Logix Communications SIP Trunking:

Cisco Unified Communications Manager 11.5.1 with Cisco Unified Border Element (CUBE 11.5.2) on ISR4321 [IOS-XE 16.3.2] using SIP

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Introduction

Service Providers today, such as Logix Communications, are offering alternative methods to connect to the PSTN via their IP network. Most of these services utilize SIP as the primary signaling method and centralized IP to TDM POP gateways to provide on-net and off-net services.

Logix Communications is a service provider that allows connection to the PSTN and may offer the end customer a viable alternative to traditional PSTN connectivity. A demarcation device between these services and customer owned services is recommended. As an intermediary device between Cisco Unified Communications Manager and Logix Communications Network, Cisco Unified Border Element (CUBE v11.5.2) on ISR 4321 running IOS-XE 16.3.2 can be used. The Cisco Unified Border Element provides demarcation, security, interworking and session control services for Cisco Unified Communications Manager 11.5.1 connected to Logix Communications IP network.

This document assumes the reader is knowledgeable with the terminology and configuration of Cisco Unified Communications Manager. Only configuration settings specifically required for Logix Communications interoperability are presented. Feature configuration and most importantly the dial plan are customer specific and need individual approach.

- This application note describes how to configure Cisco Unified Communications Manager (Cisco UCM) 11.5.1, and Cisco Unified Border Element (Cisco UBE) on ISR 4321 [IOS-XE version 16.3.2] for connectivity to Logix Communications SIP Trunking service. The deployment model covered in this application note is CPE (Cisco Unified Communications Manager 11.5.1) to PSTN (Logix).

- Testing was performed in accordance to Logix Communications generic SIP Trunking test methodology and among features verified were – basic calls, DTMF transport, Music on Hold (MOH), unattended and attended transfers, call forward, conferences and interoperability with Cisco Unity Connection (CUC).

- The Cisco UCM configuration detailed in this document is based on a lab environment with a simple dial-plan used to ensure proper interoperability between Logix Communications SIP network and Cisco Unified Communications. The configuration described in this document details the important configuration settings to have enabled for interoperability to be successful and care must be taken by the network administrator deploying Cisco UCM to interoperate to Logix Communications SIP Trunking network.

This application note does not cover the use of Calling Search Spaces (CSS) or partitions on Cisco UCM. To understand and learn how to apply CSS and partitions refer to the cisco.com link below:

Network Topology

Figure 1 Network Topology

Figure 2: Cisco UBE High Availability
System Components

Hardware Requirements

- Cisco UCS-C240-M3S VMWare host running ESXi 5.5 Standard
- Cisco ISR 4321/K9 router as Cisco UBE
- Cisco ISR4321/K9 (1RU) processor with 1651964K/6147K bytes of memory with 3 Gigabit Ethernet interfaces
- Processor board ID FLM1925W0WZ
- Cisco 2851 Fax Gateway
- IP phones 9951 (SIP) and 7941 (SCCP)
- Adtran Total Access 900e 3rd Gen eSBC – Provided and Managed by Logix Communications

Software Requirements

- Cisco Unified Communications Manager 11.5.1.12900-21
- Cisco Unity Connection 11.5.1.12900-21
- Cisco IOS XE Software, Version 16.03.02 running as Cisco Unified Border Element
- IOS 15.1(4)M5 for Cisco 2851 Fax Gateway
- Adtran Total Access 900e 3rd Gen eSBC Release 11.6.0.SA.E– Provided and Managed by Logix Communications
Features

Features Supported

- Incoming and outgoing off-net calls using G711ULAW & G729 voice codecs
- Call Hold
- Call Transfer (unattended and attended)
- Call Conference
- Call Forward (all, busy, no answer)
- Calling Line (number) Identification Presentation (CLIP)
- Calling Line (number) Identification Restriction (CLIR)
- DTMF Relay (both directions) (RFC2833)
- Media flow-through on Cisco UBE
- Fax (T.38 and G711 Pass-through)

Features Not Supported

- Cisco IP phones used in this test do not support Blind Transfer
- 0 + 10 digit dial plan – Operator assisted call is not supported by Logix Communications
- In HA Redundancy mode, the Primary cube will not take over the Primary/Active role after a reboot/network outage

Caveats

- Caller ID is not updated after attended or semi-attended transfers to off-net phones. This is due to a limitation on Cisco UBE and will be resolved in the next release. The issue does not impact the calls.
- Logix Communications supports faxing up to G3/V.17 with T.38 Version 0 and G711 Pass-through
- Call Forward Unconditional to CPE User scenarios was executed by configuring Calling Party Selection* to “First Redirect number (External)” under Trunk Settings
Configuration

Configuring the Cisco Unified Border Element

Network interface
Configure Ethernet IP address and sub interface. The IP address and VLAN encapsulation used are for illustration only, the actual IP address can vary. For SIP trunks two IP addresses must be configured—LAN and WAN.

```
interface GigabitEthernet0/0/0
  description LogixCommunication WAN
  ip address 10.70.29.5 255.255.255.0
  media-type rj45
  negotiation auto
  redundancy rii 4
  redundancy group 1 ip 10.70.29.4 exclusive

interface GigabitEthernet0/0/1
  description LogixCommunication LAN
  ip address 10.80.18.11 255.255.255.0
  negotiation auto
  redundancy rii 3
  redundancy group 1 ip 10.80.18.10 exclusive
```
Global Cisco UBE settings

In order to enable Cisco UBE IP2IP gateway functionality, enter the following:

```bash
! 
voice service voip
no ip address trusted authenticate
address-hiding
mode border-element license capacity 20
allow-connections sip to sip
redundancy-group 1
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
sip
rel1xx supported "rel100"
session refresh
asserted-id pai
privacy pstn
early-offer forced
midcall-signaling passthru
g729 annexb-all
!
```

