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:

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.

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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 g711ulaw
sip
session refresh
asserted-id pai
privacy pstn
early-offer forced
privacy-policy passthru
g729 annexb-all
pass-thru subscribe-notify-events all
!
!
### Explanation

<table>
<thead>
<tr>
<th>Command</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>allow-connections sip to sip</td>
<td>Allow IP2IP connections between two SIP call legs</td>
</tr>
<tr>
<td>fax protocol</td>
<td>Specifies the fax protocol</td>
</tr>
<tr>
<td>asserted-id</td>
<td>Specifies the type of privacy header in the outgoing SIP requests and response messages</td>
</tr>
<tr>
<td>pass-thru subscribe-notify-events all</td>
<td>This command is to configure pass-through for all SUBSCRIBE-NOTIFY events</td>
</tr>
</tbody>
</table>

### Codecs

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

```plaintext
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.

```plaintext
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](image)

![Figure 3: Inbound Voice Call](image)
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.X.XXX
host ipv4: 209.XXX.X.XX
host ipv4: 209.XXX.X.XX
host ipv4: 209.XXX.X.XX
host ipv4: 209.XXX.X.XX
host ipv4: 209.XXX.X.XX
host ipv4: 209.XXX.X.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 covert_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.XXX.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
dtmf-relay rtp-nte
description Inbound peer match FROM DIDFORSALE
shutdow
dtmf-relay rtp-nne
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-nne
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 DIForsale-SecondTrunk - WAN facing
translation-profile outgoing converting_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.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.XX
host ipv4: 209.XXX.X.XX
host ipv4: 209.XXX.X.XX
host ipv4: 209.XXX.X.XX
host ipv4: 209.XXX.X.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 /\(^\d\d\d\d\d\d\)...$/ /1/1/
voice translation-profile converting_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 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
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-npe
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 converting_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-npe
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

![Cisco UCM Version](image)

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

![Service Parameters](image)

Figure 8: Service Parameters
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.)

![Figure 9: SIP Trunk Security Profile](image)

Explanation

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Incoming Transport Type</td>
<td>TCP + UDP</td>
<td>SIP trunks to DIDforSale, SBC should use UDP as a transport protocol for SIP. This is configured</td>
</tr>
<tr>
<td>Outgoing Transport Type</td>
<td>UDP</td>
<td></td>
</tr>
</tbody>
</table>
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
Figure 10: SIP Profile
<table>
<thead>
<tr>
<th>Configuration Item</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>DSCP for Audio Calls</td>
<td>Use System Default</td>
</tr>
<tr>
<td>DSCP for Video Calls</td>
<td>Use System Default</td>
</tr>
<tr>
<td>DSCP for Audio Portion of Video Calls</td>
<td>Use System Default</td>
</tr>
<tr>
<td>DSCP for TelePresence Calls</td>
<td>Use System Default</td>
</tr>
<tr>
<td>DSCP for Audio Portion of TelePresence Calls</td>
<td>Use System Default</td>
</tr>
<tr>
<td>Call Pickup URI</td>
<td>x-cisco-serviceuri-pickup</td>
</tr>
<tr>
<td>Call Pickup Group Other URI</td>
<td>x-cisco-serviceuri-pickup</td>
</tr>
<tr>
<td>Call Pickup Group URI</td>
<td>x-cisco-serviceuri-pickup</td>
</tr>
<tr>
<td>Meet Me Service URI</td>
<td>x-cisco-serviceuri-meetme</td>
</tr>
<tr>
<td>User Info</td>
<td>None</td>
</tr>
<tr>
<td>DTMF DB Level</td>
<td>Nominal</td>
</tr>
<tr>
<td>Call Hold Ring Back</td>
<td>Off</td>
</tr>
<tr>
<td>Anonymous Call Block</td>
<td>Off</td>
</tr>
<tr>
<td>Caller ID Blocking</td>
<td>Off</td>
</tr>
<tr>
<td>Do Not Disturb Control</td>
<td>User</td>
</tr>
<tr>
<td>Telnet Level for 7940 and 7960</td>
<td>Disabled</td>
</tr>
<tr>
<td>Resource Priority Namespace</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Timer Keep Alive Expires (seconds)</td>
<td>120</td>
</tr>
<tr>
<td>Timer Subscribe Expires (seconds)</td>
<td>120</td>
</tr>
<tr>
<td>Timer Subscribe Delta (seconds)</td>
<td>5</td>
</tr>
<tr>
<td>Maximum Redirects</td>
<td>70</td>
</tr>
<tr>
<td>Off Hook To First Digit Timer (milliseconds)</td>
<td>15000</td>
</tr>
<tr>
<td>Call Forward URI</td>
<td>x-cisco-serviceuri-chwdial</td>
</tr>
<tr>
<td>Speed Dial (Abbreviated Dial) URI</td>
<td>x-cisco-serviceuri-obbdial</td>
</tr>
<tr>
<td>Conference Join Enabled</td>
<td></td>
</tr>
<tr>
<td>RFC 2543 hold</td>
<td></td>
</tr>
<tr>
<td>Semi Attended Transfer</td>
<td></td>
</tr>
<tr>
<td>Enable VAD</td>
<td></td>
</tr>
<tr>
<td>Stutter Message Waiting</td>
<td></td>
</tr>
<tr>
<td>MLPP User Authorization</td>
<td></td>
</tr>
</tbody>
</table>

