



DIDforSale SIP Trunking:

Cisco Unified Communications Manager 12.0.1 with Cisco Unified Border Element (CUBE 12.1.0) on ISR 4331 [IOS-XE – 16.08.01] using SIP

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Introduction

Service Providers today, such as DIDforSale, 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.

DIDforSale is a service provider offering 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 DIDforSale network, Cisco Unified Border Element (Cisco UBE) ISR 4331/K9 running IOS-XE 16.08.01 can be used. The Cisco Unified Border Element 16.08.01 provides demarcation, security, interworking and session control services for Cisco Unified Communications Manager 12.0.1 connected to DIDforSale IP network.

This document assumes the reader is knowledgeable with the terminology and configuration of Cisco UCM (Cisco Unified Communications Manager). Only configuration settings specifically required for DIDforSale 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 a Cisco Unified Communications Manager (Cisco UCM) 12.0.1 and Cisco Unified Border Element (Cisco UBE) on ISR 4331/K9 [IOS-XE – 16.08.01] connectivity to DIDforSale SIP Trunking service. The deployment model covered in this application note is CPE (Cisco UCM 12.0.1) to PSTN (DIDforSale).
- Testing was performed in accordance to DIDforSale 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 DIDforSale 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 DIDforSale 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

Network Topology

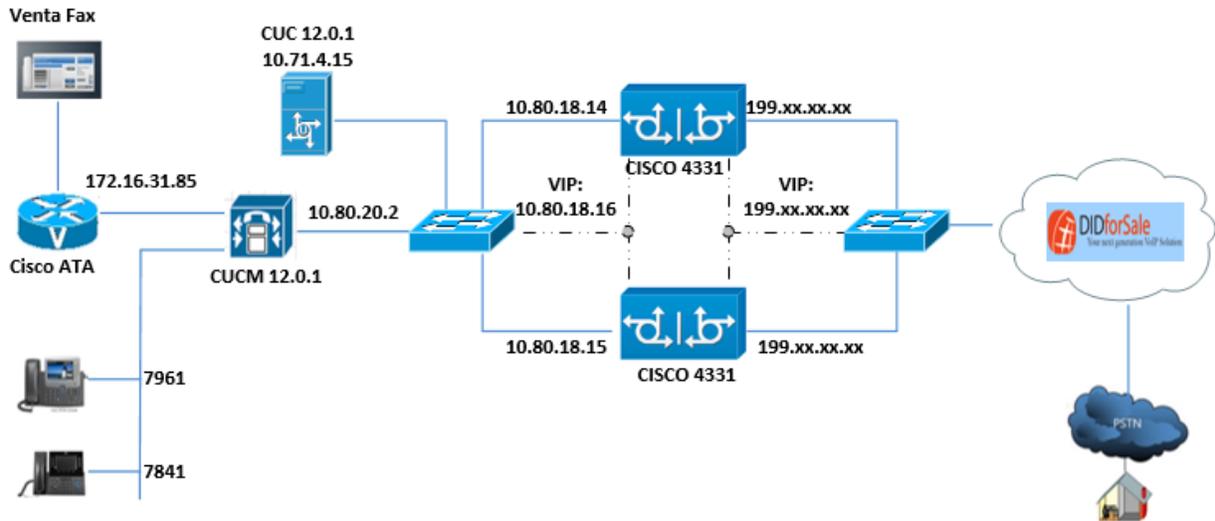


Figure 1: Network Topology

- Cisco IP Phones 7942, 7961 and 7841 phones are the devices primarily used throughout the testing to place or receive calls
- VentaFax Soft Client is used to perform all fax related scenarios. The fax client is connected to SIP Gateway (Cisco ATA) via FXS port which in turn communicates with Cisco UCM over SIP.



System Components

Hardware Requirements

- Cisco UCS-C240-M3S VMWare host running ESXi 5.5 Standard
- Cisco ISR 4331/K9 router as CUBE
- Cisco ATA SPA112
- IP phones 7841 (SIP), 7961 (SCCP) and 7942 (SCCP)

Software Requirements

- Cisco Unified Communications Manager 12.0.1
- Cisco Unity Connection 12.0.1
- IOS-XE 16.08.01 for ISR 4331/K9 Cisco Unified Border Element
- Firmware Version 1.3.5 (004p) for Cisco ATA SPA112

Features

Features Supported

- Incoming and outgoing off-net calls using G711ULaw and G729.
- Call hold
- Call transfer (unattended and attended)
- Call forward (all, busy and no answer)
- Calling Line (number) Identification Presentation (CLIP)
- Calling Line (number) Identification Restriction (CLIR)
- DTMF (RFC2833)
- Media flow-through on Cisco UBE
- Fax (G.711 pass-through) and T.38
-

Features Not Supported

- Cisco IP phones used in this test do not support blind transfer

Caveats

- 911 emergency call has been tested with only with G.711 as voice codec.
- CLID is not updated on PSTN phones for call transfer (attended and unattended) to OffNet PSTN scenarios. Caller ID is not updated at PSTN once transfer is completed by PBX.
- Network sends invite without "+" for the inbound call. Network will supports E.164 Numbering Plan for outbound call.
- In midcall re-invite during conference call, CUCM sends invite without SDP to the network and the network responded back with "200 OK "without SDP. With that CUCM is sends "BYE" with cause code=47 (resource unavailable) and terminates the call. This issue has been resolved by adding "voice-class sip midcall-signaling pass-through media-change command in dial-peer. .
- For T.38 FAX, Invite coming from the network is adding "+1" in the "FROM" and "TO" header. For the outbound T.38 FAX, network responds back with "200 Ok " to the initial "INVITE" from the FAX ATA with adding "+1" in the "contact" header. However FAX pages transmitted successfully.



Configuration

Configuring 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 - for LAN and WAN.

```
interface GigabitEthernet0/0/0
description DIDforsale CUBE1 WAN
ip address 199.XXX.XXX.XX 255.255.255.224
negotiation auto
redundancy rii 4
redundancy group 2 ip 199.XXX.XXX.XX exclusive
```

!

```
interface GigabitEthernet0/0/1
description DIDforsale CUBE1 LAN
ip address 10.80.18.14 255.255.255.0
negotiation auto
redundancy rii 3
redundancy group 2 ip 10.80.18.16 exclusive
```

!



Global Cisco UBE Settings

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

```
voice service voip
ip address trusted list
  ipv4 0.0.0.0 0.0.0.0
no ip address trusted authenticate
address-hiding
mode border-element license capacity 20
allow-connections sip to sip
redundancy-group 2
supplementary-service media-renegotiate
redirect ip2ip
fax protocol pass-through g711 ulaw
sip
  session refresh
  asserted-id pai
  privacy pstn
  early-offer forced
  privacy-policy passthru
  g729 annexb-all
  pass-thru subscribe-notify-events 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
pass-thru subscribe-notify-events all	This command is to configure pass-through for all SUBSCRIBE-NOTIFY events

Codecs

G711ulaw is used as the preferred codec for this testing and changed the preferences according to the test plan description

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

Dial Peer

Cisco UBE uses dial-peers to route the call accordingly based on the digits

```
!
dial-peer voice 222 voip
  description outbound-from PBX-PSTN - LAN facing
  destination-pattern 1626313....
  session protocol sipv2
  session target ipv4:10.80.20.2:5060
  voice-class codec 1
  voice-class sip early-offer forced
  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 disable
fax protocol pass-through g711alaw
no vad
!
dial-peer voice 111 voip
description Inbound peer match FROM DIDFORSALE
session protocol sipv2
incoming uri via trunk1
voice-class codec 1
voice-class sip options-keepalive
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 disable
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 201 voip
description Inbound peer match FROM CUCM
session protocol sipv2
incoming called-number .T
voice-class codec 1
no voice-class sip early-offer forced
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
voice-class sip midcall-signaling passthru media-change
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate 14400
fax nsf 000000
```



```
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 101 voip
description Outgoing Call from PBX to PSTN-WAN facing
destination-pattern .T
session protocol sipv2
session server-group 1
voice-class codec 1
voice-class sip early-offer forced
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
voice-class sip midcall-signaling passthru media-change
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate disable
fax protocol pass-through g711alaw
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 “21” 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 “2”. A “2.@” 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 DIDforSale, Caller dial 2 prefix followed by the target 1+10-digits number, 2 was stripped and the remaining digits were send to Cisco UBE, Cisco UBE pass the DID under Dial Peer 101 and send to DIDforSale network which will direct back to Cisco UBE and handled same as normal incoming PSTN call.