**Explanation**

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>allow-connections sip to sip</td>
<td>Allow IP2IP connections between two SIP call legs</td>
</tr>
<tr>
<td>fax protocol</td>
<td>Specifies the fax protocol</td>
</tr>
<tr>
<td>asserted-id</td>
<td>Specifies the type of privacy header in the outgoing SIP requests and response messages</td>
</tr>
<tr>
<td>early-offer forced</td>
<td>Enables SIP Delayed-Offer to Early-Offer globally</td>
</tr>
<tr>
<td>midcall-signaling passthru</td>
<td>Passes SIP messages from one IP leg to another IP leg</td>
</tr>
</tbody>
</table>
Codecs

G711ulaw and G729 voice codecs are used for this testing. Codec preferences can be changed according to the test plan description

  voice class codec 1
  codec preference 1 g711ulaw
  codec preference 2 g729r8

Dial Peer

Cisco UBE uses dial-peer to route the call based on the digit to route the call accordingly.

  dial-peer voice 500 voip
    description Outgoing Call to Logix - LAN facing
    huntstop
    session protocol sipv2
    session transport udp
    incoming called-number [0-9]T
    voice-class codec 1
    voice-class sip asserted-id pai
    voice-class sip bind control source-interface GigabitEthernet0/0/1
    voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate 14400
fax nsf 000000
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad

  dial-peer voice 510 voip
    description Outgoing call to Logix- WAN facing
    huntstop
destination-pattern [0-9]T
session protocol sipv2
session target sip-server
session transport udp
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip profiles 101
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-n-te
fax-relay ecm disable
fax rate 14400
fax nsf 000000
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 520 voip
description Incoming call to PBX - WAN facing
huntstop
session protocol sipv2
session transport udp
incoming called-number 469708....
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-n-te
fax-relay ecm disable
fax rate 14400
fax nsf 000000
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 530 voip
description Incoming call to PBX - LAN facing
translation-profile outgoing LogixCom
huntstop
destination-pattern 469708....
session protocol sipv2
session target ipv4:10.80.18.3:5060
session transport udp
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate 14400
fax nsf 000000
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
Call flow

In the sample configuration presented here, Cisco UCM is provisioned with four-digit directory numbers corresponding to the last four DID digits. No digit manipulation is performed on the Cisco UBE.

For incoming PSTN calls, the Cisco UBE presents the full ten-digit DID number to Cisco UCM. The Cisco UCM picks up the last 4 significant Digits configured under SIP Trunk and routes the call based on those 4 digits. Voice calls are routed to IP phones; Fax calls are routed via a 4-digit route pattern over a SIP trunk that terminates on the Fax Gateway and in turn to the VentaFax client connected to the Fax Gateway.

CPE callers make outbound PSTN calls by dialing a “8” prefix followed by the destination number. For outbound fax calls from the analog fax endpoint, Cisco fax Gateway sends to Cisco UCM the DID with leading access code “8”. A “8.@” route pattern strips the prefix and routes the call with the remaining digits via a SIP trunk terminating on the Cisco UBE for Voice call or Fax. For PBX to PBX via Logix Communications, Caller dial 8 prefix followed by the target 1+10-digits number, 8 was stripped and the remaining digits were send to Cisco UBE, Cisco UBE pass the DID under Dial Peer 510 and send to Logix Communications network which will direct back to Cisco UBE and handled same as normal incoming PSTN call.

![Figure 3: Outbound Voice Call](image)

![Figure 4: Outbound Fax Call](image)
Figure 5: Inbound Voice Call

Figure 6: Inbound Fax Call

Figure 7: PBX to PBX via Logix Call
Configuration Example

The following configuration snippet contains a sample configuration of Cisco Unified Border Element with all parameters mentioned previously.

**Active Cisco UBE**

Logix1#sh running-config
Building configuration...

Current configuration: 8264 bytes
!
version 16.3
service timestamps debug datatime msec localtime
service timestamps log datatime msec localtime
service password-encryption
service internal
service sequence-numbers
no platform punt-keepalive disable-kernel-core
!
hostname Logix1
!
boot-start-marker
boot system bootflash:isr4300-universalk9.16.03.02.SPA.bin
boot-end-marker
!
vrf definition Mgmt-intf
!
address-family ipv4
exit-address-family
!
address-family ipv6
exit-address-family

! no logging queue-limit
no logging buffered
no logging rate-limit
enable secret 5

! no aaa new-model
no ip domain lookup

! subscriber templating
multilink bundle-name authenticated

! crypto pki trustpoint TP-self-signed-1270583006
   enrollment selfsigned
   subject-name cn=IOS-Self-Signed-Certificate-1270583006
   revocation-check none
   rsakeypair TP-self-signed-1270583006

! crypto pki certificate chain TP-self-signed-1270583006
   certificate self-signed 01
      30820330 30820218 A0030201 02020101 300D0609 2A864886 F70D0101 05050030
      31312F30 2D060355 04031326 494F532D 53656C66 2D536967 6E65642D 43657274
      69666963 6174652D 31323730 35383330 3036301E 170D3137 30323130 31343237
      34354117 0D323030 31303130 30303030 305A3031 312F302D 06035504 03132649
      4F532D53 656C662D 5369676E 65642D43 65727469 66696361 74652D31 32373035
      38333030 36308201 22300D06 092A8648 86F70D01 01010500 0382010F 00308201
      0A028201 0100A34F 136DD7A1 E1815DA1 B05C5396 E3B88AC9 7DE1A1D7 12F1BEA7
      4985E12B C6858F9D 95E7082B 3BBC56CD AAFDEC4F 7250D4CE 892713BE 509A6DCE
quit

voice service voip

no ip address trusted authenticate

address-hiding

mode border-element license capacity 20

allow-connections sip to sip

redundancy-group 1

fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none

sip

rel1xx supported "rel100"

session refresh

asserted-id pai
privacy pstn
early-offer forced
midcall-signaling passthru
g729 annexb-all

voice class codec 1
codec preference 1 g711ulaw
codec preference 2 g729r8
!
voice class codec 2
codec preference 1 g729r8
codec preference 2 g711ulaw
!
voice class codec 3
codec preference 1 g729r8
!

