CableONE: Connecting Cisco Unified Communication Manager Express (CME 12.1) using SIP

March 2018

Note: Testing was conducted in tekVizion labs

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Introduction

Cable ONE Business SIP (Session Initiation Protocol) trunk service gives your business a direct physical connection from your PBX to our nationwide, fiber-optic network — for an affordable cost. Our SIP Trunk service can power your phone system allowing you to take advantage of the latest technologies without compromising network reliability, security and continuity.

- This application note describes how to configure a Cisco Unified Communications Manager Express (Cisco Unified CME) 16.07.01, Cisco Unity Connection 12.0.1 with connectivity to CableONE SIP trunk service. The application note also covers support and configuration example Cisco Unity Connection (CUC) messaging integrated into the Cisco Unified Communications Manager Express. The deployment model covered in this application note is Customer Premises Equipment (Cisco Unified CME/CUC) to PSTN (CableONE). CableONE provides inbound and outbound call service.

- Testing was performed in accordance to CableONE test plan and all features were verified. Key features verified are: inbound and outbound basic call (including international calls), calling name delivery, calling number and name restriction, CODEC negotiation, intra-site transfers, intra-site conferencing, call hold and resume, call forward (forward all, busy and no answer), leaving and retrieving voicemail (Cisco Unity Connection), Cisco auto-attendant (BACD), fax using G.711 (G3 and SG3 speeds).

- Consult your Cisco representative for the correct IOS/IOS-XE image and for the specific application and Device Unit License and Feature License requirements for all your Cisco Unified CME.

Note: Testing was conducted in tekVizion labs

Network Topology

Figure 1. Basic Call Setup
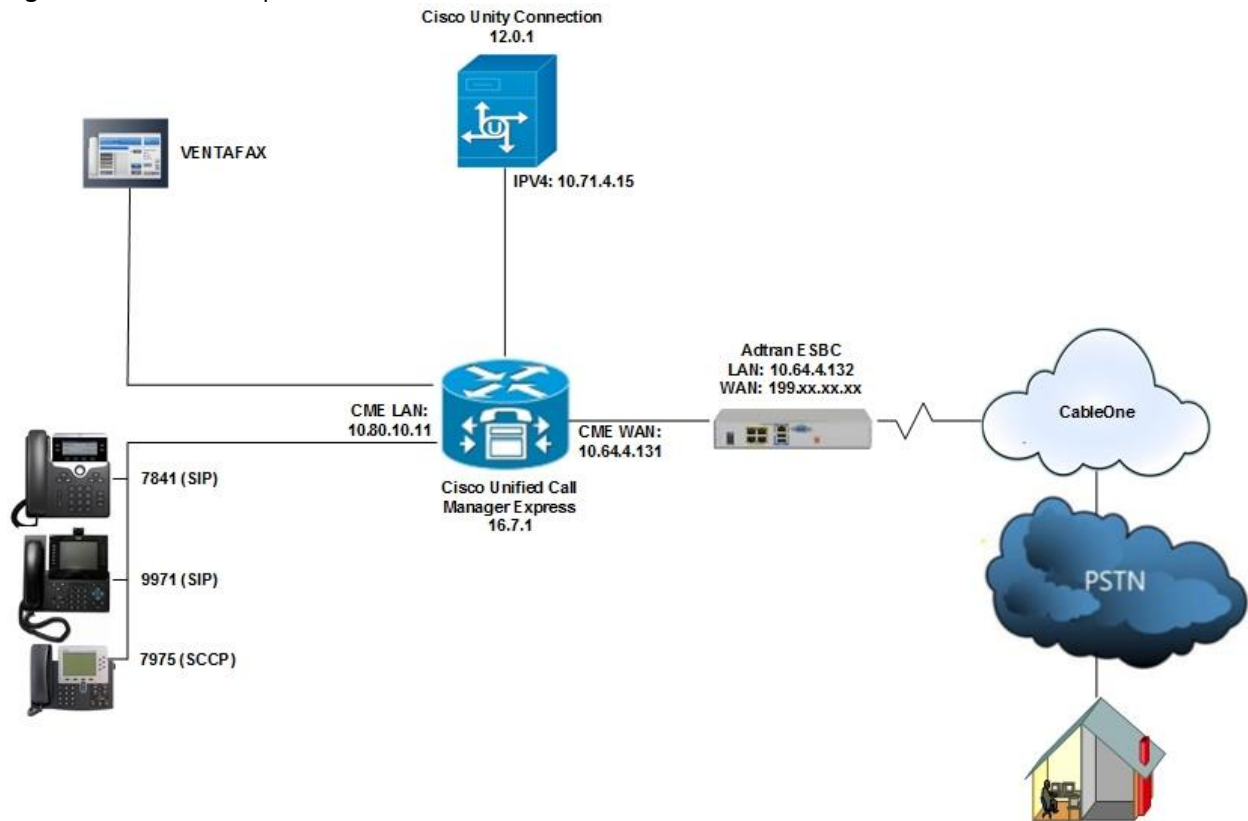
The Cisco Unified CME in Figure 1