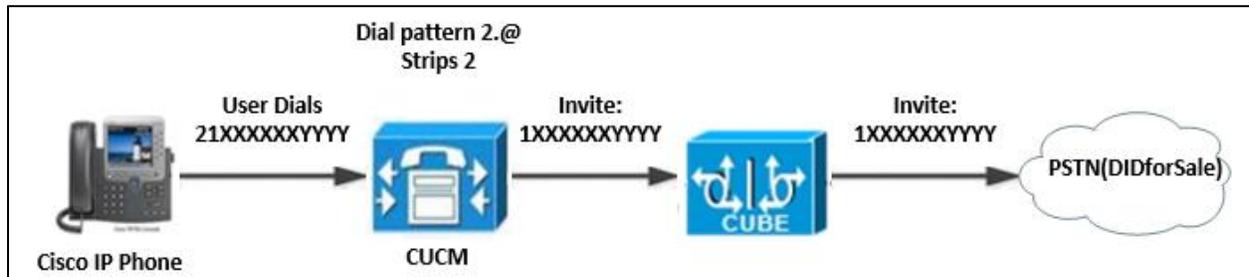


Figure 2: Outbound Voice Call

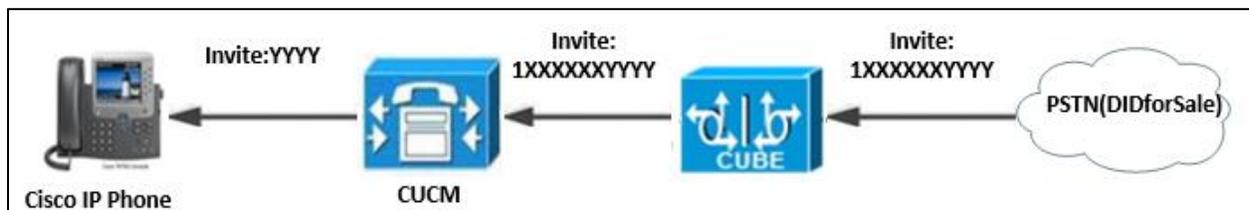


Figure 3: Inbound Voice Call

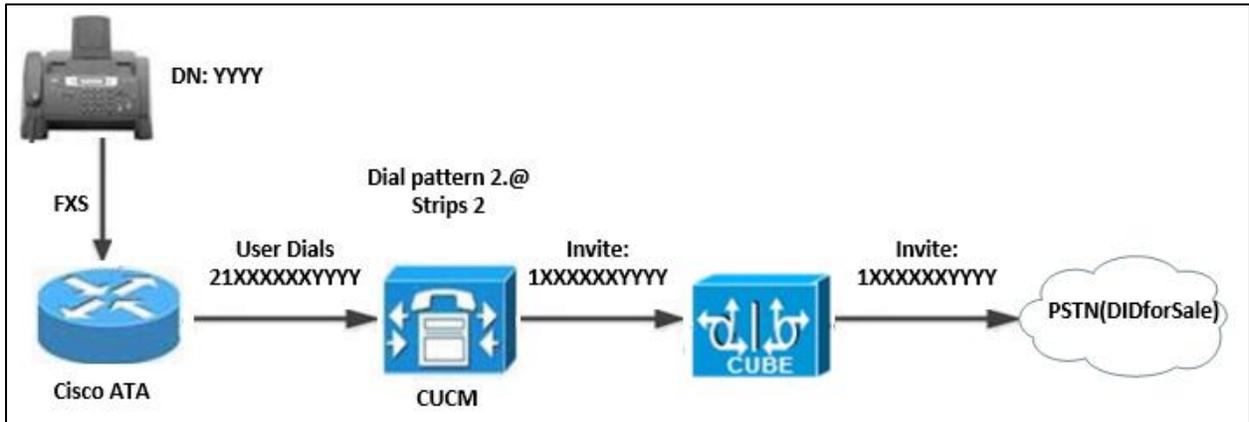


Figure 4: Outbound Fax Call

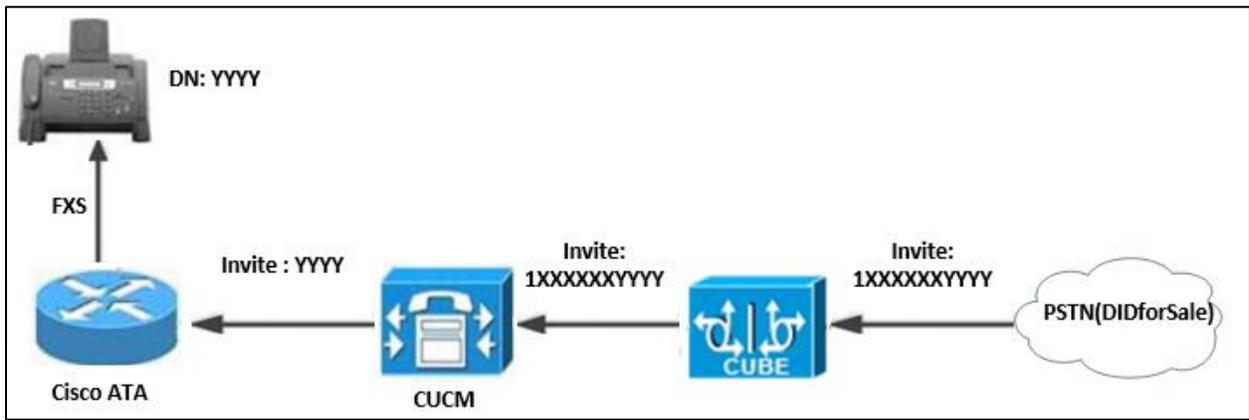


Figure 5: Inbound Fax Call

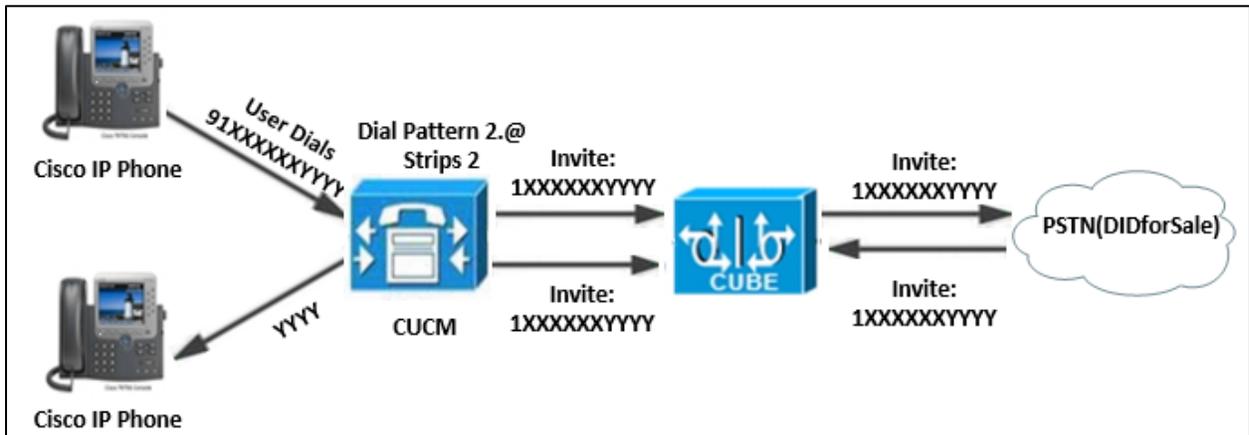


Figure 6: PBX to PBX via DIDforSale Call



Configuration Example

The following configuration snippet contains a sample configuration of Cisco UBE with all parameters mentioned previously

Active Cisco UBE

```
cube9#sh run
!
version 16.8
service timestamps debug datetime msec
service timestamps log datetime msec
platform qfp utilization monitor load 80
no platform punt-keepalive disable-kernel-core
!
hostname cube9
!
boot-start-marker
boot-end-marker
!
vrf definition Mgmt-intf
!
address-family ipv4
exit-address-family
!
address-family ipv6
exit-address-family
!
no aaa new-model
subscriber templating
multilink bundle-name authenticated
!
crypto pki trustpoint TP-self-signed-3793611302
```



```
enrollment selfsigned
subject-name cn=IOS-Self-Signed-Certificate-3793611302
revocation-check none
rsakeypair TP-self-signed-3793611302
!
voice service voip
ip address trusted list
  ipv4 0.0.0.0 0.0.0.0
no ip address trusted authenticate
address-hiding
mode border-element license capacity 20
allow-connections sip to sip
redundancy-group 2
supplementary-service media-renegotiate
redirect ip2ip
fax protocol pass-through g711ulaw
sip
  session refresh
  asserted-id pai
  privacy pstn
  early-offer forced
  privacy-policy passthru
  g729 annexb-all
  pass-thru subscribe-notify-events all
!
voice class uri trunk1 sip
  host ipv4: 209.XXX.X.XXX
  host ipv4: 209.XXX.X.XXX
  host ipv4: 209.XXX.X.XXX
  host ipv4: 209.XXX.X.XXX
```



```
host ipv4: 209.XXX.X.XXX
host ipv4: 209.XXX.XX.XX
voice class codec 2
  codec preference 1 g711alaw
  codec preference 2 g711ulaw
!
voice class codec 1
  codec preference 2 g729r8
  codec preference 3 g711alaw
  codec preference 4 g711ulaw
!
voice class server-group 1
  ipv4 209.XXX.X.XXX preference 1
  ipv4 209.XXX.X.XXX preference 1
!
voice translation-rule 1
  rule 1 ^(\^.....$)/ /1\1/
!
!
voice translation-profile coverting_to_11Digits
  translate called 1
!
voice-card 0/4
  no watchdog
!
license udi pid ISR4331/K9 sn FDO41381F1G
```



```
no license smart enable
diagnostic bootup level minimal
!
spanning-tree extend system-id
!
!
!
!
redundancy
mode none
application redundancy
group 2
name Voice-b2bha_DIDforsale
priority 100 failover threshold 75
timers delay 30 reload 60
control GigabitEthernet0/0/2 protocol 1
data GigabitEthernet0/0/2
track 1 shutdown
track 2 shutdown
!
!
!
track 1 interface GigabitEthernet0/0/0 line-protocol
!
track 2 interface GigabitEthernet0/0/1 line-protocol
!
!
!
!
!
!
```



```
interface GigabitEthernet0/0/0
description DIDforsale CUBE1 WAN
ip address 199.XXX.XXX.XX 255.255.255.224
shutdown
negotiation auto
redundancy rii 4
redundancy group 2 ip 199.XXX.XXX.XX.26 exclusive
!
interface GigabitEthernet0/0/1
description DIDforsale CUBE1 LAN
ip address 10.80.18.14 255.255.255.0
negotiation auto
redundancy rii 3
redundancy group 2 ip 10.80.18.16 exclusive
!
interface GigabitEthernet0/0/2
description CUBE HA
ip address 10.89.20.7 255.255.255.0
negotiation auto
!
interface Service-Engine0/4/0
!
interface GigabitEthernet0
