



# 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:

[http://www.cisco.com/c/en/us/td/docs/voice\\_ip\\_comm/cucm/srnd/collab10/collab10/dialplan.html](http://www.cisco.com/c/en/us/td/docs/voice_ip_comm/cucm/srnd/collab10/collab10/dialplan.html)

## Network Topology

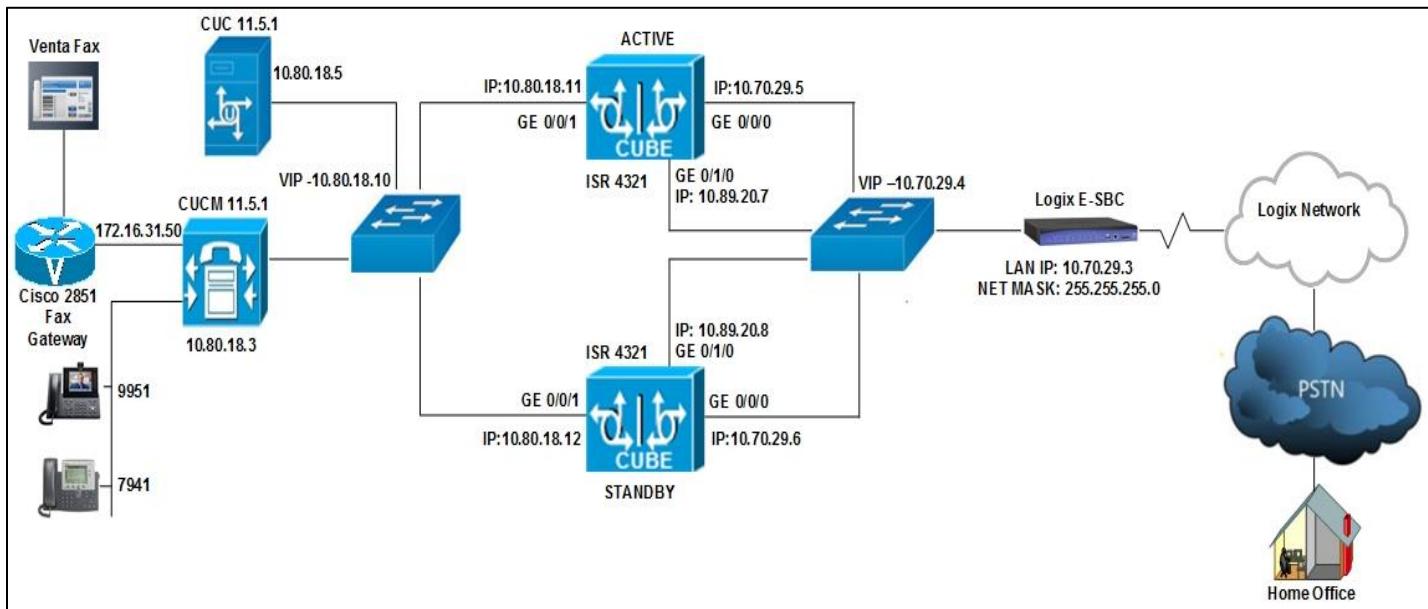


Figure 1 Network Topology

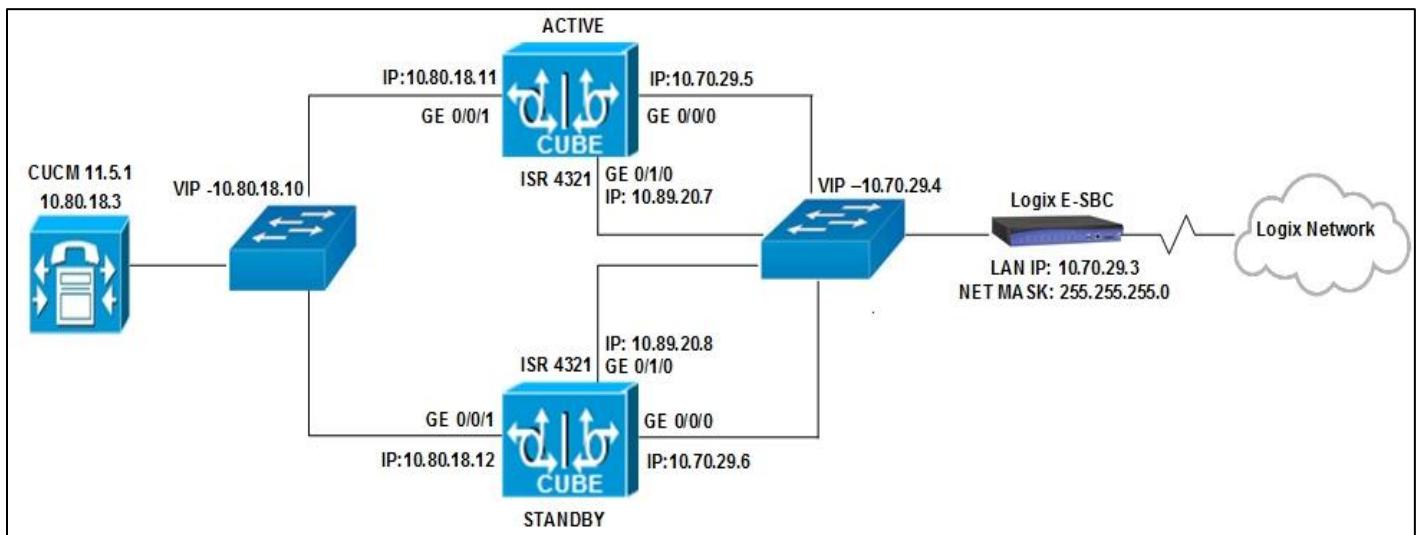


Figure 2: Cisco UBE High Availability



## 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:

```
!  
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

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



## 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-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-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.

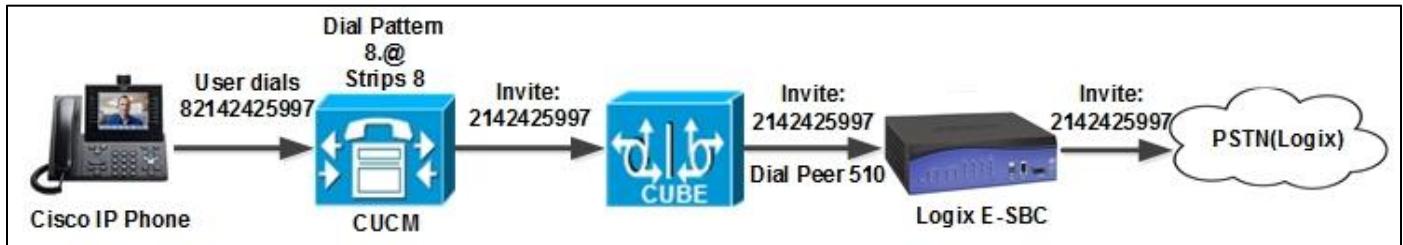


Figure 3: Outbound Voice Call

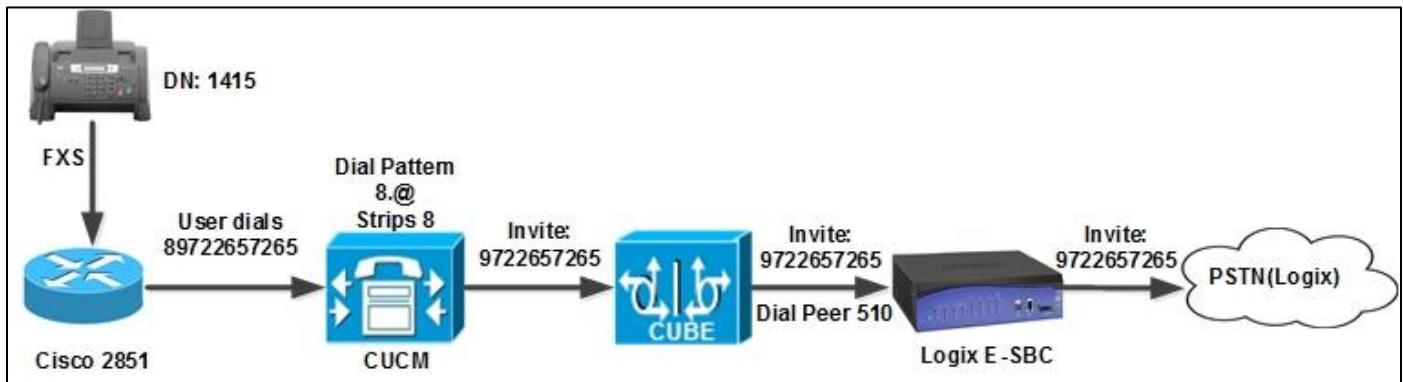