Hardware Components
- This solution was tested with Cisco ISR4331.
- Cisco IP Phones. This solution was tested with 7841, 7975 and 9971 phones, but any Cisco IP Phone model supporting RFC2833 can be used.
- Cisco ISR4331/K9 (1RU) processor with 3841357K/6147K bytes of memory. Processor board ID FLM2141V250.
- 3 Gigabit Ethernet interfaces and 2 Voice FXS interfaces.
- Cisco Unity Connection - VMware - 1 vCPU: Intel(R) Xeon(R) CPU X5675 @ 3.07GHz.
- HDD 160 GB, Memory 4096 Mbytes RAM.

Software Requirements
- Cisco IOS-XE gateway running IOS-XE Version 16.07.01 [CME version 12.1]. This solution was tested with Cisco IOS image: "bootflash:isr4300-universalk9.16.07.01.SPA.bin" .
- This solution was tested with Cisco Unity Connection version (Version 12.0.1.21900-10) .

Note: Testing was conducted in tekVizion labs

Features

Features - Supported

- Basic Call using G711ulaw.
- Calling Party Number Presentation and Restriction.
- Calling Name.
- Call Transfer.
- Conference.
- Call Hold and Resume (See Caveat section for details).
- Call Forward All, Busy and No Answer.
- Fax over G.711 (See Caveat section for details).
- Incoming DID Translation and Routing.
- Outbound calls and Inbound calls.
- Voicemail.
- Auto-attendant (See Caveat section for details).

Features Not Supported

- Cisco IP phones used in this test do not support Blind Transfer, only Semi-attendant and Attendant transfers were tested.
- CableONE does not support G729 codec at the moment.
- Dual codecs (G711 & G729) are not supported by CableONE at the moment.

Caveats

Fax:

- T.38 (G3 or SG3) is not supported by CableONE at the moment.
- The maximum fax rate achieved using G711 Pass-through (G3 or SG3) fax protocol is only 14400 kbps.
- For G711Passthrough test achieved using "fax protocol pass-through g711ulaw".

Auto-Attendant:

- The Cisco Unified CME Basic Automatic Call Distribution (BACD) was employed to enable the auto-attendant feature. The test was performed using the default codec G711u for auto attendant prompts.

Hold & Resume:

- When Cisco CME sends hold INVITE with a=sendonly, CableONE is sending a=inactive in the 200 OK and thus MOH was not played. SCCP phones does not send hold notification in INVITE. SCCP phones were used for hold-resume test cases and MOH worked fine.

DTMF

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- Transcoder is configured in CME to convert inband voice DTMF coming from CableONE to RFC2833. DTMF is send from CME to Unity Connection as RFC2833. Unity Connection does not recognize inband voice DTMF.

Configuration Considerations

- 10 Digit BTN (Billing Telephone Number) is used to register Cisco Unified CME to CableONE SIP trunk (See configuration section for details).

- CableONE require that the SIP Diversion header contain the full 10-digit DID number of the forwarding party. In this application note the assumption was made that a typical customer will utilize extension numbers (4-digit assignments in this example) and map 10-digit DID number using Cisco Unified CME translation patterns. Because 4-digit extensions configured on Cisco Unified CME IP phones, it is necessary to expand the 4-digit extension included in the Diversion header of a forwarding INVITE message, to its full 10-digit DID number when the IP phone is set to call-forward. The requirement to expand the Diversion-Header has been achieved by the use of a SIP profile in Cisco Unified CME (See configuration section for details).

