NanoCUBE -- Emergency Number Preemption

The emergency number preemption (also called 911-preemption) feature enables you to configure a list of emergency numbers. When the maximum number of incoming or outgoing connections on a dial-peer is reached, the other non-emergency calls are preempted from the session initiation protocol (SIP) dial-peer, allowing the emergency call to go through.

Finding Feature Information

Your software release may not support all the features documented in this module. For the latest caveats and feature information, see Bug Search Tool and the release notes for your platform and software release. To find information about the features documented in this module, and to see a list of the releases in which each feature is supported, see the feature information table.

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.

Restrictions for Emergency Number Preemption

The Emergency Number Preemption feature works only for maximum-connections-based call admission control (CAC) and not bandwidth-based CAC.

Information About Emergency Number Preemption

The emergency number is 911 in the U.S.A., but it may not be the same in other parts of the world; so this emergency number can configured by the user. This feature is triggered by the following two conditions:

- When the SIP dial peer connection reaches the maximum connections configured.
- When the current call's called number matches with the emergency number.

When the emergency preemption feature is triggered, the non-emergency call is preempted and the current emergency call is allowed to go through. If it cannot find another non-emergency call to preempt, the current emergency call will fail.

Note Using this feature, the existing multi-level precedence preemption (MLPP) feature is leveraged, as MLPP can preempt calls.

How to Configure Emergency Number Preemption

Configuring the Emergency Number

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice service voip**
4. **emergency** *number*
5. **end**

DETAILED STEPS

	Command or Action	Purpose
Step 1	**enable** **Example:** `Device> enable`	Enables privileged EXEC mode. • Enter your password if prompted.
Step 2	**configure terminal** **Example:** `Device# configure terminal`	Enters global configuration mode.

	Command or Action	Purpose
Step 3	**voice service voip** **Example:** `Device(config)# voice service voip`	Enters voice service VoIP configuration mode.
Step 4	**emergency** *number* **Example:** `Device(conf-voi-serv)# emergency 912 913`	Configures the list of emergency numbers. You can set multiple emergency numbers, which will be set to highest priority if MLPP is configured.
Step 5	**end** **Example:** `Device(conf-voi-serv)# end`	Returns to privileged EXEC mode.

Configuring Preemption and the Maximum Connections on SIP Dial Peer

SUMMARY STEPS

1. **enable**
2. **configure terminal**
3. **voice mlpp**
4. **preemption voip-trunk sip**
5. **exit**
6. **dial-peer voice** *tag* **voip**
7. **max-conn** *number*
8. **end**

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	**voice mlpp** **Example:** `Device(config)# voice mlpp`	Enters voice MLPP configuration mode that is used for MLPP commands.
Step 4	**preemption voip-trunk sip** **Example:** `Device(config-voice-mlpp)# preemption voip-trunk sip`	Enables preemption for VoIP trunk SIP dial peer.
Step 5	**exit** **Example:** `Device(config-voice-mlpp)# exit`	Exits voice MLPP configuration mode and returns to privileged EXEC mode.
Step 6	**dial-peer voice** *tag* **voip** **Example:** `Device(config)# dial-peer voice 30 voip`	Enters dial peer voice configuration mode.
Step 7	**max-conn** *number* **Example:** `Device(config-dial-peer)# max-conn 2`	Configures the maximum number of connections on SIP dial peers.
Step 8	**end** **Example:** `Device(config-dial-peer)# end`	Returns to privileged EXEC mode.

Verifying Emergency Number Preemption

Perform this task to verify the configuration for the emergency number preemption. The **show** commands can be entered in any order.

SUMMARY STEPS

1. **enable**
2. **show dial-peer voice** *tag*
3. **show voice mlpp voip-trunk sip**

DETAILED STEPS

Step 1 **enable**

Enables privileged EXEC mode.

Example:

```
Device> enable
```

Step 2 **show dial-peer voice** *tag*

Displays the information for voice dial peers. The following sample output shows the MLPP SIP queues of calls (last line of the output) for a particular dial peer.