voice class sip-profiles 101
request INVITE sip-header Diversion modify "<sip:(.*)@(.*)>" "sip:469708\1@\2"
!
voice translation-rule 1004
rule 1 /4697081413/ /4000/
!
voice translation-profile LogixCom
translate called 1004
!
license udi pid ISR4321/K9 sn FDO17860MQ8
license boot level appxk9
license boot level uck9
!
no diagnostic bootup level
spanning-tree extend system-id

redundancy
mode none
application redundancy
group 1
    name b2bhalogixcommunications
    priority 100 failover threshold 75
timers delay 30 reload 60
control GigabitEthernet0/1/0 protocol 1
data GigabitEthernet0/1/0
track 1 shutdown
track 2 shutdown

vlan internal allocation policy ascending

track 1 interface GigabitEthernet0/0/0 line-protocol
track 2 interface GigabitEthernet0/0/1 line-protocol

interface GigabitEthernet0/0/0
    description LogixCommunication WAN
    ip address 10.70.29.5 255.255.255.0
    media-type rj45
    negotiation auto
    redundancy rii 4
    redundancy group 1 ip 10.70.29.4 exclusive

interface GigabitEthernet0/0/1
    description LogixCommunication LAN
    ip address 10.80.18.11 255.255.255.0
    negotiation auto
redundancy rii 3
redundancy group 1 ip 10.80.18.10 exclusive
!
interface GigabitEthernet0/1/0
description CUBE HA
ip address 10.89.20.7 255.255.255.0
negotiation auto
!
interface GigabitEthernet0
vrf forwarding Mgmt-intf
no ip address
negotiation auto
!
interface Vlan1
no ip address
shutdown
!
ip forward-protocol nd
no ip http server
no ip http secure-server
ip tftp source-interface GigabitEthernet0
ip route 0.0.0.0 0.0.0.0 10.70.29.1
ip route 10.64.0.0 255.255.0.0 10.80.18.1
ip route 10.70.0.0 255.255.0.0 10.70.29.1
ip route 10.80.0.0 255.255.0.0 10.80.18.1
ip route 172.16.0.0 255.255.0.0 10.80.18.1
!
control-plane
!
mgcp behavior rsip-range tgcp-only
mgcp behavior comedia-role none
mgcp behavior comedia-check-media-src disable
mgcp behavior comedia-sdp-force disable
!
mgcp profile default
!
dial-peer voice 500 voip
description Outgoing Call to Logix - LAN facing
huntstop
session protocol sipv2
session transport udp
incoming called-number [0-9]T
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-nce
fax-relay ecm disable
fax rate 14400
fax nsf 000000
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 510 voip
description Outgoing call to Logix- WAN facing
huntstop
destination-pattern [0-9]T
session protocol sipv2
session target sip-server
session transport udp
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip profiles 101
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate 14400
fax nsf 000000
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 520 voip
description Incoming call to PBX - WAN facing
huntstop
session protocol sipv2
session transport udp
incoming called-number 469708....
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate 14400
fax nsf 000000
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 530 voip
description Incoming call to PBX - LAN facing
translation-profile outgoing LogixCom
huntstop
destination-pattern 469708....
session protocol sipv2
session target ipv4:10.80.18.3:5060
session transport udp
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-n-te
fax-relay ecm disable
fax rate 14400
fax nsf 000000
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
sip-ua
keepalive target ipv4:10.70.29.3:5060
timers keepalive active 60
sip-server ipv4:10.70.29.3:5060
!
!
line con 0
stopbits 1
line aux 0
stopbits 1
line vty 0 4
exec-timeout 0 0
password 7
login
line vty 5
  exec-timeout 0 0
password 7
login
!
end
Standby Cisco UBE

version 16.3
service timestamps debug datetime msec
service timestamps log datetime msec
service password-encryption
no platform punt-keepalive disable-kernel-core

hostname Logix2

boot-start-marker
boot system bootflash:isr4300-universalk9.16.03.02.SPA.bin
boot-end-marker