Figure 4: Outbound Fax Call

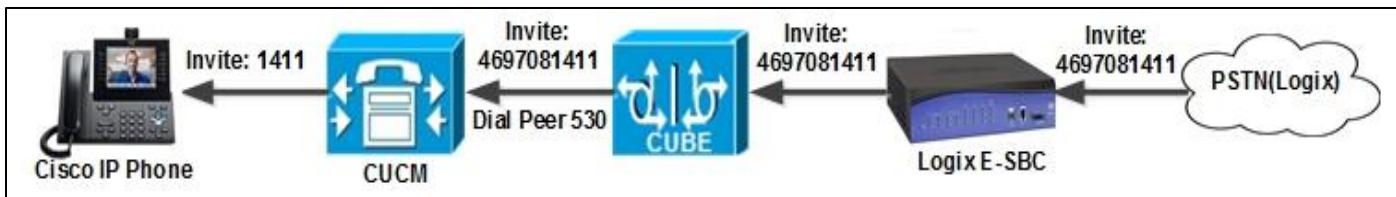


Figure 5: Inbound Voice Call

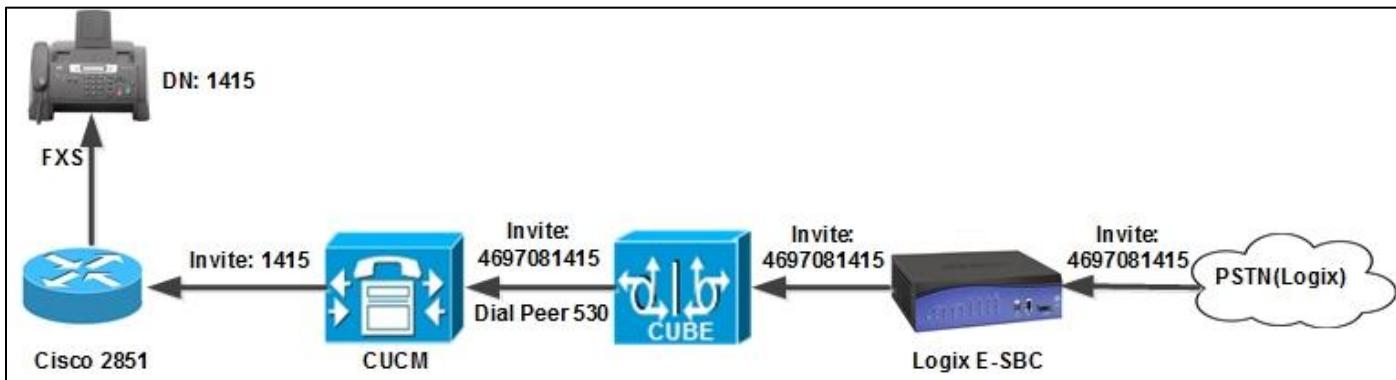


Figure 6 : Inbound Fax Call

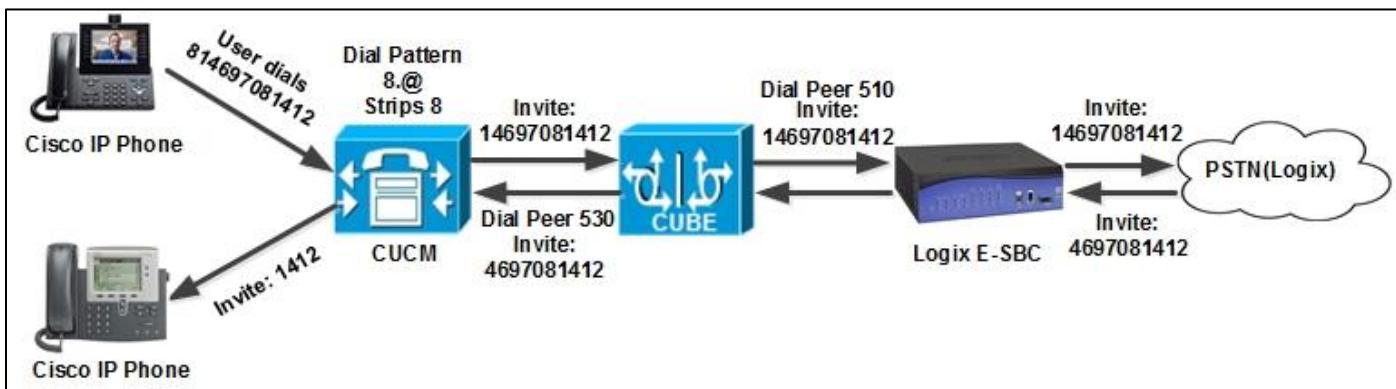


Figure 7 : PBX to PBX via Logix Call



## 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 datetime msec localtime
```

```
service timestamps log datetime 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
34345A17 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
```



```
05FD3768 ED1EF293 B3C2C1CE 4684F8F9 E920AE8F 33F4DFE0 FF04BE27 B75A28C1
6A2084C5 31BFF5C1 CD07916D 83FD56EF 9023C974 A9835AAF AC1AAF93 C0FB6856
5CA7B10A AF9EFCE1 5DE8651F D30847FF 024D6EF3 3AADB77D 68519BA9 F21AC1FE
5A50CA58 A00CDBB5 25C693E8 4D8C639D 6E5A3935 2F050F4D A3A7B2AD 47942BDD
4D78EFEE 81FDAFE0 F26220A6 6AF1D505 C601A2B3 56B2D2FE 5DD60B95 7B149AC6