Example:

```
Device# show dial-peer voice 1

VoiceOverIpPeer1
        peer type = voice, system default peer = FALSE, information type = voice,
        description = `',
        tag = 1, destination-pattern = `',
        voice reg type = 0, corresponding tag = 0,
        allow watch = FALSE
        answer-address = `', preference=0,
        CLID Restriction = None
        CLID Network Number = `'
        CLID Second Number sent
        CLID Override RDNIS = disabled,
        rtp-ssrc mux = system
        source carrier-id = `', target carrier-id = `',
        source trunk-group-label = `',  target trunk-group-label = `',
        numbering Type = `unknown'
        group = 1, Admin state is up, Operation state is down,
        incoming called-number = `', connections/maximum = 0/2,
        bandwidth/maximum = 0/unlimited,
        DTMF Relay = disabled,
        modem transport = system,
        URI classes:
            Incoming (Request) =
            Incoming (Via) =
            Incoming (To) =
            Incoming (From) =
            Destination =
        Destination route-string = None
        huntstop = disabled,
        in bound application associated: 'DEFAULT'
        out bound application associated: ''
        dnis-map =
        permission :both
        incoming COR list:maximum capability
        outgoing COR list:minimum requirement
        outgoing LPCOR:
        Translation profile (Incoming):
        Translation profile (Outgoing):
        incoming call blocking:
```

```
translation-profile = `'
disconnect-cause = `no-service'
advertise 0x40 capacity_update_timer 25 addrFamily 4 oldAddrFamily 4
mailbox selection policy: none
type = voip, session-target = `',
technology prefix:
settle-call = disabled
ip media DSCP = ef, ip media rsvp-pass DSCP = ef
ip media rsvp-fail DSCP = ef, ip signaling DSCP = af31,
ip video rsvp-none DSCP = af41,ip video rsvp-pass DSCP = af41
ip video rsvp-fail DSCP = af41,
ip defending Priority = 0, ip preemption priority = 0
ip policy locator voice:
ip policy locator video:
UDP checksum = disabled,
IPv6 UDP checksum = disabled
session-protocol = sipv2, session-transport = system,
req-qos = best-effort, acc-qos = best-effort,
req-qos video = best-effort, acc-qos video = best-effort,
req-qos audio def bandwidth = 64, req-qos audio max bandwidth = 0,
req-qos video def bandwidth = 384, req-qos video max bandwidth = 0,
RTP dynamic payload type values: NTE = 101
Cisco: NSE=100, fax=96, fax-ack=97, dtmf=121, fax-relay=122
       CAS=123, TTY=119, ClearChan=125, PCM switch over u-law=0,
       A-law=8, GSMAMR-NB=117 iLBC=116, AAC-ld=114, iSAC=124
       lmr_tone=0, nte_tone=0
       h263+=118, h264=119
       G726r16 using static payload
       G726r24 using static payload
RTP comfort noise payload type = 19
fax rate = voice,   payload size =  20 bytes
fax protocol = system
fax-relay ecm enable
Fax Relay ans treatment disabled
Fax Relay ans enabled
Fax Relay SG3-to-G3 Enabled (by system configuration)
fax NSF = 0xAD0051 (default)
codec = g729r8,   payload size =  20 bytes,
video codec = None
voice class codec = `'
voice class sip session refresh system
voice class sip rsvp-fail-policy voice post-alert mandatory keep-alive interval 30
voice class sip rsvp-fail-policy voice post-alert optional keep-alive interval 30
voice class sip rsvp-fail-policy video post-alert mandatory keep-alive interval 30
voice class sip rsvp-fail-policy video post-alert optional keep-alive interval 30
text relay = disabled
Media Setting = forking (disabled) flow-through (global)stats-disconnect (disabled)
Expect factor = 10, Icpif = 20,
Playout Mode is set to adaptive,
Initial 60 ms, Max 1000 ms
Playout-delay Minimum mode is set to default, value 40 ms
Fax nominal 300 ms
Max Redirects = 1, signaling-type = cas,
VAD = enabled, Poor QOV Trap = disabled,
Source Interface = NONE
voice class sip url = system,
voice class sip tel-config = system,
voice class sip rel1xx = system,
tvoice class sip outbound-proxy = system,
voice class sip asserted-id = system,
voice class sip privacy = system,
voice class sip e911 = system,
voice class sip history-info = system,
voice class sip pass-thru headers = system,
voice class sip pass-thru subscribe-notify-events = system,
voice class sip pass-thru content unsupp = system,
voice class sip pass-thru content sdp = system,
voice class sip copy-list = system,
voice class sip anat = system,
voice class sip g729 annexb-all = system,
voice class sip early-offer forced = system,
```

```
        voice class sip negotiate cisco = system,
        voice class sip reset timer expires 183 = system,
        voice class sip block 180 = system,
        voice class sip block 181 = system,
        voice class sip block 183 = system,
        voice class sip preloaded-route = system,
        voice class sip random-contact = system,
        voice class sip random-request-uri validate = system,
        voice class sip call-route p-called-party-id = system,
        voice class sip call-route history-info = system,
        voice class sip call-route url = system,
        voice class sip privacy-policy send-always = system,
        voice class sip privacy-policy passthru = system,
        voice class sip privacy-policy strip history-info = system,
        voice class sip privacy-policy strip diversion = system,
        voice class sip bandwidth audio  = system,
        voice class sip bandwidth video  = system,
        voice class sip error-code-override options-keepalive failure = system,
        voice class sip error-code-override call spike failure = system,
        voice class sip error-code-override cac-bandwidth failure = system,
        voice class sip encap clear-channel  = system,
        voice class sip send 180 sdp = system,
        voice class sip map resp-code 181 = system,
        voice class sip bind control = system,
        voice class sip bind media = system,
        voice class sip registration passthrough = System
        voice class sip nat mode = System
        voice class sip conn reuse = System
        voice class sip authenticate redirecting-number = system,
        voice class sip referto-passing = system,
        voice class sip extension = system,
        voice class phone proxy name: None
        voice class phone proxy config: N/A
        redirect ip2ip = disabled
        local peer = false
        probe disabled,
        Secure RTP: system (use the global setting)
        mobility=0, snr=, snr_noan=, snr_delay=0, snr_timeout=0
        snr calling-number local=disabled, snr ring-stop=disabled, snr answer-too-soon timer=0
rtcp_keepalive = system

        voice class perm tag = `'
        Time elapsed since last clearing of voice call statistics never
        Connect Time = 0, Charged Units = 0,
        Successful Calls = 0, Failed Calls = 0, Incomplete Calls = 0
        Accepted Calls = 0, Refused Calls = 0,
        Bandwidth CAC Accepted Calls = 0, Bandwidth CAC Refused Calls = 0,
        Last Disconnect Cause is "",
        Last Disconnect Text is "",
        Last Setup Time = 0.
        Last Disconnect Time = 0.
       MLPP SIP Queues:  0-1 1-0 2-0 3-0 4-0 5-0 6-0 7-0 8-1 9-0
```