vrf definition Mgmt-intf

address-family ipv4
exit-address-family

address-family ipv6
exit-address-family

no logging queue-limit
no logging buffered
no logging rate-limit
enable secret 5

no aaa new-model

no ip domain lookup
subscriber templating
multilink bundle-name authenticated

! crypto pki trustpoint TP-self-signed-2548443246
  enrollment selfsigned
  subject-name cn=IOS-Self-Signed-Certificate-2548443246
  revocation-check none
  rsakeypair TP-self-signed-2548443246
  !
  crypto pki certificate chain TP-self-signed-2548443246
  certificate self-signed 01
  30820330 30820218 A0030201 02020101 300D0609 2A864886 F70D0101 05050030
  31312F30 2D060355 04031326 494F5F32D 53656C66 2D536967 6E65642D 43657374
  69666963 6174652D 32353438 34343332 3436301E 170D3137 30323310 31373132
  33365A17 0D323030 31303130 30303030 305A3031 012F302D 06035504 03132649
  4F532D53 656C662D 5369676E 65642D43 65727469 66696361 74652D32 35343834
  34333234 36308021 22300D06 09A86488 86F70D01 01010500 0382010F 00308201
  0A028201 0100B73A 8AE876C0 62A381D9 4C331F21 6FBF60E9 20F942A0 6F2C3A5A
  2DA4B74B 1A9B55DB 65BC3A4B 0164E96 3CB638A7 31C61AA1 A2E8EF3E FE7733F5
  A0035F13 9AE153CE D55D4F6F FBBCA3CE EC8D110A 6490B2CD 445909DE 1A460E75
  66CF3C75 3DF0BBBE 7B27306D C2ACDBA2 A3497E3D 7EFDC2B 1902A0A8 038AD01E
  68FD339C B2620BFF 00E70305 88DD7796 B7C0351F 27BFF1EC 791E8F53 87B57E81
  166B26BA 1428E6F7 A7484680 ACF2B8F7 BB95977B C3854F78 D4295377 CE568896
  451A72F7 5D117423 6A69CEEE 8E13BDAB 96B61B29 1165C7C4 E7DB2B8B 1A2095F6
  E9C80DDD 9B1DCED7 CE87C3F0 BF726628 6AF272BA 9A3D33B0 4AD1444F 87DF933F
  9BCF78D 3C6B2003 010001A3 53305130 0F060355 1D130101 FF040530 030101FF
  301F0603 551D2308 18301680 2EADBB3 9378001B 592B6EC4 0CD4C83F 3FA14FA3
  AD301D06 03551D0E 04160414 EADDB393 78001B59 2B6EC40C D4C83F3F A14FA3AD
  300D0609 2A864886 F70D0101 05050030 82010100 A99C7D6F 325E52A8 9F0221FE
  3BC2460A 5383DF63 B1A66575 D8CE6F8 F725B14FA F18879B1 38173AF8 3A05D05F
  B72318A1 23B11F12 E14DE814 CF93938D 41F75435 21999BC2 6FFAFBAC 84347F5F
quit

voice service voip
no ip address trusted authenticate
address-hiding
mode border-element license capacity 20
allow-connections sip to sip
redundancy-group 1
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
sip
rel1xx supported "rel100"
session refresh
asserted-id pai
privacy pstn
early-offer forced
midcall-signaling passthru
g729 annexb-all

voice class codec 1
codec preference 1 g711ulaw
codec preference 2 g729r8

voice class codec 2
codec preference 1 g729r8
codec preference 2 g711ulaw
voice class codec 3
  codec preference 1 g729r8

voice class sip-profiles 101
  request INVITE sip-header Diversion modify "<sip:(.*)@(.*)>" "sip:469708\1@\2"

voice translation-rule 1004
  rule 1 /4697081413/ /4000/

voice translation-profile LogixCom
  translate called 1004

license udi pid ISR4321/K9 sn FDO17860MW3
  license boot level appxk9
  license boot level uck9

  diagnostic bootup level minimal
  spanning-tree extend system-id

redundancy
  mode none
  application redundancy
    group 1
      name b2bhalogixcommunications
      priority 100 failover threshold 75
      timers delay 30 reload 60
      control GigabitEthernet0/1/0 protocol 1
      data GigabitEthernet0/1/0
      track 1 shutdown
      track 2 shutdown
vlan internal allocation policy ascending

track 1 interface GigabitEthernet0/0/0 line-protocol
track 2 interface GigabitEthernet0/0/1 line-protocol

interface GigabitEthernet0/0/0
  description LogixCommunication WAN
  ip address 10.70.29.6 255.255.255.0
  media-type rj45
  negotiation auto
  redundancy rii 4
  redundancy group 1 ip 10.70.29.4 exclusive

interface GigabitEthernet0/0/1
  description LogixCommunication LAN
  ip address 10.80.18.12 255.255.255.0
  negotiation auto
  redundancy rii 3
  redundancy group 1 ip 10.80.18.10 exclusive

interface GigabitEthernet0/1/0
  description CUBE HA
  ip address 10.89.20.8 255.255.255.0
  negotiation auto

interface GigabitEthernet0
  vrf forwarding Mgmt-intf
  no ip address
  negotiation auto

interface Vlan1
no ip address
shutdown

ip forward-protocol nd
no ip http server
no ip http secure-server
ip tftp source-interface GigabitEthernet0
ip route 0.0.0.0 0.0.0.0 10.70.29.1
ip route 0.0.0.0 0.0.0.0 10.64.1.1
ip route 10.70.0.0 255.255.0.0 10.70.29.1
ip route 10.80.0.0 255.255.0.0 10.80.18.1
ip route 172.16.0.0 255.255.0.0 10.80.18.1

! control-plane

! mgcp behavior rsip-range tgcp-only
mgcp behavior comedia-role none
mgcp behavior comedia-check-media-src disable
mgcp behavior comedia-sdp-force disable

! mgcp profile default

! dial-peer voice 500 voip
description Outgoing Call to Logix - LAN facing
huntstop
session protocol sipv2
session transport udp
incoming called-number [0-9]T
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip bind control source-interface GigabitEthernet0/0/1
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate 14400
fax nsf 000000
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 510 voip
description Outgoing call to Logix- WAN facing
huntstop
destination-pattern [0-9]T
session protocol sipv2
session target sip-server
session transport udp
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip profiles 101
voice-class sip bind control source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate 14400
fax nsf 000000
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 520 voip
description Incoming call to PBX - WAN facing
huntstop
session protocol sipv2
session transport udp
incoming called-number 469708....
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate 14400
fax nsf 000000
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
dial-peer voice 530 voip
description Incoming call to PBX - LAN facing
huntstop
destination-pattern 469708....
session protocol sipv2
session target ipv4:10.80.18.3:5060
session transport udp
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate 14400
fax nsf 000000
fax protocol t38 version 0 ls-redundancy 0 hs-redundancy 0 fallback none
no vad
!
!
sip-ua
  keepalive target ipv4:10.70.29.3:5060
timers keepalive active 60
  sip-server ipv4:10.70.29.3:5060
!
!
line con 0
  stopbits 1
line aux 0
  stopbits 1
line vty 0 5
  exec-timeout 0 0
  password 7
  login
!