EB0CACE9 CA5D42CC 4B0CA1DE 2895251A 4C1AFBC0 4FD54872 50BC69B2 445DF62E
CA556655 9BEB0203 010001A3 53305130 0F060355 1D130101 FF040530 030101FF
301F0603 551D2304 18301680 140623A5 BE29331A B5FF4081 20569978 FB29842F
58301D06 03551D0E 04160414 0623A5BE 29331AB5 FF408120 569978FB 29842F58
300D0609 2A864886 F70D0101 05050003 82010100 9A8F4A49 4CC83788 4EC24211
CE24EE7A B8552513 F9F34632 04B8119D 612FFA57 370471EF 123E1385 EBC74CBD
92DD9795 086536A0 F2469390 219B288F 3D9EC787 48A4EE78 5A492BA1 1680D1C9
F3A8A820 1D065DEB E8F0D00E 37A2A866 F759FB2D E30CD9B8 8900E25E 8171A288
FB2BB185 B6A6ED29 3BAA4495 31FCC789 0305E830 6EBA491E 211F0B7C FE808066
4E78E657 D16239B2 6E40B8A0 41631417 40EAB264 D31C2A10 24FEFD2B F0C5A1D9
693C7384 D1C54B99 BAC2A7AA 5B646CED 6E31FEC1 EB64C663 9F703970 BFA72795
06252993 E38182F3 7F760357 37556092 A5FE18F4 4FCC6BA3 716886FA 76106709
8D4EF4C5 14A81EB9 F0A29EB9 DB41CAB7 F98A5D15
```