vrf forwarding Mgmt-intf
no ip address
shutdown
negotiation auto
!
ip forward-protocol nd
ip http server
ip http authentication local
```



```
ip http secure-server
ip tftp source-interface GigabitEthernet0
ip route 0.0.0.0 0.0.0.0 199.182.124.1
ip route 10.64.0.0 255.255.0.0 10.80.18.1
ip route 172.16.24.0 255.255.248.0 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 201 voip
description Inbound peer match FROM CUCM
session protocol sipv2
incoming called-number .T
voice-class codec 1
no voice-class sip early-offer forced
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 disable
fax protocol pass-through g711alaw
no vad
!
dial-peer voice 101 voip
description Outgoing Call from PBX to PSTN-WAN facing
destination-pattern .T
session protocol sipv2
session server-group 1
voice-class codec 1
voice-class sip early-offer forced
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
voice-class sip midcall-signaling passthru media-change
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate disable
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 222 voip
description outbound-from PBX-PSTN - LAN facing
destination-pattern 1626313....
session protocol sipv2
session target ipv4:10.80.20.2:5060
voice-class codec 1
voice-class sip early-offer forced
```



```
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 disable
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 111 voip
description Inbound peer match FROM DIDFORSALE
session protocol sipv2
incoming uri via trunk1
voice-class codec 1
voice-class sip options-keepalive
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 disable
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 203 voip
description Inbound peer match FROM CUCM
shutdown
session protocol sipv2
incoming called-number .T
voice-class codec 1
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 protocol pass-through g711ulaw
no vad
!
dial-peer voice 1010 voip
description Outgoing Call from PBX to PSTN-WAN facing
shutdown
destination-pattern 12T
session protocol sipv2
session target ipv4:209.XXX.XX.XX:5060
voice-class codec 1
voice-class sip early-offer forced
voice-class sip options-keepalive
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 disable
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 1000 voip
description PBX to PBX call via DIDforsale - WAN facing
translation-profile outgoing coverting_to_11Digits
destination-pattern 626313....
session protocol sipv2
session target ipv4: 209.XXX.XX.XX:5060
voice-class codec 1
voice-class sip early-offer forced
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 disable
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 1002 voip
description PBX to PBX call via DIDforsale-SecondTrunk - WAN facing
translation-profile outgoing coverting_to_11Digits
destination-pattern 626313...
session protocol sipv2
session target ipv4: 209.XXX.XX.XX:5060
voice-class codec 1
voice-class sip early-offer forced
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 disable
fax protocol pass-through g711ulaw
no vad
!
sip-ua
!
line con 0
exec-timeout 0 0
login
transport input none
stopbits 1
line aux 0
```



```
stopbits 1
line vty 0 4
exec-timeout 0 0
password tekV1z10n
login
transport input telnet
!
wsma agent exec
!
wsma agent config
!
wsma agent filesys
!
wsma agent notify
!
!
end

cube9#
```



