure sending Early-Offer

Command	Description
encap	Configure SDP encapsulation
error-code-override	Configure sip error code
error- passthru	SIP error response pass-thru functionality
exit	Exits from the voice class configuration mode
g729	G729 codec interoperability settings
handle-replaces	Handle INVITE with REPLACES header at SIP spi
header-passing	SIP Headers need to be passed to applications
help	Description of the interactive help system
history-info	History Info header support
host-registrar	Use sip-ua registrar value in Diversion and Contact header for 3xx messages
interop-handling	Enable interop-handling
localhost	Specify the DNS name for the localhost
map	Mapping options
max-forwards	Change number of max-forwards for SIP Methods
midcall -signaling	Configure method to handle mid-call signaling
nat	SIP nat global config
no	Negate a command or set its defaults
notify	SIP Signaling Notify Configuration
offer	Configure settings for Offers made from the Gateway
options-ping	Send OPTION pings to remote end
outbound-proxy	Configure an Outbound Proxy Server
pass-thru	SIP pass-through global config
permit	Permit hostname for this gateway
preloaded-route	Use pre-loaded route header for outgoing calls, if available
privacy	Configure SIP UA privacy settings
privacy-policy	Set privacy behavior for outgoing SIP messages

random-contact	Use Random Contact for outgoing calls, if available
random-request- uri	Configure options for Request-URI having random value
reason-header	Configure settings for supporting SIP Reason Header
redirection	Enable call redirection (3xx) handling
refer- ood	Configure maximum number of out-of-dialog refer made to the Gateway
referto -passing	Refer-To needs to be passed through for transfer
registrar	Configure SIP registrar VoIP Interface
registration	Enable registration options
rel1xx	Type of reliable provisional response support
remote-party-id	Enable Remote-Party-ID support in SIP User Agent
requri -passing	Request URI needs to be passed through
reset	SIP Reset Options
retry	Change default retries for each SIP Method
send	Configure outgoing message options
session	SIP Voice Protocol session config
sip-profiles	SIP Profiles global config
sip-server	Configure a SIP Server Interface
srtp	Allow SIP related SRTP options
srtp-auth	Allow to set preferred suites
tel-config	Tel format cfg for headers other than req -line in
timers	SIP Signaling Timers Configuration
update- callerid	Enable sending updates for callerid
url	Url configuration for request-line url in outgoing INVITE
video	Video related config for sip
warn-header	SIP Warning-Header global config

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 2: Feature Information

Feature Name	Releases	Feature Information
Support for Configuring Multi Tenants on SIP Trunk	Cisco IOS 15.6(2)T Cisco IOS XE Denali 16.3.1	This feature allows the provision to configure specific global configurations for multiple tenants on SIP trunks. The following commands were introduced: **voice class tenant** *tag* and **voice-class sip tenant** *tag* .

Feature Characteristics of Configurable SIP Trunk Listen Port

- For Cisco IOS XE Cupertino 17.8.1a and later releases, you can also configure a listen-port at the tenant level. Before this release, you could configure the listen-port only at the global configuration level.
- Multiple inbound TLS, TCP, or UDP connections can be established using different IP ports. Each port is mapped to a tenant trunk configuration, which may have its own TLS profile validation criteria.
- A tenant listen port may only be configured when there are no active calls on associated dial-peers.
- Tenant level listen-port configuration is supported for both secure (TLS) and nonsecure (TCP/UDP) transport types.
- Interface binding must be configured for a tenant to use a SIP trunk listen port.
- IPv4 and IPv6 listen ports may be configured for TLS, TCP or UDP transport types.
- The listen-port along with the bind interface must be unique across all:
  - Global and tenant level configuration modes
  - Secure and nonsecure ports
- If you modify the interface to which a tenant is bound, the existing listen-port will be closed and re-opened with the latest interface details.
- When there is a configuration change at the **bind** or **tenant level**, all the associated active connections are closed.
- The nonsecure listen-port range is limited to 5000 - 5500 to avoid overlap with the RTP port range, especially for UDP.

- Connections get segregated at the tenant level during inbound dial-peer matching. For this, the tenant tag in the inbound dial-peer is matched with the tenant tag that is identified during connection establishment.

To use the SIP trunk listen port feature, must configure the associated tenant with a SIP listen port:

- **tls-profile** <*tag*> under **voice class tenant** *tag* configuration mode.

For more information on the CLI commands, see Cisco IOS Voice Command Reference Guide.

Feature Characteristics of Trunk Specific TLS Policy

- For TLS connections, the trustpoint selection is as follows:
  - The trustpoint is selected based on tenant configuration.
  - If this is not available, then the remote-IP or global configurations are used.

Note Except for the CN-SAN certificate validation, CUBE retains the same behavior for inbound nonsecure connections (TCP and UDP transport types).

To use a trunk specific TLS policy, you must configure the associated tenant with a TLS policy:

- **listen-port** {**non-secure** *port-number* | **secure** *port-number*} under **voice class tenant** *tag* configuration mode.