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-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-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-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 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 494F532D 53656C66 2D536967 6E65642D 43657274

69666963 6174652D 32353438 34343332 3436301E 170D3137 30323130 31373132

33365A17 0D323030 31303130 30303030 305A3031 312F302D 06035504 03132649

4F532D53 656C662D 5369676E 65642D43 65727469 66696361 74652D32 35343834

34333234 36308201 22300D06 092A8648 86F70D01 01010500 0382010F 00308201

0A028201 0100B73A 8AE876C0 62A381D9 4C331F21 6FBF60E9 20F9420A 6F2C3A5A

2DA4B74B 1A9B55DB 65BC3A4B 016D4E96 3CB638A7 31C61AA1 A2E8EF3E FE7733F5

A0035F13 9AE153CE D55D4F64 FBBCA3CE EC8D110A 6490B2CD 44509DEE 14A60E75

66CF37C5 3DF0BBBE 7B27306D C2ACDBA2 A3497E3D 7EFDC2B 1902A0A8 038AD01E

68FD339C B2620BFF 00E703AB 88DD7796 B7C0351F 27BFF1EC 791ECF53 87B57E81

166B26BA 1428E6F7 A7484680 ACF2B8F7 BB95977B C3854F78 D4295377 CE568896

451A72F7 5D117423 6A69CEEE 8E13BDAB 96B61B29 1165C7C4 E7DB2BCB 1A2095F6

E9C80DDD 9B1DCED7 CE87C3F0 BF726628 6AF272BA 9A3D33B0 4AD1444F 87DF933F

9BCCF78D 3C6B0203 010001A3 53305130 0F060355 1D130101 FF040530 030101FF

301F0603 551D2304 18301680 14EADBB3 9378001B 592B6EC4 0CD4C83F 3FA14FA3

AD301D06 03551D0E 04160414 EADBB393 78001B59 2B6EC40C D4C83F3F A14FA3AD

300D0609 2A864886 F70D0101 05050003 82010100 A99C7D6F 325E52A8 9F0221FE

3BC2460A 5383DF63 B1A66575 D8CE62F8 725B14FA F18879B1 38173AF8 3A05D05F

B72318A1 23B11F12 E14DE814 CF93938D 41F75435 21999BC2 6FFAFBAC 84347F5F



```
0B4787B0 2A54C190 8E4A7505 3957998E 8A58C6A1 D6BBD261 11C24F7F 61FA6931
3A311AC7 E7D50544 7A6DA790 09A5E366 4635FE2F C5824130 AB3FDA56 AE0BEA53
E674A90C C13EF3B9 2C0D7F9B 3ED7CD5A 3FABB4D6 A0B9E76A 180E8DC1 E1E8024F
E431C813 64B5F09E A3BB9F8D 669E33F5 2BFB2295 34FDDE83 1290F246 61BA7FEB
943F10C5 29499BFE 173CE107 6D938477 5A4DB917 B88D2CC5 6A2F77B1 3E605311
B4A19B46 1A05F455 3E2816E0 59D13336 027F827D
```

```
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
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-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

The screenshot shows the Cisco Unified CM Administration interface. At the top, there's a navigation bar with links like System, Call Routing, Media Resources, Advanced Features, Device, Application, User Management, and Help. The user is logged in as 'administrator'. Below the header, a banner displays the system version: 'System version: 11.5.1.12900-21'. Another banner below it provides hardware information: 'VMware Installation: 1 vCPU Intel(R) Xeon(R) CPU E5-2680 0 @ 2.70GHz, disk 1: 120Gbytes, 4096Mbytes RAM, Partitions aligned'. On the right side of the page, there's a small image of a server room.

Figure 8: Cisco UCM Version

### Cisco Call Manager Service Parameters

**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

The screenshot shows the 'Service Parameters' configuration page. It starts with a 'Select Server and Service' section where 'Server\*' is set to 'Clus28Sub1--CUCM Voice/Video (Active)' and 'Service\*' is set to 'Cisco CallManager (Active)'. A note below says: 'All parameters apply only to the current server except parameters that are in the cluster-wide group(s.)'. The next section is 'Cisco CallManager (Active) Parameters on server Clus28Sub1--CUCM Voice/Video (Active)'. It contains a table with columns: Parameter Name, Parameter Value, and Suggested Value. Under the 'Call Throttling' section, the following parameters are listed:

Parameter Name	Parameter Value	Suggested Value
Code Yellow Entry Latency *	20	20
Code Yellow Exit Latency Calculation *	40	40
Code Yellow Duration *	5	5
Max Events Allowed *	2000	2000
System Throttle Sample Size *	10	10

Figure 9: Service Parameters



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

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**

SIP Trunk Security Profile Configuration

Related Links: Back To Find/List Go

SIP Trunk Security Profile Information

Name*	LogixCommunications Non Secure SIP Trunk Profile
Description	Non Secure SIP Trunk Profile authenticated by null String
Device Security Mode	Non Secure
Incoming Transport Type*	TCP+UDP
Outgoing Transport Type	UDP

Enable Digest Authentication  
Nonce Validity Time (mins)\* 600  
X.509 Subject Name  
Incoming Port\* 5060  
 Enable Application level authorization  
 Accept presence subscription  
 Accept out-of-dialog refer\*\*  
 Accept unsolicited notification  
 Accept replaces header  
 Transmit security status  
 Allow charging header  
SIP V.150 Outbound SDP Offer Filtering\* Use Default Filter

Save Delete Copy Reset Apply Config Add New

Figure 11: SIP Trunk Security Profile

### Explanation

Parameter	Value	Description