Standby Cisco UBE

```
CUBE10#sh run
!
version 16.8
service timestamps debug datetime msec
service timestamps log datetime msec
platform qfp utilization monitor load 80
no platform punt-keepalive disable-kernel-core
!
hostname CUBE10
!
boot-start-marker
boot-end-marker
!
!
vrf definition Mgmt-intf
!
address-family ipv4
exit-address-family
!
address-family ipv6
exit-address-family
!
no aaa new-model
subscriber templating
multilink bundle-name authenticated
!
```



```
crypto pki trustpoint TP-self-signed-2930804041
  enrollment selfsigned
  subject-name cn=IOS-Self-Signed-Certificate-2930804041
  revocation-check none
  rsakeypair TP-self-signed-2930804041
!
crypto pki certificate chain TP-self-signed-2930804041
!
voice service voip
  ip address trusted list
    ipv4 0.0.0.0 0.0.0.0
  no ip address trusted authenticate
  address-hiding
  mode border-element license capacity 20
  allow-connections sip to sip
  redundancy-group 2
  supplementary-service media-renegotiate
  redirect ip2ip
  fax protocol pass-through g711ulaw
  sip
  session refresh
  asserted-id pai
  privacy pstn
  early-offer forced
  privacy-policy passthru
  g729 annexb-all
  pass-thru subscribe-notify-events all
!
```



```
!  
voice class uri trunk1 sip  
  host ipv4: 209.XXX.X.XXX  
  host ipv4: 209.XXX.XX.XX  
  host ipv4: 209.XXX.XX.XX  
  host ipv4: 209.XXX.XX.XX  
  host ipv4: 209.XXX.XX.XX  
  host ipv4: 209.XXX.XX.XX  
voice class codec 1  
  codec preference 1 g729r8  
  codec preference 2 g711alaw  
  codec preference 3 g711ulaw  
!  
voice class codec 2  
  codec preference 1 g711alaw  
  codec preference 2 g711ulaw  
!  
voice class server-group 1  
  ipv4 209.XXX.XX.XX preference 1  
  ipv4 209.XXX.X.XXX preference 1  
!  
voice translation-rule 1  
  rule 1 /(^.....$)/ /1\1/  
!
```



```
!  
voice translation-profile coverting_to_11Digits  
translate called 1  
!  
voice-card 0/4  
no watchdog  
!  
license udi pid ISR4331/K9 sn FDO41381F17  
no license smart enable  
diagnostic bootup level minimal  
!  
spanning-tree extend system-id  
!  
!  
!  
!  
redundancy  
mode none  
application redundancy  
group 2  
name Voice-b2bha_DIDforsale  
priority 100 failover threshold 75  
timers delay 30 reload 60  
control GigabitEthernet0/0/2 protocol 1  
data GigabitEthernet0/0/2  
track 1 shutdown  
track 2 shutdown  
!
```



!

!

track 1 interface GigabitEthernet0/0/0 line-protocol

!

track 2 interface GigabitEthernet0/0/1 line-protocol

!

!

!

!

!

!

interface GigabitEthernet0/0/0

description DIDforsale CUBE1 WAN

ip address 199.XXX.XXX.XX 255.255.255.224

negotiation auto

redundancy rii 4

redundancy group 2 ip 199.XXX.XXX.XX exclusive

!

interface GigabitEthernet0/0/1

description DIDforsale CUBE1 LAN

ip address 10.80.18.15 255.255.255.0

negotiation auto

redundancy rii 3

redundancy group 2 ip 10.80.18.16 exclusive

!

interface GigabitEthernet0/0/2

description CUBE HA

ip address 10.89.20.8 255.255.255.0



```
negotiation auto
!
interface Service-Engine0/4/0
!
interface GigabitEthernet0
vrf forwarding Mgmt-intf
no ip address
shutdown
negotiation auto
!
ip forward-protocol nd
ip http server
ip http authentication local
ip http secure-server
ip http client source-interface GigabitEthernet0/0/0
ip route 0.0.0.0 0.0.0.0 199.182.124.1
ip route 10.64.0.0 255.255.0.0 10.80.18.1
ip route 172.16.24.0 255.255.248.0 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 1010 voip
description Outgoing Call from PBX to PSTN-WAN facing
shutdown
destination-pattern .T
session protocol sipv2
session target ipv4:209.XXX.XX.XX:5060
voice-class codec 1
voice-class sip early-offer forced
voice-class sip options-keepalive
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 disable
fax protocol pass-through g711 ulaw
no vad
```

!

```
dial-peer voice 101 voip
description Outgoing Call from PBX to PSTN-WAN facing
destination-pattern .T
session protocol sipv2
session server-group 1
incoming called-number 0T
voice-class codec 1
voice-class sip asserted-id pai
```



```
voice-class sip privacy-policy passthru
voice-class sip early-offer forced
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
voice-class sip midcall-signaling passthru media-change
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate disable
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 222 voip
description outbound-from PBX-PSTN - LAN facing
destination-pattern 1626313....
session protocol sipv2
session target ipv4:10.80.20.2:5060
voice-class codec 1
voice-class sip early-offer forced
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
voice-class sip midcall-signaling passthru media-change
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate disable
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 111 voip
```



```
description Inbound peer match FROM DIDFORSALE
session protocol sipv2
incoming uri via trunk1
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
voice-class sip midcall-signaling passthru media-change
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate disable
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 1000 voip
description PBX to PBX call via DIDforsale - WAN facing
translation-profile outgoing coverting_to_11Digits
destination-pattern 626313....
session protocol sipv2
session server-group 1
voice-class codec 1
voice-class sip early-offer forced
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 disable
fax protocol pass-through g711ulaw
!
dial-peer voice 201 voip
```



```
description Inbound peer match FROM CUCM
session protocol sipv2
incoming called-number .T
voice-class codec 1
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
voice-class sip midcall-signaling passthru media-change
dtmf-relay rtp-nte
fax-relay ecm disable
fax rate disable
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 203 voip
description Inbound peer match FROM CUCM
session protocol sipv2
incoming called-number .T
voice-class codec 1
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 disable
fax protocol pass-through g711ulaw
no vad
!
!
sip-ua
!
```



```
!  
line con 0  
  exec-timeout 0 0  
  password tekV1z10n  
  login  
  transport input none  
  stopbits 1  
line aux 0  
  stopbits 1  
line vty 0 4  
  exec-timeout 0 0  
  login  
  transport input telnet  
!  
ntp server 34.208.249.133  
wsma agent exec  
!  
wsma agent config  
!  
wsma agent filesys  
wsma agent notify  
!  
pnp profile pnp_redirection_profile  
  transport http ipv4 127.0.0.1 port 80  
end
```