**Normalization Script**:<br>

<table>
<thead>
<tr>
<th>Parameter Name</th>
<th>Parameter Value</th>
</tr>
</thead>
<tbody>
<tr>
<td></td>
<td></td>
</tr>
</tbody>
</table>

**Incoming Requests FROM URI Settings**:<br>

<table>
<thead>
<tr>
<th>Caller ID DN</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Caller Name</td>
<td></td>
</tr>
</tbody>
</table>

Figure 11: SIP Profile (Cont.)
### Trunk Specific Configuration

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

### SIP OPTIONS Ping

- Enable OPTIONS Ping to monitor destination status for Trunks with Service Type "None (Default)"

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Ping Interval for In-service and Partially In-service Trunks (seconds)</td>
<td>60</td>
<td>OPTIONS message parameters- interval time</td>
</tr>
<tr>
<td>Ping Retry Timer (milliseconds)</td>
<td>120</td>
<td>OPTIONS message parameters- interval time</td>
</tr>
<tr>
<td>Ping Retry Count</td>
<td>500</td>
<td>OPTIONS message parameters- interval time</td>
</tr>
</tbody>
</table>

### SDP Information

- Send send-receive SDP in mid-call INVITE

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Allow Presentation Sharing using BFCP</td>
<td></td>
<td>OPTIONS message parameters- interval time</td>
</tr>
<tr>
<td>Allow iX Application Media</td>
<td></td>
<td>OPTIONS message parameters- interval time</td>
</tr>
<tr>
<td>Allow multiple codecs in answer SDP</td>
<td></td>
<td>OPTIONS message parameters- interval time</td>
</tr>
</tbody>
</table>

### Explanation

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

Device Pool will be later associated with the SIP trunk

**Navigation Path:** System > Device Pool

Name* = DIDforsale_Codec G711_devicepool

---

**Figure 13: Device Pool**

---

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Figure 13: Device Pool (cont.)
<table>
<thead>
<tr>
<th>Number Type</th>
<th>Prefix</th>
<th>Strip Digits</th>
<th>Calling Search Space</th>
</tr>
</thead>
<tbody>
<tr>
<td>National Number</td>
<td>Default</td>
<td>0</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>International Number</td>
<td>Default</td>
<td>0</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Unknown Number</td>
<td>Default</td>
<td>0</td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Subscriber Number</td>
<td>Default</td>
<td>0</td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

**Phone Settings**

**Caller ID For Calls From This Phone**

**Connected Party Settings**

**Redirecting Party Settings**

Save  Delete  Copy  Reset  Apply Config  Add New
SIP Trunk Configuration

Create SIP trunks to Cisco UBE

**Navigation Path:** Device > Trunk

![SIP Trunks List](image)

*Figure 14: SIP Trunks List*
Figure 13: SIP Trunk to Cisco UBE
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.