Incoming Transport Type	TCP + UDP	
Outgoing Transport Type	UDP	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.



## 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

**SIP Profile Configuration**      Related Links: [Back To Find>List](#) [Go](#)

Save Copy Apply Config

**SIP Profile Information**

Name*	LogixCommunications Standard SIP Profile
Description	Default SIP Profile
Default MTP Telephony Event Payload Type*	101
Early Offer for G.Clear Calls*	Disabled
User-Agent and Server header information*	Send Unified CM Version Information as User-Agent
Version in User Agent and Server Header*	Major And Minor
Dial String Interpretation*	Phone number consists of characters 0-9, *, #, and
Confidential Access Level Headers*	Disabled
<input type="checkbox"/> Redirect by Application	
<input type="checkbox"/> Disable Early Media on 180	
<input type="checkbox"/> Outgoing T.38 INVITE include audio mline	
<input type="checkbox"/> Offer valid IP and Send/Receive mode only for T.38 Fax Relay	
<input type="checkbox"/> Use Fully Qualified Domain Name in SIP Requests	
<input type="checkbox"/> Assured Services SIP conformance	
<input type="checkbox"/> Enable External QoS**	

**SDP Information**

SDP Session-level Bandwidth Modifier for Early Offer and Re-invites*	TIAS and AS
SDP Transparency Profile	Pass all unknown SDP attributes
Accept Audio Codec Preferences in Received Offer*	Default
<input type="checkbox"/> Require SDP Inactive Exchange for Mid-Call Media Change	
<input type="checkbox"/> Allow RR/RS bandwidth modifier (RFC 3556)	

**Parameters used in Phone**

Timer Invite Expires (seconds)*	180
Timer Register Delta (seconds)*	5
Timer Register Expires (seconds)*	3600
Timer T1 (msec)*	500
Timer T2 (msec)*	4000
Retry INVITE*	6
Retry Non-INVITE*	10
Media Port Ranges	<input checked="" type="radio"/> Common Port Range for Audio and Video <input type="radio"/> Separate Port Ranges for Audio and Video

Figure 12: SIP Profile



Start Media Port*	16384				
Stop Media Port*	32766				
DSCP for Audio Calls	Use System Default				
DSCP for Video Calls	Use System Default				
DSCP for Audio Portion of Video Calls	Use System Default				
DSCP for TelePresence Calls	Use System Default				
DSCP for Audio Portion of TelePresence Calls	Use System Default				
Call Pickup URI*	x-cisco-serviceuri-pickup				
Call Pickup Group Other URI*	x-cisco-serviceuri-opickup				
Call Pickup Group URI*	x-cisco-serviceuri-gpickup				
Meet Me Service URI*	x-cisco-serviceuri-meetme				
User Info*	None				
DTMF DB Level*	Nominal				
Call Hold Ring Back*	Off				
Anonymous Call Block*	Off				
Caller ID Blocking*	Off				
Do Not Disturb Control*	User				
Telnet Level for 7940 and 7960*	Disabled				
Resource Priority Namespace	< None >				
Timer Keep Alive Expires (seconds)*	120				
Timer Subscribe Expires (seconds)*	120				
Timer Subscribe Delta (seconds)*	5				
Maximum Redirections*	70				
Off Hook To First Digit Timer (milliseconds)*	15000				
Call Forward URI*	x-cisco-serviceuri-cfwdall				
Speed Dial (Abbreviated Dial) URI*	x-cisco-serviceuri-abbrdial				
<input checked="" type="checkbox"/> Conference Join Enabled					
<input type="checkbox"/> RFC 2543 Hold					
<input checked="" type="checkbox"/> Semi Attended Transfer					
<input type="checkbox"/> Enable VAD					
<input type="checkbox"/> Stutter Message Waiting					
<input type="checkbox"/> MLPP User Authorization					
<b>Normalization Script</b>					
Normalization Script	< None >				
<input type="checkbox"/> Enable Trace					
<table border="1"><thead><tr><th>Parameter Name</th><th>Parameter Value</th></tr></thead><tbody><tr><td>1</td><td></td></tr></tbody></table>		Parameter Name	Parameter Value	1	
Parameter Name	Parameter Value				
1					
	<input type="button" value="+"/> <input type="button" value="-"/>				

Figure 13: SIP Profile (Cont.)



**Incoming Requests FROM URI Settings**

Caller ID DN	<input type="text"/>
Caller Name	<input type="text"/>

**Trunk Specific Configuration**

Reroute Incoming Request to new Trunk based on*	Never
Resource Priority Namespace List	< None >
SIP Rel1XX Options*	Send PRACK if 1xx Contains SDP
Video Call Traffic Class*	Mixed
Calling Line Identification Presentation*	Default
Session Refresh Method*	Invite
Early Offer support for voice and video calls*	Disabled (Default value)

Enable ANAT  
 Deliver Conference Bridge Identifier  
 Allow Passthrough of Configured Line Device Caller Information  
 Reject Anonymous Incoming Calls  
 Reject Anonymous Outgoing Calls  
 Send ILS Learned Destination Route String  
 Connect Inbound Call before Playing Queuing Announcement

**SIP OPTIONS Ping**

<input checked="" type="checkbox"/> Enable OPTIONS Ping to monitor destination status for Trunks with Service Type "None (Default)"	
Ping Interval for In-service and Partially In-service Trunks (seconds)*	60