Configuring Cisco Unified Communications Manager Cisco UCM Version

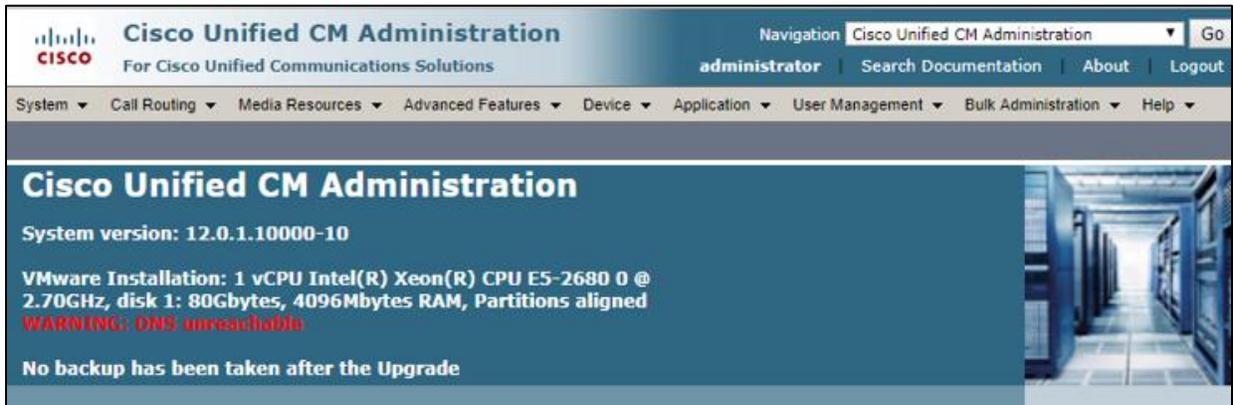


Figure 7: Cisco UCM Version

Cisco Call Manager Service Parameters

Navigation path: System > Service Parameters

Select Server* = Clus20pub1--CUCM Voice/Video (Active)

Select Service* = Cisco CallManager (Active)

All other fields are set to default values

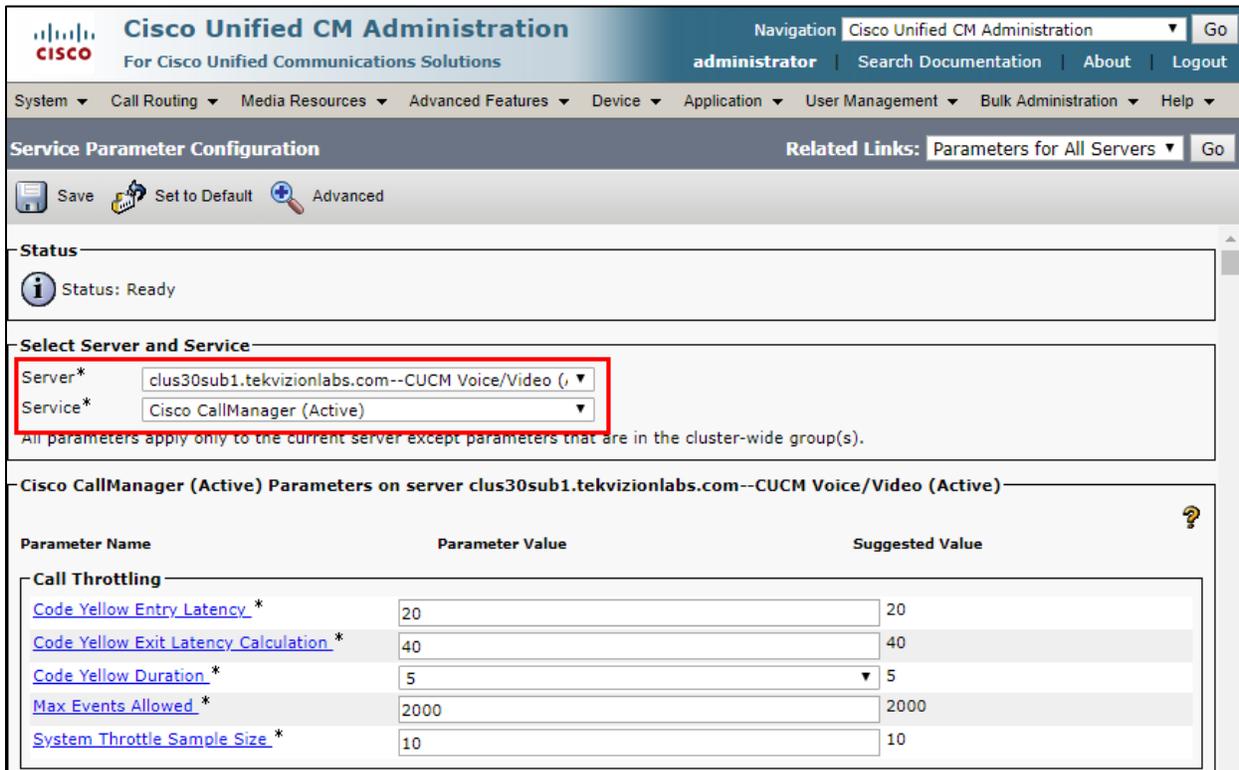


Figure 8: Service Parameters

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Offnet Calls via DIDforSale SIP Trunk

Off-net calls are served by SIP trunks configured between Cisco UCM and DIDforSale Network and calls are routed via Cisco UBE

SIP Trunk Security Profile

Navigation Path: System > Security > SIP Trunk Security Profile

Name*= non secure UDP sip trunk profile for DIDforSale

Description = Non Secure SIP Trunk Profile

(Note:- For this test the Outgoing Transport type was provisioned as UDP. DIDforsale supports TCP as well.)

The screenshot shows the Cisco Unified CM Administration interface. The main heading is "SIP Trunk Security Profile Configuration". Below the heading, there are several icons for actions: Save, Delete, Copy, Reset, Apply Config, and Add New. The status is "Ready".

The "SIP Trunk Security Profile Information" section is highlighted with a red box and contains the following fields:

- Name*: non secure UDP sip trunk profile for DIDforSALE
- Description: non secure UDP sip trunk profile for DIDforSALE
- Device Security Mode: Non Secure
- Incoming Transport Type*: TCP+UDP
- Outgoing Transport Type: UDP

Below this section, there are additional fields:

- Enable Digest Authentication
- Nonce Validity Time (mins)*: 600
- X.509 Subject Name: (empty text area)
- Incoming Port*: 5060

Figure 9: SIP Trunk Security Profile

Explanation

Parameter	Value	Description
Incoming Transport Type	TCP + UDP	
Outgoing Transport Type	UDP	SIP trunks to DIDforSale, 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

SIP Profile will be later associated with the SIP trunk

Navigation Path: Device > Device Settings > SIP Profile

Name*= DIDFORSALE_standard_SIP_profile

Description = DIDFORSALE_standard_SIP_profile

The screenshot displays the Cisco Unified CM Administration interface for SIP Profile Configuration. The page title is "SIP Profile Configuration" and it includes a navigation menu at the top. The main content area is divided into sections: "Status" and "SIP Profile Information".

Status:

- Status: Ready
- All SIP devices using this profile must be restarted before any changes will take affect.

SIP Profile Information:

Name*	DIDFORSALE_standard_SIP_Profile
Description	DIDFORSALE_standard_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

Additional options (checkboxes):

- Redirect by Application
- Disable Early Media on 180
- Outgoing T.38 INVITE include audio mline
- Offer valid IP and Send/Receive mode only for T.38 Fax Relay
- Use Fully Qualified Domain Name in SIP Requests
- Assured Services SIP conformance



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 checked="" 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
Start Media Port*	16384
Stop Media Port*	32766

Figure 10: SIP Profile



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	

Normalization Script											
Normalization Script	< None >										
<input type="checkbox"/> Enable Trace											
	<table border="1"> <thead> <tr> <th></th> <th>Parameter Name</th> <th>Parameter Value</th> <th></th> <th></th> </tr> </thead> <tbody> <tr> <td>1</td> <td></td> <td></td> <td>+</td> <td>-</td> </tr> </tbody> </table>		Parameter Name	Parameter Value			1			+	-
	Parameter Name	Parameter Value									
1			+	-							

Incoming Requests FROM URI Settings	
Caller ID DN	
Caller Name	

Figure 11: SIP Profile (Cont.)



