SIP Devices Configuration

Set Up Ingress Gateway to Use Redundant Proxy Servers

Configure the gateway with the following code to send calls to redundant proxy servers as resolved using DNS SRV lookup:

```
ip domain name <your domain name>
ip name-server <your DNS server>
sip-ua
sip-server dns:<your SRV cluster domain name>
dial-peer voice 1000 voip
session target sip-server
```

Set Up Call Server with Redundant Proxy Servers

Use redundant proxy servers for Unified CVP outbound calls by using a DNS-based SRV cluster name or a non-DNS SRV cluster name (also known as Server Group Name).

See the *Operations Console User's Guide for Cisco Unified Customer Voice Portal* on how to configure local based SRV records.

Local SRV File Configuration Example for SIP Messaging Redundancy

Load-Balancing SIP Calls

SIP calls can be load balanced across destinations in several different ways as outlined below:

- Using the CUSP server, define several static routes with the same route pattern, priorities, and weights.
- Using DNS, configure SRV records with priorities and weights. Both the DNS client and the server settings must be configured and operating successfully for DNS "A" and "SRV" type queries to work. Configure SRV queries to be used wherever outbound SIP calls are made, such as on the IOS Ingress gateway, on the Call Server itself, and on Unified CM.

Note Refer to DNS Zone File Configuration for Comprehensive Call Flow Model for information about load balancing and failover without a Proxy Server. Only the DNS SRV method is supported for load balancing and failover without a Proxy Server.

Related Topics

DNS Zone File Configuration for Comprehensive Call Flow Model

Cisco Unified SIP Proxy (CUSP) Configuration

The following configuration shows a CUSP proxy in Unified CVP. The highlighted lines are specific to a Unified CVP solution. For additional configuration details, refer to the Configuring Cisco Unified SIP Proxy Server guide.

Configuration Example:

```
server-group sip global-load-balance call-id
                            server-group sip retry-after 0
                            server-group sip element-retries udp 1
                            server-group sip element-retries tls 1
                            server-group sip element-retries tcp 1
                            sip dns-srv
                            no enable
                            no naptr
                            end dns
                        !
                        no sip header-compaction
                        no sip logging
                        !
                        sip max-forwards 70
                        sip network netA noicmp
                        non-invite-provisional 200
                        allow-connections
                        retransmit-count invite-server-transaction 9
                        retransmit-count non-invite-client-transaction 9
```

```
retransmit-count invite-client-transaction 2
retransmit-timer T4 5000
retransmit-timer T2 4000
retransmit-timer T1 500
retransmit-timer TU2 32000
retransmit-timer TU1 5000
retransmit-timer clientTn 64000
retransmit-timer serverTn 64000
end network
!
no sip peg-counting
!
sip privacy service
sip queue message
drop-policy head
low-threshold 80
size 2000
thread-count 20
end queue
!
sip queue radius
drop-policy head
low-threshold 80
size 2000
thread-count 20
end queue
!
sip queue request
drop-policy head
low-threshold 80
size 2000
thread-count 20
end queue
!
sip queue response
drop-policy head
low-threshold 80
size 2000
thread-count 20
end queue
!
sip queue st-callback
drop-policy head
low-threshold 80
size 2000
thread-count 10
end queue
!
sip queue timer
drop-policy none
low-threshold 80
size 2500
thread-count 8
end queue
!
sip queue xcl
drop-policy head
low-threshold 80
size 2000
thread-count 2
end queue
!
route recursion
!
```

```
sip tcp connection-timeout 240
sip tcp max-connections 256
!
no sip tls
!
trigger condition in-netA
    sequence 1
    in-network netA
    end sequence
    end trigger condition
    !
    trigger condition mid-dialog
    sequence 1
    mid-dialog
    end sequence
    end trigger condition
    !
    trigger condition out-netA
    sequence 1
    out-network netA
    end sequence
    end trigger condition
!
accounting
no enable
no client-side
no server-side
end accounting
!
server-group sip group cucm-cluster.cisco.com netA
 element ip-address 10.86.129.219 5060 udp q-value 1.0 weight 10
 element ip-address 10.86.129.62 5060 udp q-value 1.0 weight 10
 element ip-address 10.86.129.63 5060 udp q-value 1.0 weight 10
 failover-resp-codes 503
 lbtype global
 ping
 end server-group
 !
 server-group sip group cvp-call-servers.cisco.com netA
 element ip-address 10.86.129.220 5060 udp q-value 1.0 weight 10
 element ip-address 10.86.129.224 5060 udp q-value 0.9 weight 10
 failover-resp-codes 503
 lbtype global
 ping
 end server-group
 !
 server-group sip group vxml-gws.cisco.com netA
 element ip-address 10.86.129.229 5060 udp q-value 1.0 weight 10
 element ip-address 10.86.129.228 5060 udp q-value 1.0 weight 10
 failover-resp-codes 503
 lbtype global
 ping
 end server-group
 !
 route table cvp-route-table
 key 9 target-destination vxml-gws.cisco.com netA
 key 8 target-destination cvp-call-servers.cisco.com netA
 key 7 target-destination vxml-gws.cisco.com netA
 key 700699 target-destination cvp-call-servers.cisco.com netA
 key 2 target-destination cucm-cluster.cisco.com netA
 key 1 target-destination cucm-cluster.cisco.com netA
 key 7000 target-destination 172.19.151.41 netA
 key 777333 target-destination cvp-call-servers.cisco.com netA
 key 1004 target-destination 10.86.139.84 netA
```

```
 key 7105 target-destination dialer-gws netA
 end route table
 !
 policy lookup cvp-policy
 sequence 1 cvp-route-table request-uri uri-component user
 rule prefix
 end sequence
 end policy
 !
 trigger routing sequence 1 by-pass condition mid-dialog
 trigger routing sequence 10 policy cvp-policy condition in-netA
 !
 server-group sip ping-options netA 10.86.129.200 5038
 method OPTIONS
 ping-type adaptive 5000 10000
 timeout 500
 end ping
 !
 server-group sip global-ping
 sip listen netA udp 10.86.129.200 5060
!
end
```