Ping Interval for Out-of-service Trunks (seconds)*	120
Ping Retry Timer (milliseconds)*	500
Ping Retry Count*	6

**SDP Information**

<input type="checkbox"/> Send send-receive SDP in mid-call INVITE
<input type="checkbox"/> Allow Presentation Sharing using BFCP
<input type="checkbox"/> Allow iX Application Media
<input type="checkbox"/> Allow multiple codecs in answer SDP

Figure 14: SIP Profile (Cont.)

## Explanation

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



## SIP Trunk Configuration

Create SIP trunks to Cisco UBE

Navigation Path: Device → Trunk

Trunks (1 - 9 of 9)												Rows per Page 25			
Find Trunks where Device Name begins with												Find	Clear Filter	Add	Remove
Select item or enter search text															
#	Name	Description	Calling Search Space	Device Pool	Route Pattern	Partition	Route Group	Priority	Trunk Type	SIP Trunk Status	SIP Trunk Duration	SIP Trunk Security Profile			
<input type="checkbox"/>	<a href="#">Trunk_to_FAX_gateway_for_Logix</a>	Trunk_to_FAX_gateway_for_Logix		<a href="#">LogixCommunication_Devicepool</a>	14XX				SIP Trunk	Full Service	Time In Full Service: 0 day 18 hours 22 minutes	<a href="#">LogixCommunications Non Secure SIP Trunk Profile</a>			
<input type="checkbox"/>	<a href="#">Trunk_to_CUBE_for_LogixCommunications</a>	Trunk_to_CUBE_for_LogixCommunications		<a href="#">LogixCommunication_Devicepool</a>	8*.@				SIP Trunk	Full Service	Time In Full Service: 6 days 19 hours 21 minutes	<a href="#">LogixCommunications Non Secure SIP Trunk Profile</a>			
<input type="checkbox"/>	<a href="#">Trunk_to_CUBE_for_LogixCommunications</a>	Trunk_to_CUBE_for_LogixCommunications		<a href="#">LogixCommunication_Devicepool</a>	8.*@				SIP Trunk	Full Service	Time In Full Service: 6 days 19 hours 21 minutes	<a href="#">LogixCommunications Non Secure SIP Trunk Profile</a>			
<input type="checkbox"/>	<a href="#">Trunk_to_CUBE_for_LogixCommunications</a>	Trunk_to_CUBE_for_LogixCommunications		<a href="#">LogixCommunication_Devicepool</a>	X11				SIP Trunk	Full Service	Time In Full Service: 6 days 19 hours 21 minutes	<a href="#">LogixCommunications Non Secure SIP Trunk Profile</a>			
<input type="checkbox"/>	<a href="#">Logix_Trunk_to_Unity</a>	Logix_Trunk_to_Unity		<a href="#">LogixCommunication_Devicepool</a>	4000				SIP Trunk	Full Service	Time In Full Service: 2 days 3 hours 7 minutes	<a href="#">Non Secure SIP Trunk Profile Unity Connection</a>			

Figure 15: SIP Trunks List



**SIP Trunk Status**

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

**Device Information**

Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
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
<input type="checkbox"/> Media Termination Point Required	
<input checked="" type="checkbox"/> Retry Video Call as Audio	
<input type="checkbox"/> Path Replacement Support	
<input type="checkbox"/> Transmit UTF-8 for Calling Party Name	
<input type="checkbox"/> Transmit UTF-8 Names in QSIG APDU	
<input type="checkbox"/> Unattended Port	
<input type="checkbox"/> SRTP Allowed - When this flag is checked, Encrypted TLS needs to be configured in the network to provide end to end security. Failure to do so will expose keys and other information.	
Consider Traffic on This Trunk Secure*	When using both sRTP and TLS
Route Class Signaling Enabled*	Default
Use Trusted Relay Point*	Default
<input checked="" type="checkbox"/> PSTN Access	
<input type="checkbox"/> Run On All Active Unified CM Nodes	

Figure 16: SIP Trunk to Cisco UBE



**-MLPP and Confidential Access Level Information-**

MLPP Domain < None >  
Confidential Access Mode < None >  
Confidential Access Level < None >

**Call Routing Information**

Remote-Party-Id  
 Asserted-Identity  
Asserted-Type\* Default  
SIP Privacy\* Default

**Inbound Calls**

Significant Digits\* 4  
Connected Line ID Presentation\* Default  
Connected 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.

**Clear Prefix Settings** **Default Prefix Settings**

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	Default	0	< None >	<input checked="" type="checkbox"/>

**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.

**Clear Prefix Settings** **Default Prefix Settings**

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	Default	0	< None >	<input checked="" type="checkbox"/>

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



**SIP Information**

**Destination**

Destination Address is an SRV

	Destination Address	Destination Address IPv6	Destination Port
1*	10.80.18.10		5060

MTP Preferred Originating Codec\* 711ulaw

BLF Presence Group\* Standard Presence group

SIP Trunk Security Profile\* LogixCommunications Non Secure SIP Trunk Profile

Rerouting Calling Search Space < None >

Out-Of-Dialog Refer Calling Search Space < None >

SUBSCRIBE Calling Search Space < None >