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

### Incoming Called Party Settings

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

### 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**: Originator
- **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**
Figure 15: SIP Trunk to Cisco UBE (Cont.)
**Explanation**

<table>
<thead>
<tr>
<th>Parameter</th>
<th>Value</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Device Name</td>
<td>SIP_TRUNK_DIDforsale</td>
<td>Name for the trunk</td>
</tr>
<tr>
<td>Device Pool</td>
<td>DIDforsale_codecg711_devicepool</td>
<td>Default Device Pool is used for this trunk</td>
</tr>
<tr>
<td>Media Resource Group List</td>
<td>MRG_DIDFORSALE</td>
<td>MRG with resources: ANN, CFB, MOH and MTP</td>
</tr>
<tr>
<td>Significant Digits</td>
<td>4</td>
<td>4 digits Extension for all CPE phones</td>
</tr>
<tr>
<td>Destination Address</td>
<td>10.80.18.16</td>
<td>IP address of the Cisco UBE Virtual LAN</td>
</tr>
<tr>
<td>SIP Trunk Security Profile</td>
<td>non Secure UDP sip trunk profile for DIDforSale</td>
<td>SIP Trunk Security Profile configured earlier</td>
</tr>
<tr>
<td>SIP Profile</td>
<td>DIDforSale_standard_SIP_Profile</td>
<td>SIP Profile configured earlier</td>
</tr>
</tbody>
</table>

**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

![Figure 16: Route Patterns List](image-url)
### Figure 17: Route Pattern for Voice

<table>
<thead>
<tr>
<th>Calling Party Transformations</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Require Client Matter Code</td>
<td></td>
</tr>
<tr>
<td>Use Calling Party’s External Phone Number Mask</td>
<td></td>
</tr>
<tr>
<td>Calling Party Transform Mask</td>
<td></td>
</tr>
<tr>
<td>Prefix Digits (Outgoing Calls)</td>
<td></td>
</tr>
<tr>
<td>Calling Line ID Presentation</td>
<td>Allowed</td>
</tr>
<tr>
<td>Calling Name Presentation</td>
<td>Allowed</td>
</tr>
<tr>
<td>Calling Party Number Type</td>
<td>Cisco CallManager</td>
</tr>
<tr>
<td>Calling Party Numbering Plan</td>
<td>Cisco CallManager</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Connected Party Transformations</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Connected Line ID Presentation</td>
<td>Default</td>
</tr>
<tr>
<td>Connected Name Presentation</td>
<td>Default</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Called Party Transformations</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Discard Digits</td>
<td>PreDot</td>
</tr>
<tr>
<td>Called Party Transform Mask</td>
<td></td>
</tr>
<tr>
<td>Prefix Digits (Outgoing Calls)</td>
<td></td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>Called Party Transformations</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Discard Digits</td>
<td>PreDot</td>
</tr>
<tr>
<td>Called Party Transform Mask</td>
<td></td>
</tr>
<tr>
<td>Prefix Digits (Outgoing Calls)</td>
<td></td>
</tr>
<tr>
<td>Called Party Number Type</td>
<td>Cisco CallManager</td>
</tr>
<tr>
<td>Called Party Numbering Plan</td>
<td>Cisco CallManager</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>ISDN Network-Specific Facilities Information Element</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Network Service Protocol</td>
<td>-- Not Selected --</td>
</tr>
<tr>
<td>Carrier Identification Code</td>
<td></td>
</tr>
<tr>
<td>Network Service</td>
<td>Service Parameter Name</td>
</tr>
<tr>
<td>-- Not Selected --</td>
<td>-- Not Exist --</td>
</tr>
</tbody>
</table>