end
Configuring Cisco Unified Communications Manager

Cisco UCM Version

Navigation: System > Service Parameters

1. Select Server* = Clus28Sub1--CUCM Voice/Video (Active)
2. Select Service* = Cisco CallManager (Active)
3. Duplex Streaming Enabled = True
4. All other fields are set to default values

Figure 8: Cisco UCM Version

Cisco Call Manager Service Parameters

Figure 9: Service Parameters
<table>
<thead>
<tr>
<th>Clusterwide Parameters (Service)</th>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Default Network Hold MCH Audio Source ID *</td>
<td>1</td>
<td>1</td>
</tr>
<tr>
<td>Default User Hold MCH Audio Source ID *</td>
<td>1</td>
<td>1</td>
</tr>
<tr>
<td>Duplex Streaming Enabled *</td>
<td>True</td>
<td>False</td>
</tr>
<tr>
<td>Media Exchange Interface Capability Timer *</td>
<td>8</td>
<td>8</td>
</tr>
<tr>
<td>Send Multicast MCH in H.245 OLC Message *</td>
<td>True</td>
<td>True</td>
</tr>
<tr>
<td>Media Exchange Timer *</td>
<td>12</td>
<td>12</td>
</tr>
<tr>
<td>Media Exchange Stop Streaming Timer *</td>
<td>8</td>
<td>8</td>
</tr>
<tr>
<td>Open Video Channel Response Timer for SIP Interop *</td>
<td>500</td>
<td>500</td>
</tr>
<tr>
<td>Port Received Timer After Call Connection *</td>
<td>500</td>
<td>500</td>
</tr>
<tr>
<td>Media Resource Allocation Timer *</td>
<td>12</td>
<td>12</td>
</tr>
<tr>
<td>RSVP and Transcoder Resource Throttling Percentage *</td>
<td>95</td>
<td>95</td>
</tr>
<tr>
<td>Intercluster Capabilities Mismatch Timer *</td>
<td>1000</td>
<td>1000</td>
</tr>
<tr>
<td>Silence Suppression *</td>
<td>True</td>
<td>False</td>
</tr>
<tr>
<td>Silence Suppression for Gateways *</td>
<td>True</td>
<td>False</td>
</tr>
<tr>
<td>Strip G.729 Annex B (Silence Suppression) from Capabilities *</td>
<td>True</td>
<td>False</td>
</tr>
<tr>
<td>Enable Source IP Address Verification for Software Media Devices *</td>
<td>True</td>
<td>True</td>
</tr>
</tbody>
</table>

Figure 10: Service Parameters (Cont.)
Off-net Calls via Logix Communications SIP Trunk
Off-net calls are served by SIP trunks configured between Cisco UCM and Logix Communications Network and calls are routed via Cisco UBE

**SIP Trunk Security Profile**

**Navigation:** System → Security → SIP Trunk Security Profile

1. Name* = LogixCommunications Non Secure SIP Trunk Profile is used as an example
2. Description = Non Secure SIP Trunk Profile authenticated by null String is used as an example
3. Device Security Mode = Non Secure
4. Incoming Transport Type* = TCP + UDP
5. Outgoing Transport Type = UDP

---

**Figure 11: SIP Trunk Security Profile**

**Explanation**

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Incoming Transport Type</td>
<td>TCP + UDP</td>
<td>SIP trunks to Logix Communications SBC should use UDP as a transport protocol for SIP. This is configured using SIP Trunk Security profile, which is later assigned to the SIP trunk itself.</td>
</tr>
<tr>
<td>Outgoing Transport Type</td>
<td>UDP</td>
<td></td>
</tr>
</tbody>
</table>
SIP Profile Configuration

Note: SIP Profile will be later associated with the SIP trunk

Navigation: Device → Device Settings → SIP Profile

1. Name* = LogixCommunications Standard SIP Profile is used as an example
2. Description = Default SIP Profile is used as an example

Figure 12: SIP Profile
**Figure 13: SIP Profile (Cont.)**

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Start Media Port</td>
<td>16384</td>
</tr>
<tr>
<td>Stop Media Port</td>
<td>32766</td>
</tr>
<tr>
<td>DSCP for Audio Calls</td>
<td>Use System Default</td>
</tr>
<tr>
<td>DSCP for Video Calls</td>
<td>Use System Default</td>
</tr>
<tr>
<td>DSCP for Audio Portion of Video Calls</td>
<td>Use System Default</td>
</tr>
<tr>
<td>DSCP for TelePresence Calls</td>
<td>Use System Default</td>
</tr>
<tr>
<td>DSCP for Audio Portion of TelePresence Calls</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Call Pickup URI</td>
<td>x-cisco-serviceuri-pickup</td>
</tr>
<tr>
<td>Call Pickup Group Other URI</td>
<td>x-cisco-serviceuri-opickup</td>
</tr>
<tr>
<td>Call Pickup Group URI</td>
<td>x-cisco-serviceuri-gpickup</td>
</tr>
<tr>
<td>Meet Me Service URI</td>
<td>x-cisco-serviceuri-meetme</td>
</tr>
<tr>
<td>User Info</td>
<td>None</td>
</tr>
<tr>
<td>DTMF DB Level</td>
<td>Nominal</td>
</tr>
<tr>
<td>Call Hold Ring Back</td>
<td>Off</td>
</tr>
<tr>
<td>Anonymous Call Block</td>
<td>Off</td>
</tr>
<tr>
<td>Caller ID Blocking</td>
<td>Off</td>
</tr>
<tr>
<td>Do Not Disturb Control</td>
<td>User</td>
</tr>
<tr>
<td>Telnet Level for 7940 and 7960</td>
<td>Disabled</td>
</tr>
<tr>
<td>Resource Priority Namespace</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Timer Keep Alive Expires (seconds)</td>
<td>120</td>
</tr>
<tr>
<td>Timer Subscribe Expires (seconds)</td>
<td>120</td>
</tr>
<tr>
<td>Timer Subscribe Delta (seconds)</td>
<td>5</td>
</tr>
<tr>
<td>Maximum Redirections</td>
<td>70</td>
</tr>
<tr>
<td>Off Hook To First Digit Timer (milliseconds)</td>
<td>15000</td>
</tr>
<tr>
<td>Call Forward URI</td>
<td>x-cisco-serviceuri-fwdall</td>
</tr>
<tr>
<td>Speed Dial (Abbreviated Dial) URI</td>
<td>x-cisco-serviceuri-abbrevdial</td>
</tr>
<tr>
<td>Conference Join Enabled</td>
<td></td>
</tr>
<tr>
<td>RFC 2543 Hold</td>
<td></td>
</tr>
<tr>
<td>Semi Attended Transfer</td>
<td></td>
</tr>
<tr>
<td>Enable VAD</td>
<td></td>
</tr>
<tr>
<td>Stutter Message Waiting</td>
<td></td>
</tr>
<tr>
<td>MLPP User Authorization</td>
<td></td>
</tr>
</tbody>
</table>