Configure Custom Streaming Ringtones

You can configure custom ringtone patterns that enable you to play an audio stream to a caller in place of the usual ringtone. Customized streaming ringtones are based on the dialed number destination and, when configured, play an in-progress broadcast stream to the caller while the call is transferred an agent.

Procedure

Step 1 Configure Helix for streaming audio.

The default installation and configuration of the Helix server is all that is required for use with Unified CVP. See the *Helix Server Administration Guide* for information about installing and configuring the Helix Server.

Step 2 In the Operations Console, perform the following steps to configure custom streaming ringtones:

a) Select **System** > **Dialed Number Pattern** .
b) Click **Add New**.
c) Complete the following fields to assoicate a dialed number pattern with a custom ringtone.

Table 1: Dialed Number Pattern Configuration Settings

Property	Description	Default	Value
General Configuration			

Property	Description	Default	Value
Dialed Number Pattern	The actual Dialed Number Pattern.	None	Must be unique Maximum length of 24 characters Can contain alphanumeric characters, wildcard characters such as exclamation point (!) or asterisk (*), single digit matches such as the letter X or period (.) Can end with an optional greater than (>) wildcard character
Description	Information about the Dialed Number Pattern.	None	Maximum length of 1024 characters
Enable Custom Ringtone	Enables customized ring tone. • **Ringtone media filename -** Enter the name of the file that is to be played for the respective dialed number pattern. Provide the URL for the stream name in the following format: `rtsp://<streaming server IP address>:<port>/<directory>/<filename>.rm`	Disabled none	Maximum length of 256 characters Cannot contain whitespace characters

d) Click **Save** to save the Dialed Number Pattern.

You are returned to the **Dialed Number Pattern** page. To deploy the Dialed Number Pattern configuration, click **Deploy** to deploy the configuration to all Unified CVP Call Server devices.

e) Access the IOS device in global configuration mode and add the following commands on your VXML Gateway:

```
rtsp client timeout 10

rtsp message timeout 10
```

The range is 1 to 20; the recommended value is 10 seconds.

Step 3 Add a Send to VRU node in your ICM script before any Queue node.

The explicit Send to VRU node is used to establish the VRU leg before the transfer to the agent; this is required to play streaming audio ringtones to a caller.