Step 3 **show voice mlpp voip-trunk sip**

Displays information about the SIP MLPP call queues. In the sample output, (A) means ACTIVE when a call ID is non-local call and (I) means INACTIVE when callID is a local call. During preemption, INACTIVE calls are skipped. LEVEL 0 means internal precedence level is 0.

Example:

`Device#` **show voice mlpp voip-trunk sip**

```
 dial-peer voice 3 voip
 MLPP Preempted Count = 0.
 LEVEL 0: 23(A) 22(A) 24(I)
 LEVEL 8: 28(A)
```

Troubleshooting Tips

Use the **debug voip mlpp** command to troubleshoot the Emergency Number Preemption feature.

Configuration Examples for Emergency Number Preemption

Example: Configuring Emergency Number

```
Device> enable
Device# configure terminal
Device(config)# voice service voip
Device(conf-voi-serv)# emergency 912 913
Device(conf-voi-serv)# end
```

Example: Configuring Preemption and the Maximum Connections on SIP Dial Peer

```
Device> enable
Device# configure terminal
Device(config)# voice mlpp
Device(config-voice-mlpp)# preemption voip-trunk sip
Device(config-voice-mlpp)# exit
Device(config)# dial-peer voice 30 voip
Device(config-dial-peer)# max-conn 2
Device(config-dial-peer)# end
```

Feature Information for Emergency Number Preemption

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 Emergency Number Preemption

Feature Name	Releases	Feature Information
Nano CUBE (Emergency Number Preemption)	15.3(3)M	The emergency number preemption (also called 911-preemption) feature enables you to configure a list of emergency numbers. When the maximum number of incoming or outgoing connections on a dial-peer is reached, the other non-emergency calls are preempted from the session initiation protocol (SIP) dial-peer, allowing the emergency call to go through.