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**Figure 18: Route Pattern for Voice (Cont.)**

<table>
<thead>
<tr>
<th>ISDN Network-Specific Facilities Information Element</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Network Service Protocol</strong></td>
</tr>
<tr>
<td><strong>Carrier Identification Code</strong></td>
</tr>
<tr>
<td><strong>Network Service</strong></td>
</tr>
<tr>
<td>--- Not Selected ---</td>
</tr>
</tbody>
</table>

![Route Pattern Configuration Diagram]

<table>
<thead>
<tr>
<th>Pattern Definition</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Route Pattern</strong></td>
</tr>
<tr>
<td><strong>Route Partition</strong></td>
</tr>
<tr>
<td><strong>Description</strong></td>
</tr>
<tr>
<td><strong>Numbering Plan</strong></td>
</tr>
<tr>
<td><strong>Route Filter</strong></td>
</tr>
<tr>
<td><strong>MLPP Precedence</strong></td>
</tr>
<tr>
<td><strong>Apply Call Blocking Percentage</strong></td>
</tr>
<tr>
<td><strong>Resource Priority Namespace Network Domain</strong></td>
</tr>
<tr>
<td><strong>Route Class</strong></td>
</tr>
<tr>
<td><strong>Gateway/Route List</strong></td>
</tr>
<tr>
<td><strong>Route Option</strong></td>
</tr>
<tr>
<td><strong>External Classification</strong></td>
</tr>
<tr>
<td><strong>External Call Control Profile</strong></td>
</tr>
<tr>
<td><strong>Require Device Override</strong></td>
</tr>
<tr>
<td><strong>Provide Outside Dial Tone</strong></td>
</tr>
<tr>
<td><strong>Allow Overlap Sending</strong></td>
</tr>
<tr>
<td><strong>Allow Urgent Priority</strong></td>
</tr>
<tr>
<td><strong>Require Forced Authorization Code</strong></td>
</tr>
<tr>
<td><strong>Authorization Level</strong></td>
</tr>
</tbody>
</table>
### Route Pattern for Fax

Figure 19: Route Pattern for Fax

<table>
<thead>
<tr>
<th>Feature</th>
<th>Setting</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>External Call Control Profile</strong></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td><strong>Allow Device Override</strong></td>
<td></td>
</tr>
<tr>
<td><strong>Provide Outside Dial Tone</strong></td>
<td></td>
</tr>
<tr>
<td><strong>Allow Overlap Sending</strong></td>
<td></td>
</tr>
<tr>
<td><strong>Urgent Priority</strong></td>
<td></td>
</tr>
<tr>
<td><strong>Require Forced Authorization Code</strong></td>
<td></td>
</tr>
<tr>
<td><strong>Authorization Level</strong></td>
<td>0</td>
</tr>
<tr>
<td><strong>Require Client Matter Code</strong></td>
<td></td>
</tr>
</tbody>
</table>

#### Calling Party Transformations

<table>
<thead>
<tr>
<th>Transformation</th>
<th>Setting</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Use Calling Party's External Phone Number Mask</strong></td>
<td></td>
</tr>
<tr>
<td><strong>Calling Party Transform Mask</strong></td>
<td></td>
</tr>
<tr>
<td><strong>Prefix Digits (Outgoing Calls)</strong></td>
<td></td>
</tr>
<tr>
<td><strong>Calling Line ID Presentation</strong></td>
<td>Default</td>
</tr>
<tr>
<td><strong>Calling Name Presentation</strong></td>
<td>Default</td>
</tr>
<tr>
<td><strong>Calling Party Number Type</strong></td>
<td>Cisco CallManager</td>
</tr>
<tr>
<td><strong>Calling Party Numbering Plan</strong></td>
<td>Cisco CallManager</td>
</tr>
</tbody>
</table>

#### Connected Party Transformations

<table>
<thead>
<tr>
<th>Transformation</th>
<th>Setting</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Connected Line ID Presentation</strong></td>
<td>Default</td>
</tr>
<tr>
<td><strong>Connected Name Presentation</strong></td>
<td>Default</td>
</tr>
</tbody>
</table>