Trunk Specific Configuration

Reroute Incoming Request to new Trunk based on*

Resource Priority Namespace List

SIP Rel1XX Options*

Video Call Traffic Class*

Calling Line Identification Presentation*

Session Refresh Method*

Early Offer support for voice and video calls*

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

SIP OPTIONS Ping

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)*

Ping Interval for Out-of-service Trunks (seconds)*

Ping Retry Timer (milliseconds)*

Ping Retry Count*

SDP Information

Send send-receive SDP in mid-call INVITE

Allow Presentation Sharing using BFCP

Allow iX Application Media

Allow multiple codecs in answer SDP

Figure 12: 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 Messages	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



Device Pool Configuration

Device Pool will be later associated with the SIP trunk

Navigation Path: System > Device Pool

Name*= DIDforsale_Codec G711_devicepool

The screenshot shows the Cisco Unified CM Administration interface for Device Pool Configuration. The page title is "Device Pool Configuration" and it includes a navigation menu at the top. The main content area is divided into three sections: "Device Pool Information", "Device Pool Settings", and "Roaming Sensitive Settings".

Device Pool Information: Device Pool: DIDforsale_Codec G711_devicepool (5 members**)

Device Pool Settings:

Device Pool Name*	DIDforsale_Codec G711_devicepool
Cisco Unified Communications Manager Group*	Default
Calling Search Space for Auto-registration	< None >
Adjunct CSS	< None >
Reverted Call Focus Priority	Default
Intercompany Media Services Enrolled Group	< None >

Roaming Sensitive Settings:

Date/Time Group*	CMLocal
Region*	DIDforsale_Codec G711_Region
Media Resource Group List	< None >
Location	< None >
Network Locale	< None >
SRST Reference*	Disable
Connection Monitor Duration***	
Single Button Barge*	Default
Join Across Lines*	Default

Figure 13: Device Pool



Physical Location

Device Mobility Group

Wireless LAN Profile Group [View Details](#)

Local Route Group Settings

Standard Local Route Group

Device Mobility Related Information****

Device Mobility Calling Search Space

AAR Calling Search Space

AAR Group

Calling Party Transformation CSS

Called Party Transformation CSS

Geolocation Configuration

Geolocation

Geolocation Filter

Figure 13: Device Pool (cont.)

Call Routing Information

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.

Number Type	Prefix	Strip Digits	Calling Search Space
National Number	<input type="text" value="Default"/>	<input type="text"/>	<input type="text" value="< None >"/>
International Number	<input type="text" value="Default"/>	<input type="text"/>	<input type="text" value="< None >"/>
Unknown Number	<input type="text" value="Default"/>	<input type="text"/>	<input type="text" value="< None >"/>
Subscriber Number	<input type="text" value="Default"/>	<input type="text"/>	<input type="text" value="< None >"/>

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.



Number Type	Prefix	Strip Digits	Calling Search Space
National Number	<input type="text" value="Default"/>	<input type="text" value="0"/>	<input type="text" value=" < None >"/>
International Number	<input type="text" value="Default"/>	<input type="text" value="0"/>	<input type="text" value=" < None >"/>
Unknown Number	<input type="text" value="Default"/>	<input type="text" value="0"/>	<input type="text" value=" < None >"/>
Subscriber Number	<input type="text" value="Default"/>	<input type="text" value="0"/>	<input type="text" value=" < None >"/>

Phone Settings

Caller ID For Calls From This Phone

Calling Party Transformation CSS

Connected Party Settings

Connected Party Transformation CSS

Redirecting Party Settings

Redirecting Party Transformation CSS



SIP Trunk Configuration

Create SIP trunks to Cisco UBE

Navigation Path: Device > Trunk

The screenshot shows the Cisco Unified CM Administration interface. The top navigation bar includes "System", "Call Routing", "Media Resources", "Advanced Features", "Device", "Application", "User Management", "Bulk Administration", and "Help". The main content area is titled "Find and List Trunks" and shows a search filter for "didforsale" applied to the "Device Name" field. The table below lists three SIP trunks:

Name	Description	Calling Search Space	Device Pool	Route Pattern	Partition	Route Group	Priority	Trunk Type	SIP Trunk Status	SIP Trunk Duration
SIP_TRUNK_DIDforsale	SIP_TRUNK_DIDforsale		DIDforsale_Codec G711_devicepool	2_@				SIP Trunk	Full Service	Time In Full Service: 0 day 21 hours 33 minutes
Unity_Connection_DIDforsale	Unity_Connection_DIDforsale		DIDforsale_Codec G711_devicepool	4000				SIP Trunk	Full Service	Time In Full Service: 0 day 0 hour 0 minute
fax_DIDforsale	fax_didforsale		DIDforsale_Codec G711_devicepool	2028				SIP Trunk	Full Service	Time In Full Service:

Figure 14: SIP Trunks List

Cisco Unified CM Administration
For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration Go
administrator | Search Documentation | About | Logout

System | Call Routing | Media Resources | Advanced Features | Device | Application | User Management | Bulk Administration | Help

Trunk Configuration Related Links: Back To Find/List Go

Save ✖ Delete ↺ Reset + Add New

Device Information

Product:	SIP Trunk
Device Protocol:	SIP
Trunk Service Type	None(Default)
Device Name*	SIP_TRUNK_DIDforsale
Description	SIP_TRUNK_DIDforsale
Device Pool*	DIDforsale_Codec G711_devicepool
Common Device Configuration	< None >
Call Classification*	Use System Default
Media Resource Group List	MRGL_DIDFORSALE
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
 Retry Video Call as Audio

Figure 13: SIP Trunk to Cisco UBE



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 checked="" type="checkbox"/> Run On All Active Unified CM Nodes	

MLPP and Confidential Access Level Information

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

Call Routing Information

<input checked="" type="checkbox"/> Remote-Party-Id	
<input checked="" type="checkbox"/> Asserted-Identity	
Asserted-Type*	Default
SIP Privacy*	Default
Trust Received Identity*	Trust All (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	
<input checked="" type="checkbox"/> Redirecting Diversion Header Delivery - Inbound	

Figure 14: SIP Trunk to Cisco UBE (cont.)



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.

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS
Incoming Number	<input type="text" value="Default"/>	<input type="text" value="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.

Number Type	Prefix	Strip Digits	Calling Search Space	Use Device Pool CSS

Connected Party Settings

Connected Party Transformation CSS

Use Device Pool Connected Party Transformation CSS

Outbound Calls

Called Party Transformation CSS

Use Device Pool Called Party Transformation CSS

Calling Party Transformation CSS

Use Device Pool Calling Party Transformation CSS

Calling Party Selection*

Calling Line ID Presentation*

Calling Name Presentation*

Calling and Connected Party Info Format*

Redirecting Diversion Header Delivery - Outbound

Redirecting Party Transformation CSS

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

SIP Information

Destination

Destination Address is an SRV

Destination Address	Destination Address IPv6	D
1* 10.80.18.16		5060

MTP Preferred Originating Codec*

BLF Presence Group*

SIP Trunk Security Profile*

Rerouting Calling Search Space

Out-Of-Dialog Refer Calling Search Space

SUBSCRIBE Calling Search Space

SIP Profile* [View Details](#)

DTMF Signaling Method*

Normalization Script

Normalization Script

Enable Trace

	Parameter Name	Parameter Value
1	<input type="text"/>	<input type="text"/>

Recording Information

None

This trunk connects to a recording-enabled gateway

This trunk connects to other clusters with recording-enabled gateways

Geolocation Configuration

Geolocation

Geolocation Filter

Send Geolocation Information

Figure 15: SIP Trunk to Cisco UBE (Cont.)