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Configuration

Cisco IOS Version

CableONEcme#show version
Cisco IOS XE Software, Version 16.07.01
Cisco IOS Software [Fuji], ISR Software (X86_64_LINUX_IOSD-UNIVERSALK9-M), Version 16.7.1, RELEASE SOFTWARE (fc6)
Technical Support: http://www.cisco.com/techsupport
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documentation or "License Notice" file accompanying the IOS-XE software,
or the applicable URL provided on the flyer accompanying the IOS-XE
software.

ROM: IOS-XE ROMMON

CableONEcme uptime is 1 week, 5 days, 16 hours, 1 minute
Uptime for this control processor is 1 week, 5 days, 16 hours, 3 minutes
System returned to ROM by PowerOn
System image file is "bootflash:isr4300-universalk9.16.07.01.SPA.bin"
Last reload reason: PowerOn

This product contains cryptographic features and is subject to United
States and local country laws governing import, export, transfer and
use. Delivery of Cisco cryptographic products does not imply
third-party authority to import, export, distribute or use encryption.
Importers, exporters, distributors and users are responsible for
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agree to comply with applicable laws and regulations. If you are unable
to comply with U.S. and local laws, return this product immediately.

A summary of U.S. laws governing Cisco cryptographic products may be found at:
http://www.cisco.com/wwl/export/crypto/tool/stqrg.html

If you require further assistance please contact us by sending email to

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export@cisco.com.

Suite License Information for Module:'esg'

Suite   Suite Current   Type   Suite Next reboot

FoundationSuiteK9   None   None   None
securityk9
appxk9

AdvUCSuiteK9   None   None   None
uck9
cme-srst
cube

Technology Package License Information:

Technology   Technology-package   Technology-package
         Current   Type   Next reboot
----
appxk9   None   None   None
uck9   uck9   Permanent   uck9
securityk9   None   None   None
ipbase   ipbasek9   Permanent   ipbasek9

cisco ISR4331/K9 (1RU) processor with 3841357K/6147K bytes of memory.
Processor board ID FLM2141V250
3 Gigabit Ethernet interfaces
2 Voice FXS interfaces
32768K bytes of non-volatile configuration memory.
8388608K bytes of physical memory.
7057407K bytes of flash memory at bootflash:.
0K bytes of WebUI ODM Files at webui:.

Configuration register is 0x2102

CableONEcme#

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Cisco Unified CME

CableONEcme#show running-config
Building configuration...

Current configuration : 6970 bytes
!
version 16.7
service timestamps debug datetime msec
service timestamps log datetime msec
service password-encryption
platform qfp utilization monitor load 80
no platform punt-keepalive disable-kernel-core
!
hostname CableONEcme
!
boot-start-marker
boot-end-marker
!
!
vrf definition Mgmt-intf
 !
 address-family ipv4
 exit-address-family
 !
 address-family ipv6
 exit-address-family
!
enable secret 5 1lY0j$81cjoxxxxxx
!
no aaa new-model
subscriber templating
!
!
multilink bundle-name authenticated
!
crypto pki trustpoint TP-self-signed-2194658987
 enrollment selfsigned
 subject-name cn=IOS-Self-Signed-Certificate-2194658987
 revocation-check none
 rsakeypair TP-self-signed-2194658987
!
!
crypto pki certificate chain TP-self-signed-2194658987
!

Note: Testing was conducted in tekVizion labs

voice service voip
 ip address trusted list
 no ip address trusted authenticate
 allow-connections h323 to h323
 allow-connections h323 to sip
 allow-connections sip to h323
 allow-connections sip to sip[1]
 no supplementary-service sip moved-temporarily
no supplementary-service sip refer
 redirect ip2ip
 fax protocol pass-through g711ulaw[2]
 sip
  session refresh[3]
  registrar server[4] expires max 600 min 60
  asserted-id pai
!
voice class codec 1[5]
 codec preference 1 g711ulaw
 codec preference 2 g729r8
!
!
voice class sip-profiles 1[6]
 request INVITE sip-header Privacy add "Privacy: id"[7]
 request INVITE sip-header Diversion add "Diversion:
<sip:4806538483@voip.CableONE.net>;privacy=off;reason=unconditional;screen=no"[8]
!
voice register global
 mode cme[9]
 source-address 10.80.10.11 port 2000[10]
 max-dn 20[11]