#### Called Party Transformations

<table>
<thead>
<tr>
<th>Transformation</th>
<th>Setting</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Discard Digits</strong></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td><strong>Called Party Transform Mask</strong></td>
<td>1626313XXXXX</td>
</tr>
<tr>
<td><strong>Prefix Digits (Outgoing Calls)</strong></td>
<td></td>
</tr>
<tr>
<td><strong>Called Party Number Type</strong></td>
<td>Cisco CallManager</td>
</tr>
<tr>
<td><strong>Called Party Numbering Plan</strong></td>
<td>Cisco CallManager</td>
</tr>
</tbody>
</table>

#### ISDN Network-Specific Facilities Information Element

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

---

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**Explanation**

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

**FAX ATA**

**FAX Configuration**

![Figure 21: FAX ATA Configuration](image_url)
Figure 22: FAX ATA Configuration (cont.)

<table>
<thead>
<tr>
<th>Line 1</th>
<th></th>
<th></th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>General</td>
<td>Line Enable:</td>
<td>yes ▼</td>
<td></td>
</tr>
<tr>
<td></td>
<td>Streaming Audio Server (SAS)</td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td>SAS Enable:</td>
<td>no ▼</td>
<td></td>
</tr>
<tr>
<td></td>
<td>SAS Inbound RTP Sink:</td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td>SAS DLG Refresh Interval:</td>
<td>30</td>
<td></td>
</tr>
<tr>
<td></td>
<td>NAT Settings</td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td>NAT Mapping Enable:</td>
<td>no ▼</td>
<td></td>
</tr>
<tr>
<td></td>
<td>NAT Keep Alive Message:</td>
<td>NOTIFY</td>
<td></td>
</tr>
<tr>
<td></td>
<td>NAT Keep Alive Duration:</td>
<td>no ▼</td>
<td></td>
</tr>
<tr>
<td></td>
<td>Network Settings</td>
<td></td>
<td></td>
</tr>
<tr>
<td></td>
<td>SIP ToS/Diffserv Value:</td>
<td>0x08</td>
<td></td>
</tr>
<tr>
<td></td>
<td>RTP ToS/Diffserv Value:</td>
<td>0x08</td>
<td></td>
</tr>
<tr>
<td></td>
<td>Network Jitter Level:</td>
<td>high ▼</td>
<td></td>
</tr>
<tr>
<td></td>
<td>SIP CAS Value:</td>
<td>3 [0-7]</td>
<td></td>
</tr>
<tr>
<td></td>
<td>RTP CAS Value:</td>
<td>6 [0-7]</td>
<td></td>
</tr>
<tr>
<td></td>
<td>Jitter Buffer Adjustment:</td>
<td>yes ▼</td>
<td></td>
</tr>
</tbody>
</table>

| SIP Settings | SIP Transport: | UDP ▼    |          |
|              | SIP 100REL Enable: | no ▼     |          |
|              | Auth Redirect-Immediately: | yes ▼    |          |
|              | SIP Remote-Party-ID: | yes ▼    |          |
|              | SIP Debug Option: | name ▼   |          |
|              | Restrict Source IP: | no ▼     |          |
|              | Refer Target Bye Delay: | 0        |          |
|              | Refer-To Target Contact: | no ▼     |          |
|              | Auth INVITE: | no ▼     |          |
|              | Use Anonymous With SIP ID: | yes ▼    |          |

| Call Feature Settings | Blind Attn-Axfer Enable: | no ▼     |          |
|                       | Xfer When Hangup Conf: | yes ▼    |          |
|                       | Conference Bridge Ports: | 3 ▼      |          |
|                       | Emergency Number: |          |          |
| MOH Server: | Conference Bridge URL: |          |          |
| Enable IP Dialing: | Mailbox ID: | no ▼     |          |
## Acronyms

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