**Normalization Script**

- Normalization Script: < None >
- Enable Trace: false

**Parameter Name** | **Parameter Value**
--- | ---
1 |
**Explanation**

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Default MTP Telephony Event Payload Type</td>
<td>101</td>
<td>RFC2833 DTMF payload type</td>
</tr>
<tr>
<td>SIP Rel1XX Options</td>
<td>Send PRACK for 1xx Contains SDP</td>
<td>Enable Provisional Acknowledgements (Reliable 100 messages)</td>
</tr>
<tr>
<td>Ping Interval for In-service and Partially In-service Trunks (seconds)</td>
<td>60</td>
<td>OPTIONS message parameters- interval time</td>
</tr>
<tr>
<td>Ping Interval for Out-of-service Trunks (seconds)</td>
<td>120</td>
<td>OPTIONS message parameters- interval time</td>
</tr>
</tbody>
</table>
SIP Trunk Configuration

Create SIP trunks to Cisco UBE

**Navigation Path:** Device → Trunk

<table>
<thead>
<tr>
<th>Name</th>
<th>Description</th>
<th>Device test</th>
<th>Route Failure</th>
<th>Party</th>
<th>Party Group</th>
<th>Hop</th>
<th>Trunk Protocol</th>
<th>SIP Trunk Status</th>
<th>SIP Trunk Security Profile</th>
</tr>
</thead>
<tbody>
<tr>
<td>Trunk to FAX gateway for Login</td>
<td>Trunk to FAX gateway for Login</td>
<td>LogicCommunication deployed</td>
<td>FRF</td>
<td>SIP Trunk Full Service</td>
<td>18 minutes</td>
<td>FAX Profile</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Trunk to CUBE for LogicCommunications</td>
<td>Trunk to CUBE for LogicCommunications</td>
<td>LogicCommunication deployed</td>
<td>FRF</td>
<td>SIP Trunk Full Service</td>
<td>21 minutes</td>
<td>CUBE Profile</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Trunk to CUBE for LogicCommunications</td>
<td>Trunk to CUBE for LogicCommunications</td>
<td>LogicCommunication deployed</td>
<td>FRF</td>
<td>SIP Trunk Full Service</td>
<td>21 minutes</td>
<td>CUBE Profile</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Trunk to CUBE for LogicCommunications</td>
<td>Trunk to CUBE for LogicCommunications</td>
<td>LogicCommunication deployed</td>
<td>FRF</td>
<td>SIP Trunk Full Service</td>
<td>21 minutes</td>
<td>CUBE Profile</td>
<td></td>
<td></td>
<td></td>
</tr>
<tr>
<td>Logix Trunk to Unity</td>
<td>Logix Trunk to Unity</td>
<td>LogicCommunication deployed</td>
<td>FRF</td>
<td>SIP Trunk Full Service</td>
<td>7 minutes</td>
<td>Unity Profile</td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>

**Figure 15:** SIP Trunks List
### SIP Trunk Status

**Service Status:** Full Service  
**Duration:** Time In Full Service: 6 days 19 hours 27 minutes

<table>
<thead>
<tr>
<th><strong>Device Information</strong></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Product:</strong></td>
<td>SIP Trunk</td>
</tr>
<tr>
<td><strong>Device Protocol:</strong></td>
<td>SIP</td>
</tr>
<tr>
<td><strong>Trunk Service Type:</strong></td>
<td>None (Default)</td>
</tr>
</tbody>
</table>

- **Device Name:** Trunk_to_CUBE_for_LogixCommunications  
- **Description:** Trunk_to_CUBE_for_LogixCommunications  
- **Device Pool:** LogixCommunication_Devicepool  

- **Common Device Configuration:** < None >  
- **Call Classification:** Use System Default  
- **Media Resource Group List:** MRGL_Default  

- **Location:** Hub_None  
- **AAR Group:** < None >  
- **Tunneled Protocol:** None  
- **QSIG Variant:** No Changes  
- **ASN.1 ROSE OID Encoding:** No Changes  
- **Packet Capture Mode:** None  
- **Packet Capture Duration:** 0  

- **Media Termination Point Required:**  
- **RTP Video Call as Audio:**  
- **Path Replacement Support:**  
- **Transmit UTF-8 for Calling Party Name:**  
- **Transmit UTF-8 Names in QSIG APDU:**  
- **Unattended Port:**  
- **SRTP Allowed:** When using both SRTP and TLS  