SIP Profile\* LogixCommunications Standard SIP Profile [View Details](#)

DTMF Signaling Method\* No Preference

**Normalization Script**

Normalization Script < None >

Enable Trace

Parameter Name	Parameter Value
1	

**Recording Information**

None

This trunk connects to a recording-enabled gateway

This trunk connects to other clusters with recording-enabled gateways

**Geolocation Configuration**

Geolocation < None >

Geolocation Filter < None >

Send Geolocation Information

**Action Buttons**

Figure 18: SIP Trunk to Cisco UBE (Cont.)



## Explanation

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



## Dial Plan

### 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
  - o "8" is removed before sending to Cisco UBE
2. For FAX call, Access Code "8"+10 digits number is used at Cisco Fax gateway
  - o "8" is removed at Cisco UCM
  - o 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
  - o "8\*" is removed at Cisco UCM
  - o 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 (1 - 8 of 8)							Rows per Page <select>50</select>
<input type="text"/> Find Route Patterns where <input type="text"/> Pattern <input type="button"/> begins with <input type="button"/> Find <input type="button"/> Clear Filter <input type="button"/> <input type="button"/>							
<input type="checkbox"/>	Pattern <input type="button"/>	Description	Partition	Route Filter	Associated Device	<input type="button"/> Copy	
<input type="checkbox"/>	<a href="#">14XX</a>	RP_for_LogixCommunicationsFAX			<a href="#">Trunk to FAX gateway for Logix</a>	<input type="button"/>	
<input type="checkbox"/>	<a href="#">8*._@</a>	RP_for_LogixCommunications-for Anonymous Calls			<a href="#">Trunk to CUBE for LogixCommunications</a>	<input type="button"/>	
<input type="checkbox"/>	<a href="#">8._@</a>	RP_for_LogixCommunications			<a href="#">Trunk to CUBE for LogixCommunications</a>	<input type="button"/>	
<input type="checkbox"/>	<a href="#">X11</a>	Emergency Calling			<a href="#">Trunk to CUBE for LogixCommunications</a>	<input type="button"/>	

Figure 19: Route Patterns List



**-Pattern Definition-**

Route Pattern*	8.@[		
Route Partition	< None >		
Description	RP_for_LogixCommunications		
Numbering Plan*	NANP		
Route Filter	< None >		
MLPP Precedence*	Default		
<input type="checkbox"/> Apply Call Blocking Percentage			
Resource Priority Namespace Network Domain	< None >		
Route Class*	Default		
Gateway/Route List*	Trunk_to_CUBE_for_LogixCommunications		
<a href="#">(Edit)</a>			
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern No Error		
Call Classification*	OffNet		
External Call Control Profile	< None >		
<input type="checkbox"/> Allow Device Override	<input checked="" type="checkbox"/> Provide Outside Dial Tone	<input type="checkbox"/> Allow Overlap Sending	<input type="checkbox"/> Urgent Priority
<input type="checkbox"/> Require Forced Authorization Code			
Authorization Level*	0		
<input type="checkbox"/> Require Client Matter Code			
<input checked="" type="checkbox"/> Use Calling Party's External Phone Number Mask			
Calling Party Transform Mask			
Prefix Digits (Outgoing Calls)			
Calling Line ID Presentation*	Default		
Calling Name Presentation*	Default		
Calling Party Number Type*	Cisco CallManager		
Calling Party Numbering Plan*	Cisco CallManager		
Connected Line ID Presentation*	Default		
Connected Name Presentation*	Default		
Discard Digits	PreDot		
Called Party Transform Mask			
Prefix Digits (Outgoing Calls)			
Called Party Number Type*	Cisco CallManager		
Called Party Numbering Plan*	Cisco CallManager		
Network Service Protocol	-- Not Selected --		
Carrier Identification Code			
Network Service	Service Parameter Name	Service Parameter Value	
-- Not Selected --	< Not Exist >		

Figure 20: Route Pattern for Voice



**Pattern Definition**

Route Pattern*	14XX		
Route Partition	< None >		
Description	RP_for_LogixCommunicationsFAX		
Numbering Plan	-- Not Selected --		
Route Filter	< None >		
MLPP Precedence*	Default		
<input type="checkbox"/> Apply Call Blocking Percentage			
Resource Priority Namespace Network Domain	< None >		
Route Class*	Default		
Gateway/Route List*	Trunk_to_FAX_gateway_for_Logix <a href="#">(Edit)</a>		
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern No Error		
Call Classification*	OffNet		
External Call Control Profile	< None >		
<input type="checkbox"/> Allow Device Override	<input checked="" type="checkbox"/> Provide Outside Dial Tone	<input type="checkbox"/> Allow Overlap Sending	<input type="checkbox"/> Urgent Priority
<input type="checkbox"/> Require Forced Authorization Code			
Authorization Level*	0		
<input type="checkbox"/> Require Client Matter Code			

**Calling Party Transformations**

<input checked="" type="checkbox"/> Use Calling Party's External Phone Number Mask	