For more information on the CLI commands, see Cisco IOS Voice Command Reference Guide.

Configure SIP Trunks using Voice Class Tenant

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. Use the following to configure trunks using the tenant feature:
   - **voice class tenant <tag>** in the global configuration mode

     Once you configure the **voice class tenant <tag>** command in the global mode, the configuration will move to the **voice class tenant <tag>** submode. You can configure all the sip-specific attributes in this submode.
   - **voice-class sip tenant <tag>** in the dial-peer configuration mode
4. **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	Use the following to configure trunks using the tenant feature:
	• **voice class tenant <tag>** in the global configuration mode
	Once you configure the **voice class tenant <tag>** command in the global mode, the configuration will move to the **voice class tenant <tag>** submode. You can configure all the sip-specific attributes in this submode.
	• **voice-class sip tenant <tag>** in the dial-peer configuration mode
	Example:
	In global configuration mode
	
	`! Configuring tenant 1`
	`Device(config)# voice class tenant 1`
	`Device (config-class)# ?`
	`aaa – sip-ua AAA related configuration`
	`anat – Allow alternative network address types IPV4 and IPV6`
	`asserted-id – Configure SIP-UA privacy identity settings`
	`……`
	`……`
	`……`
	`Video – video related function`
	`Warn-header – SIP related config for SIP. SIP warning-header global config.`
	`Device (config-voi-tenant)# end`
	`--------`
	`! Configuring tenant 2`
	`Device(config)# voice class tenant 2`
	`Device (config-class)# ?`
	`aaa – sip-ua AAA related configuration`
	`anat – Allow alternative network address types IPV4 and IPV6`
	`asserted-id – Configure SIP-UA privacy identity settings`
	`……`
	`……`
	`outbound-proxy - Configure an Outbound Proxy Server`
	`pass-thru  - SIP pass-through global config`
	`……`
	`……`
		Use the **voice-class sip tenant <tag>** command in the global configuration mode to configure a tenant with sip-specific attributes. This command tag can then be applied to one or more dial-peers using the **voice-class sip tenant <tag>** command under the dial-peers.

	Command or Action	Purpose

	`srtp - Allow SIP related SRTP options`
	`Warn-header – SIP related config for SIP. SIP warning-header global config.`
	`Device (config-voi-tenant)# end`
	
	Example:
	
	In dial-peer configuration mode
	
	`!Configuring tenant 1 under dial-peer 10`
	`Device (config)# dial-peer voice 10 voip`
	`Device (config-dial-peer)# voice-class sip tenant 1`
	`Device (config-dial-peer)# end`
	`-------------`
	`!Configuring tenant 2 under dial-peer 20`
	`Device (config)# dial-peer voice 20 voip`
	`Device (config-dial-peer)# voice-class sip tenant 2`
	`Device (config-dial-peer)# end`
	
	`!An example for the use of the "no" form of command voice-class sip tenant`
	`Router(config)# dial-peer voice 3000 voip`
	`Router(config-dial-peer)# voice-class sip tenant 1`
	`Router(config-dial-peer)# no voice-class sip tenant 1`
	
	When the **no** form is configured, the dial-peer is no longer associated with the tenant tag configuration. The attributes are now applied using the default order of dial-peer followed by the global configuration.	

Step 4	**end** **Example:** `Device(config-dial-peer)# end`	Returns to privileged EXEC mode.

Example: Multiple Trunks using Registration with Tenants

Trunk registration details may also be included in a tenant configuration, allowing a platform to register to multiple registrars concurrently. Tenants configured with registration details do not need to be associated with a dial-peer for the registration process to start.

```
Router# show run | sec tenant

Voice class tenant 1
registrar 1 ipv4:10.64.86.35:9051 expires 3600
credentials username aaaa password 7 06070E204D realm aaaa.com
outbound-proxy ipv4:10.64.86.35:9057
bind control source-interface GigabitEthernet0/0

Voice class tenant 2
registrar 1 ipv4:9.65.75.45:9052 expires 3600
credentials username bbbb password 7 110B1B0715 realm bbbb.com
```

```
outbound-proxy ipv4:10.64.86.40:9040
bind control source-interface GigabitEthernet0/1
```

For multi-tenancy support on Cisco Unified Border Element, you can configure voice class tenants with different credentials, but having the same registrar. In that scenario, it is recommended that you configure the CLI commands **sip-server** and **registrar** under **voice class tenant** configuration. The following is a sample configuration:

```
voice class tenant 1
 credentials number 1111 username test password 7 071B245B5D1D realm ipvoice.jp
 authentication username test password 7 06120A3258
 registrar ipv4:1.1.1.1 expires 120
 sip-server ipv4:1.1.1.1
!
voice class tenant 2
 credentials number 2222 username test password 7 09584B1E0A11 realm ipvoice.jp
 authentication username test2 password 7 071B245F5A
 registrar ipv4:1.1.1.1 expires 120
 sip-server ipv4:1.1.1.1
```