- **Consider Traffic on This Trunk Secure:**  
- **Route Class Signaling Enabled:** Default  
- **Use Trusted Relay Point:**  
- **PSTN Access:**  
- **Run On All Active Unified CM Nodes:**  

### Intercompany Media Engine (IME)  
**E.164 Transformation Profile:** < None >  

### MLPP and Confidential Access Level Information  
**MLPP Domain:** < None >  
**Confidential Access Mode:** < None >  
**Confidential Access Level:** < None >

---

**Figure 16: SIP Trunk to Cisco UBE**
**Figure 17: SIP Trunk to Cisco UBE (Cont.)**

### MLPP and Confidential Access Level Information

<table>
<thead>
<tr>
<th>MLPP Domain</th>
<th>Confidential Access Mode</th>
<th>Confidential Access Level</th>
</tr>
</thead>
<tbody>
<tr>
<td>&lt; None &gt;</td>
<td>&lt; None &gt;</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

### Call Routing Information

- Remote-Party-Id
- Asserted-Identity
- Asserted-Type: Default
- SIP Privacy: Default

### Inbound Calls

- Significant Digits: 4
- Connected Line ID Presentation: Default
- Called Name Presentation: Default
- Calling Search Space: < None >
- AAR Calling Search Space: < None >
- Prefix DN: 

- Redirecting Diversion Header Delivery - Inbound

### Incoming Calling Party Settings

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

<table>
<thead>
<tr>
<th>Number Type</th>
<th>Prefix</th>
<th>Strip Digits</th>
<th>Calling Search Space</th>
<th>Use Device Pool CSS</th>
</tr>
</thead>
<tbody>
<tr>
<td>Incoming Number</td>
<td>Default</td>
<td>0</td>
<td>&lt; None &gt;</td>
<td>☑</td>
</tr>
</tbody>
</table>

### Incoming Called Party Settings

If the administrator sets the prefix to Default this indicates call processing will use prefix at the next level setting (DevicePool/Service Parameter). Otherwise, the value configured is used as the prefix unless the field is empty in which case there is no prefix assigned.

<table>
<thead>
<tr>
<th>Number Type</th>
<th>Prefix</th>
<th>Strip Digits</th>
<th>Calling Search Space</th>
<th>Use Device Pool CSS</th>
</tr>
</thead>
<tbody>
<tr>
<td>Incoming Number</td>
<td>Default</td>
<td>0</td>
<td>&lt; None &gt;</td>
<td>☑</td>
</tr>
</tbody>
</table>

### Connected Party Settings

- Connected Party Transformation CSS: < None >
- Use Device Pool Connected Party Transformation CSS: ☑

### Outbound Calls

- Called Party Transformation CSS: < None >
- Use Device Pool Called Party Transformation CSS: ☑
- Calling Party Transformation CSS: < None >
- Use Device Pool Calling Party Transformation CSS: ☑

- Calling Party Selection: First Redirect Number (External)
- Called Line ID Presentation: Default
- Calling Name Presentation: Default
- Calling and Connected Party Info Format: Deliver DN only in connected party

- Redirecting Diversion Header Delivery - Outbound
- Redirecting Party Transformation CSS: < None >
- Use Device Pool Redirecting Party Transformation CSS: ☑

### Caller Information

- Caller ID DN: 
- Caller Name: 
- Maintain Original Caller ID DN and Caller Name in Identity Headers: ☑
Figure 18: SIP Trunk to Cisco UBE (Cont.)
## Explanation

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Device Name</td>
<td>Trunk_to_CUBE_for_LogixCommunications</td>
<td>Name for the trunk</td>
</tr>
<tr>
<td>Device Pool</td>
<td>LogixCommunication_Devicepool</td>
<td>Default Device Pool is used for this trunk</td>
</tr>
<tr>
<td>Media Resource Group List</td>
<td>MRGL_Default</td>
<td>MRG with resources: ANN, CFB, MOH and MTP</td>
</tr>
<tr>
<td>Significant Digits</td>
<td>4</td>
<td>4 digits Extension for all CPE phones</td>
</tr>
<tr>
<td>Destination Address</td>
<td>10.80.18.10</td>
<td>IP address of the Cisco UBE Virtual LAN</td>
</tr>
<tr>
<td>SIP Trunk Security Profile</td>
<td>LogixCommunications Non Secure SIP Trunk Profile</td>
<td>SIP Trunk Security Profile configured earlier</td>
</tr>
<tr>
<td>SIP Profile</td>
<td>LogixCommunications Standard SIP Profile</td>
<td>SIP Profile configured earlier</td>
</tr>
</tbody>
</table>
Route Pattern Configuration

Navigation: Call Routing → Route/Hunt → Route Pattern

Route patterns are configured as below:

1. Cisco IP phone dial “8”+10 digits number to access PSTN via Cisco UBE
   - “8” is removed before sending to Cisco UBE
2. For FAX call, Access Code “8”+10 digits number is used at Cisco Fax gateway
   - “8” is removed at Cisco UCM
   - The rest of the number is sent to Cisco UBE to Logix Communications network
3. Incoming fax call to 14XX will be sent to Cisco Fax gateway
4. For Anonymous call, access code “8*”+10 digits number is used
   - “8*” is removed at Cisco UCM
   - The rest of the number is sent to Cisco UBE to Logix Communications network
5. Cisco IP phones dial X11 for emergency call and will send all digits to Cisco UBE to Logix Communications Network

![Route Patterns List](image)