Explanation

Parameter	Value	Description
Device Name	SIP_TRUNK_DIDforsale	Name for the trunk
Device Pool	DIDforsale_codec711_devicepool	Default Device Pool is used for this trunk
Media Resource Group List	MRGL_DIDFORSALE	MRG with resources: ANN, CFB, MOH and MTP
Significant Digits	4	4 digits Extension for all CPE phones
Destination Address	10.80.18.16	IP address of the Cisco UBE Virtual LAN
SIP Trunk Security Profile	non Secure UDP sip trunk profile for DIDforSale	SIP Trunk Security Profile configured earlier
SIP Profile	DIDforSale_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:

- Cisco IP phone dial “2”+“1”+10 digits number to access PSTN via Cisco UBE
 - “2” is removed before sending to Cisco UBE
- For FAX call, Access Code “2”+10 digits number is used at Cisco Fax gateway
 - “2” is removed at Cisco UCM
 - The rest of the number is sent to Cisco UBE to DIDforSale network
- Incoming fax call to 2028 will be sent to Cisco ATA
- Cisco IP phones dial X11 for emergency call and will send all digits to Cisco UBE to DIDforSale network

Route Pattern	Description	Trunk
2.626313XXXX		SIP_TRUNK_DIDforsale
2.@	DID for Sale-PSTN Calls	SIP_TRUNK_DIDforsale
2028		fax_DIDforsale
4000	Pilot Number to Unity Connection	Unity_Connection_DIDforsale

Figure 16: Route Patterns List



Cisco Unified CM Administration For Cisco Unified Communications Solutions

Navigation: Cisco Unified CM Administration administrator | Search Documentation | About | Logout

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾

Route Pattern Configuration Related Links: [Back To Find/List](#)

Pattern Definition

Route Pattern*	2.@
Route Partition	< None >
Description	DID for Sale-PSTN 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*	SIP_TRUNK_DIDforsale Edit
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern <input type="text" value="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



Require Client Matter Code

Calling Party Transformations

Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation* Allowed ▼

Calling Name Presentation* Allowed ▼

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 PreDot ▼

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

Called Party Transformations

Discard Digits PreDot ▼

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 >	<input type="text"/>

Save Delete Copy Add New

Figure 17: Route Pattern for Voice



Cisco Unified CM Administration Navigation Cisco Unified CM Administration Go
For Cisco Unified Communications Solutions administrator | Search Documentation | About | Logout

System ▾ Call Routing ▾ Media Resources ▾ Advanced Features ▾ Device ▾ Application ▾ User Management ▾ Bulk Administration ▾

Route Pattern Configuration Related Links: Back To Find/List ▾ Go

Save ~~X~~ Delete Copy + Add New

Pattern Definition

Route Pattern*	4000		
Route Partition	< None >		
Description	Pilot Number to Unity Connection		
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*	Unity_Connection_Trunk (Edit)		
Route Option	<input checked="" type="radio"/> Route this pattern <input type="radio"/> Block this pattern No Error		
Call Classification*	OpNet		
External Call Control Profile	< None >		
<input type="checkbox"/> Allow Device Override	<input 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		

Calling Party Transformations

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: -- Not Selected --

Service Parameter Name: < Not Exist >

Service Parameter:

Save Delete Copy Add New

Figure 18: Route Pattern for Voice (Cont.)

Cisco Unified CM Administration Navigation: Cisco Unified CM Administration Go

administrator | Search Documentation | About | Logout

System Call Routing Media Resources Advanced Features Device Application User Management Bulk Administration Help

Route Pattern Configuration Related Links: Back To Find/List Go

Save Delete Copy Add New

Pattern Definition

Route Pattern*: 2028

Route Partition: < None >

Description:

Numbering Plan: -- Not Selected --

Route Filter: < None >

MLPP Precedence*: Default

Apply Call Blocking Percentage

Resource Priority Namespace Network Domain: < None >

Route Class*: Default

Gateway/Route List*: fax_DIDforsale (Edit)

Route Option:
 Route this pattern
 Block this pattern No Error

Call Classification*: OffNet

External Call Control Profile: < None >

Allow Device Override Provide Outside Dial Tone Allow Overlap Sending Urgent Priority

Require Forced Authorization Code

Authorization Level*: 0



External Call Control Profile

Allow Device Override Provide Outside Dial Tone Allow Overlap Sending Urgent Priority

Require Forced Authorization Code

Authorization Level*

Require Client Matter Code

Calling Party Transformations

Use Calling Party's External Phone Number Mask

Calling Party Transform Mask

Prefix Digits (Outgoing Calls)

Calling Line ID Presentation*

Calling Name Presentation*

Calling Party Number Type*

Calling Party Numbering Plan*

Connected Party Transformations

Connected Line ID Presentation*

Connected Name Presentation*

Connected Party Transformations

Connected Line ID Presentation*

Connected Name Presentation*

Called Party Transformations

Discard Digits

Called Party Transform Mask

Prefix Digits (Outgoing Calls)

Called Party Number Type*

Called Party Numbering Plan*

ISDN Network-Specific Facilities Information Element

Network Service Protocol

Carrier Identification Code

Network Service	Service Parameter Name	Service Parameter Value
<input style="width: 100%;" type="text" value=" -- Not Selected -- "/>	<input style="width: 100%;" type="text" value=" < Not Exist > "/>	<input style="width: 100%;" type="text"/>

Figure 19: Route Pattern for Fax



Explanation

Setting	Value	Description
Route Pattern	2.@ for Voice & International Calls, and 2028 for fax call and 4000 for Unity connection	Specify appropriate Route Pattern
Gateway/Route List	To_DIDforSale route Pattern 9.@, 2028 for SIP Trunk to FAX Gateway and 4000 for Unity Connection	SIP Trunk name configured earlier
Numbering Plan	NANP for Route Pattern 2.@	North American Numbering Plan
Call Classification	Offnet for Route Pattern 2.@, 2028 and 4000	Restrict the transferring of an external call to an external device
Discard Digits	PreDot for Route Pattern 2.@	Specifies how to modify digit before they are sending to DIDforSale network