Calling Party Transform Mask	
Prefix Digits (Outgoing Calls)	
Calling Line ID Presentation*	Default
Calling Name Presentation*	Default
Calling Party Number Type*	Cisco CallManager
Calling Party Numbering Plan*	Cisco CallManager

**Connected Party Transformations**

Connected Line ID Presentation*	Default
Connected Name Presentation*	Default

**Called Party Transformations**

Discard Digits	< None >
Called Party Transform Mask	469708XXXX
Prefix Digits (Outgoing Calls)	
Called Party Number Type*	Cisco CallManager
Called Party Numbering Plan*	Cisco CallManager

**ISDN Network-Specific Facilities Information Element**

Network Service Protocol	-- Not Selected --	
Carrier Identification Code		
Network Service	Service Parameter Name	Service Parameter Value
-- Not Selected --	< Not Exist >	

Figure 21: Route Pattern for Fax



**Pattern Definition**

Route Pattern*	8*.*@		
Route Partition	< None >		
Description	RP_for_LogixCommunications-for Anonymous Calls		
Numbering Plan*	NANP		
Route Filter	< None >		
MLPP Precedence*	Default		
<input type="checkbox"/> Apply Call Blocking Percentage			
Resource Priority Namespace Network Domain	< None >		
Route Class*	Default		
Gateway/Route List*	Trunk_to_CUBE_for_LogixCommunications		
<a href="#">(Edit)</a>			
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern No Error		
Call Classification*	OffNet		
External Call Control Profile	< None >		
<input type="checkbox"/> Allow Device Override	<input checked="" type="checkbox"/> Provide Outside Dial Tone	<input type="checkbox"/> Allow Overlap Sending	<input type="checkbox"/> Urgent Priority
<input type="checkbox"/> Require Forced Authorization Code			
Authorization Level*	0		
<input type="checkbox"/> Require Client Matter Code			
<input checked="" type="checkbox"/> Use Calling Party's External Phone Number Mask			
Calling Party Transform Mask			
Prefix Digits (Outgoing Calls)			
Calling Line ID Presentation*	Restricted		
Calling Name Presentation*	Restricted		
Calling Party Number Type*	Cisco CallManager		
Calling Party Numbering Plan*	Cisco CallManager		
Connected Line ID Presentation*	Default		
Connected Name Presentation*	Default		
Discard Digits	PreDot		
Called Party Transform Mask			
Prefix Digits (Outgoing Calls)			
Called Party Number Type*	Cisco CallManager		
Called Party Numbering Plan*	Cisco CallManager		
Network Service Protocol	-- Not Selected --		
Carrier Identification Code			
Network Service	Service Parameter Name	Service Parameter Value	
-- Not Selected --	< Not Exist >		

Figure 22: Route Pattern for Anonymous Call



**Pattern Definition**

Route Pattern*	X11		
Route Partition	< None >		
Description	Emergency Calling		
Numbering Plan	-- Not Selected --		
Route Filter	< None >		
MLPP Precedence*	Default		
<input type="checkbox"/> Apply Call Blocking Percentage			
Resource Priority Namespace Network Domain	< None >		
Route Class*	Default		
Gateway/Route List*	Trunk_to_CUBE_for_LogixCommunications <a href="#">(Edit)</a>		
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern No Error		
Call Classification*	OffNet		
External Call Control Profile	< None >		
<input type="checkbox"/> Allow Device Override	<input checked="" type="checkbox"/> Provide Outside Dial Tone	<input type="checkbox"/> Allow Overlap Sending	<input type="checkbox"/> Urgent Priority
<input type="checkbox"/> Require Forced Authorization Code			
Authorization Level*	0		
<input type="checkbox"/> Require Client Matter Code			

**Calling Party Transformations**

<input checked="" type="checkbox"/> Use Calling Party's External Phone Number Mask	
Calling Party Transform Mask	
Prefix Digits (Outgoing Calls)	
Calling Line ID Presentation*	Default
Calling Name Presentation*	Default
Calling Party Number Type*	Cisco CallManager
Calling Party Numbering Plan*	Cisco CallManager

**Connected Party Transformations**

Connected Line ID Presentation*	Default
Connected Name Presentation*	Default

**Called Party Transformations**

Discard Digits	< None >
Called Party Transform Mask	
Prefix Digits (Outgoing Calls)	
Called Party Number Type*	Cisco CallManager
Called Party Numbering Plan*	Cisco CallManager

**ISDN Network-Specific Facilities Information Element**

Network Service Protocol	-- Not Selected --	
Carrier Identification Code		
Network Service	Service Parameter Name	Service Parameter Value
-- Not Selected --	< Not Exist >	

Figure 23: Route Pattern for Emergency Call



## Explanation

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



## Acronyms

Acronym	Definitions
CPE	Customer Premise Equipment
CUBE	Cisco Unified Border Element
CUCM	Cisco Unified Communications Manager
MTP	Media Termination Point
POP	Point of Presence
PSTN	Public Switched Telephone Network
SCCP	Skinny Client Control Protocol
SIP	Session Initiation Protocol



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