Figure 19: Route Patterns List
**Pattern Definition**

<table>
<thead>
<tr>
<th>Route Pattern</th>
<th>8.8</th>
<th>Route Partition: &lt; None &gt;</th>
<th>Description: RP_for_LogixCommunications</th>
<th>Numbering Plan: NANP</th>
</tr>
</thead>
<tbody>
<tr>
<td>Route Filter</td>
<td>&lt; None &gt;</td>
<td>MLPP Precedence: Default</td>
<td>Resource Priority Namespace Network Domain: &lt; None &gt;</td>
<td>Route Class: Default</td>
</tr>
<tr>
<td>Gateway/Route List</td>
<td>Trunk_to_CUBE_for_LogixCommunications</td>
<td>Route Option: Route this pattern</td>
<td>Call Classification: OffNet</td>
<td>No Error</td>
</tr>
<tr>
<td>Call Classification</td>
<td>OffNet</td>
<td>External Call Control Profile: &lt; None &gt;</td>
<td>Allow Device Override: Provide Outside Dial Tone</td>
<td>Allow Overlap Sending: Urgent Priority</td>
</tr>
</tbody>
</table>

**Calling Party Transformations**

- Use Calling Party's External Phone Number Mask

<table>
<thead>
<tr>
<th>Calling Party Transform Mask</th>
<th>Prefix Digits (Outgoing Calls)</th>
<th>Calling Line ID Presentation</th>
<th>Calling Name Presentation</th>
<th>Calling Party Number Type</th>
<th>Calling Party Numbering Plan</th>
</tr>
</thead>
</table>

**Connected Party Transformations**

<table>
<thead>
<tr>
<th>Connected Line ID Presentation</th>
<th>Connected Name Presentation</th>
</tr>
</thead>
</table>

**Called Party Transformations**

<table>
<thead>
<tr>
<th>Discard Digits</th>
<th>Called Party Transform Mask</th>
<th>Prefix Digits (Outgoing Calls)</th>
<th>Called Party Number Type</th>
<th>Called Party Numbering Plan</th>
</tr>
</thead>
</table>

**ISDN Network-Specific Facilities Information Element**

<table>
<thead>
<tr>
<th>Network Service Protocol</th>
<th>Network Service</th>
<th>Service Parameter Name</th>
<th>Service Parameter Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>-- Not Selected --</td>
<td>-- Not Selected --</td>
<td>&lt; Not Exist &gt;</td>
<td></td>
</tr>
</tbody>
</table>

Figure 20: Route Pattern for Voice
**Pattern Definition**

<table>
<thead>
<tr>
<th>Field</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Route Pattern *</td>
<td>14XX</td>
</tr>
<tr>
<td>Route Partition</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Description</td>
<td>RP_for_LogixCommunicationsFAX</td>
</tr>
<tr>
<td>Numbering Plan</td>
<td>-- Not Selected --</td>
</tr>
<tr>
<td>Route Filter</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>MLPF Precedence *</td>
<td>Default</td>
</tr>
<tr>
<td>Apply Call Blocking Percentage</td>
<td></td>
</tr>
<tr>
<td>Resource Priority Namespace Network Domain</td>
<td></td>
</tr>
<tr>
<td>Route Class *</td>
<td>Default</td>
</tr>
<tr>
<td>Gateway/Route List *</td>
<td>Trunk_to_FAX_gateway_for_Logix</td>
</tr>
<tr>
<td>Route Option</td>
<td>Route this pattern</td>
</tr>
</tbody>
</table>

**Calling Party Transformations**

- **Use Calling Party's External Phone Number Mask**

**Connected Party Transformations**

**Called Party Transformations**

**ISDN Network-Specific Facilities Information Element**

---

Figure 21: Route Pattern for Fax
Figure 22: Route Pattern for Anonymous Call
Figure 23: Route Pattern for Emergency Call
## Explanation

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Route Pattern</td>
<td>8.@ for Voice &amp; International Calls, 14XX for Fax Call, 8*.@ for Anonymous Calls and X11 for Operator Call and Emergency Services</td>
<td>Specify appropriate Route Pattern</td>
</tr>
<tr>
<td>Gateway/Route List</td>
<td>Trunk_to_CUBE_for_LogixCommunications for Route Pattern 8.@, 8*.@, X11 and Trunk_to_FAX_gateway_for_Logix for Route Pattern 14XX</td>
<td>SIP Trunk name configured earlier</td>
</tr>
<tr>
<td>Numbering Plan</td>
<td>NANP for Route Pattern 8.@, 8*.@</td>
<td>North American Numbering Plan</td>
</tr>
<tr>
<td>Call Classification</td>
<td>Off-Net for Route Pattern 8.@, 8*.@, 14XX and X11</td>
<td>Restrict the transferring of an external call to an external device</td>
</tr>
<tr>
<td>Discard Digits</td>
<td>PreDot for Route Pattern 8.@, 8*.@</td>
<td>Specifies how to modify digit before they are sending to Logix network</td>
</tr>
</tbody>
</table>
### Acronyms

<table>
<thead>
<tr>
<th>Acronym</th>
<th>Definitions</th>
</tr>
</thead>
<tbody>
<tr>
<td>CPE</td>
<td>Customer Premise Equipment</td>
</tr>
<tr>
<td>CUBE</td>
<td>Cisco Unified Border Element</td>
</tr>
<tr>
<td>CUCM</td>
<td>Cisco Unified Communications Manager</td>
</tr>
<tr>
<td>MTP</td>
<td>Media Termination Point</td>
</tr>
<tr>
<td>POP</td>
<td>Point of Presence</td>
</tr>
<tr>
<td>PSTN</td>
<td>Public Switched Telephone Network</td>
</tr>
<tr>
<td>SCCP</td>
<td>Skinny Client Control Protocol</td>
</tr>
<tr>
<td>SIP</td>
<td>Session Initiation Protocol</td>
</tr>
</tbody>
</table>
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