FAX ATA
FAX Configuration

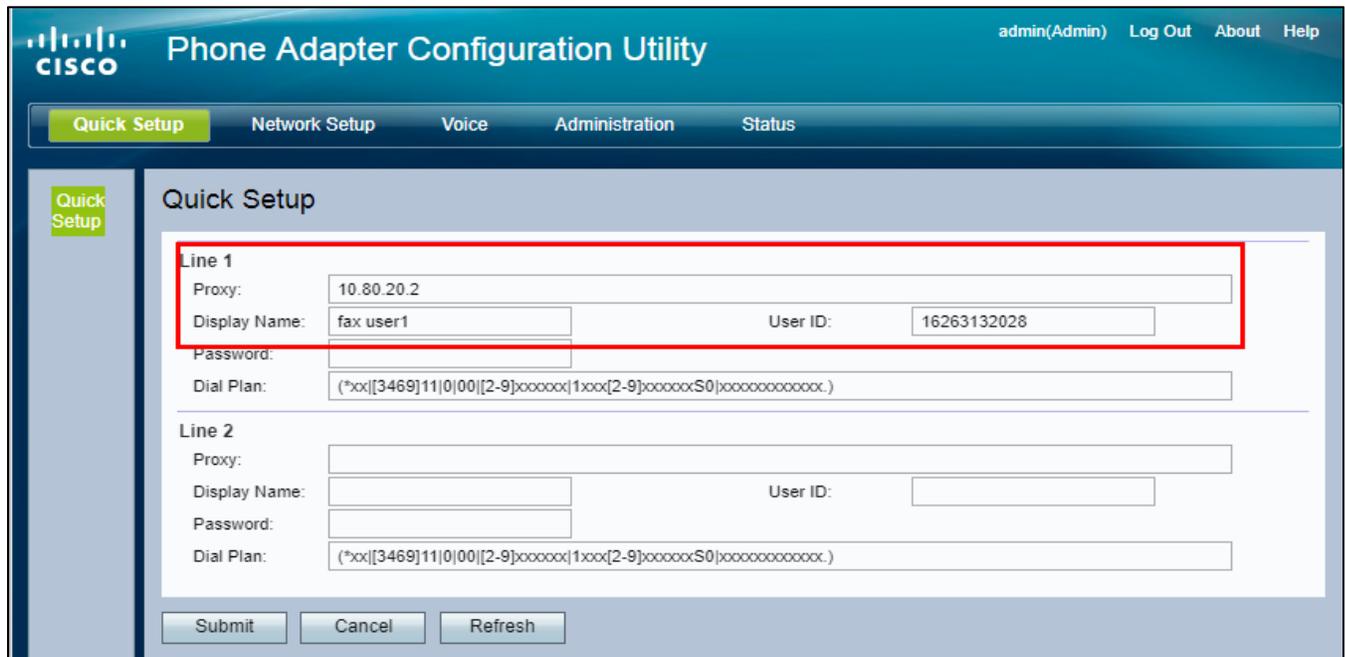


Figure 21: FAX ATA Configuration



admin(Admin) Log Out About Help

Quick Setup Network Setup **Voice** Administration Status

Information System SIP Provisioning Regional **Line 1** User 1 Line 2 User 2

Line 1

General
Line Enable: yes

Streaming Audio Server (SAS)
SAS Enable: no SAS DLG Refresh Intvl:
SAS Inbound RTP Sink:

NAT Settings
NAT Mapping Enable: no NAT Keep Alive Enable: no
NAT Keep Alive Msg: NAT Keep Alive Dest:

Network Settings
SIP ToS/DiffServ Value: SIP CoS Value: [0-7]
RTP ToS/DiffServ Value: RTP CoS Value: [0-7]
Network Jitter Level: Jitter Buffer Adjustment: yes

Figure 22: FAX ATA Configuration (cont.)

SIP Provisioning Regional **Line 1** User 1 Line 2 User 2

SIP Settings
SIP Transport: SIP Port:
SIP 100REL Enable: no EXT SIP Port:
Auth Resync-Reboot: yes SIP Proxy-Require:
SIP Remote-Party-ID: yes SIP GUID:
SIP Debug Option: RTP Log Intvl:
Restrict Source IP: no Referor Bye Delay:
Refer Target Bye Delay: Referee Bye Delay:
Refer-To Target Contact: no Sticky 183: no
Auth INVITE: no Reply 182 On Call Waiting: no
Use Anonymous With RPID: yes Use Local Addr In FROM: no

Call Feature Settings
Blind Attn-Xfer Enable: no MOH Server:
Xfer When Hangup Conf: yes Conference Bridge URL:
Conference Bridge Ports: Enable IP Dialing: no
Emergency Number: Mailbox ID:



SIP Provisioning Regional
Line 1
User 1
Line 2
User 2

Proxy and Registration

Proxy: 10.80.20.2

Outbound Proxy:

Use Outbound Proxy: no

Register: no

Register Expires: 60

Use DNS SRV: no

Proxy Fallback Intvl: 60

Mailbox Subscribe URL:

Use OB Proxy In Dialog: yes

Make Call Without Reg: yes

Ans Call Without Reg: yes

DNS SRV Auto Prefix: no

Proxy Redundancy Method: Normal

Mailbox Subscribe Expires: 2147483647

Subscriber Information

Display Name: fax user1

User ID: 16263132028

Password:

Auth ID:

SIP URI:

Use Auth ID: no

Resident Online Number:

SIP Provisioning Regional
Line 1
User 1
Line 2
User 2

Audio Configuration

Preferred Codec: G729a

Third Preferred Codec: Unspecified

Use Remote Pref Codec: no

G729a Enable: yes

G726-32 Enable: no

FAX V21 Detect Enable: yes

FAX CNG Detect Enable: yes

FAX Codec Symmetric: yes

FAX Passthru Method: ReINVITE

FAX Process NSE: yes

FAX Disable ECAN: no

DTMF Tx Strict Hold Off Time: 70

Hook Flash Tx Method: None

FAX T38 ECM Enable: yes

Symmetric RTP: no

Modem Line: no

Second Preferred Codec: G711u

Use Pref Codec Only: yes

Codec Negotiation: Default

Silence Supp Enable: yes

Silence Threshold: medium

Echo Canc Enable: yes

FAX Passthru Codec: G711u

DTMF Process INFO: yes

DTMF Process AVT: yes

DTMF Tx Method: Auto

DTMF Tx Mode: Strict

FAX Enable T38: yes

FAX T38 Redundancy: 0

FAX Tone Detect Mode: caller or callee

FAX T38 Return to Voice: no

DTX to Proxy in Remote Hold: no

Hook Flash Tx Method: None

FAX T38 ECM Enable: yes

Symmetric RTP: no

Modem Line: no

FAX T38 Redundancy: 0

FAX Tone Detect Mode: caller or callee

FAX T38 Return to Voice: no

RTP to Proxy in Remote Hold: no

Dial Plan

Dial Plan: (*xx|[3469]11|0|00|[2-9]xxxxx|1xxx[2-9]xxxxxS0|xxxxxxxxxx.)

FXS Port Polarity Configuration

Idle Polarity: Forward

Caller Conn Polarity: Forward

Callee Conn Polarity: Forward

Submit Cancel Refresh



Acronyms

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



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**Corporate Headquarters**

Cisco Systems, Inc.
170 West Tasman Drive
San Jose, CA 95134-1706
USA
www.cisco.com
Tel: 408 526-4000
800 553-NETS (6387)
Fax: 408 526-4100

European Headquarters

CiscoSystems
International BV
Haarlerbergpark
Haarlerbergweg 13-19
1101 CH Amsterdam
The Netherlands
www-europe.cisco.com
Tel: 31 0 20 357 1000
Fax: 31 0 20 357 1100

Americas Headquarters

Cisco Systems, Inc.
170 West Tasman Drive
San Jose, CA 95134-1706
USA
www.cisco.com
Tel: 408 526-7660
Fax: 408 527-0883

AsiaPacific Headquarters

Cisco Systems, Inc.
Capital Tower
168 Robinson Road
#22-01 to #29-01
Singapore 068912
www.cisco.com
Tel: +65 317 7777
Fax: +65 317 7799

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