---

[1] This command enables Cisco Unified CME to perform basic SIP to SIP voice communication

[2] This command enables G711 fax at global level, meaning all VoIP dial-peers not configured for specific fax protocol will use this setting. G711 fax protocol may be configured under appropriate dial-peers

[3] Use the SIP session refresh command to send the session refresh request

[4] Enable Local SIP Registrar which is required for SIP phones in Cisco Unified CME

[5] This command enables multiple codec support and performs codec filtering required for correct interoperability between CableOne SIP network and Cisco Unified CME

[6] SIP Profiles can be used to manipulate SIP header attributes

[7] "request INVITE sip-header Privacy add "privacy id" is added to make call From a CPE Phone to some PSTN phone; Pass Calling Party Number (CPN), marked private and Verify display at called party phone

[8] "request INVITE sip-header Diversion add "Diversion:
<sip:4806538483@voip.cableone.net>;privacy=off;reason=unconditional;screen=no" is added to make call from CPE phone to some PSTN phone while calling number marked as anonymous.

[9] Enables the mode for configuring SIP IP phones in Cisco Unified CME

[10] This is the source address for SIP phone registration

[11] Configuration for maximum extensions

Note: Testing was conducted in tekVizion labs

```
 max-pool 10¹²
 load 9971 sip9971.9-3-1-33¹³
 load 7841 sip78xx.12-0-1-11
 mwi stutter
 tftp-path flash:¹⁴
 file text
 create profile sync 0040873931329405¹⁵
 auto-register
 !
 !
voice register dn  1
 number 8483
 name CableONE Phone1
 mwi
 !
voice register dn  2
 number 8484
 name CableONE Phone2
 mwi
 !
voice register pool  1
 busy-trigger-per-button 2
 id mac B838.6116.382D
 type 7841
 number 1 dn 1
 voice-class codec 1
 username 8483 password 1234
 !
voice register pool  2
 id mac C07B.BCA1.B811
 type 9971
 number 1 dn 2
 voice-class codec 1
 username 8484 password 1234
 !
voice moh-group 1
 moh enable-g711 "flash:music-on-hold.au"
 !
voice translation-rule 1¹⁶
 rule 1 /\(^8...\)/ /480653\1/
 !
```

---

¹² Configuration for maximum phones
¹³ Specify phone loads for each phone type
¹⁴ Setup path for tftp files
¹⁵ Create configuration files for all phones
¹⁶ This translation rule/profile is used to modify the calling number

Note: Testing was conducted in tekVizion labs

voice translation-rule 2[17]
 rule 1 /480653\(....\)/ /\1/
 !
 !
voice translation-profile To-10Dig
 translate calling 1
 translate redirect-called 1
 !
voice translation-profile to_ext
 translate called 2
 !
voice-card 0/1
 no watchdog
 !
voice-card 0/4
 dsp services dspfarm
 no watchdog
 !
license udi pid ISR4331/K9 sn FOD21381GMV
no license smart enable
diagnostic bootup level minimal
 !
spanning-tree extend system-id
 !
username cisco privilege 15 password 7 111D1C0E3243115D5424
 !
redundancy
 mode none
 !
interface GigabitEthernet0/0/0
 ip address 10.64.4.131 255.255.0.0
 negotiation auto
 !
interface GigabitEthernet0/0/1
 ip address 10.80.10.11 255.255.255.0
 negotiation auto
 !
interface GigabitEthernet0/0/2
 no ip address
 negotiation auto
 !
interface Service-Engine0/1/0
 no ip address
 !
interface Service-Engine0/4/0

---

[17] This translation rule/profile is used to modify the called number

Note: Testing was conducted in tekVizion labs

```
!
interface GigabitEthernet0
 vrf forwarding Mgmt-intf
 no ip address
 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 tftp source-interface GigabitEthernet0/0/1
ip route 0.0.0.0 0.0.0.0 10.64.1.1
ip route 10.71.4.0 255.255.255.0 10.80.10.1
ip route 172.16.24.0 255.255.248.0 10.80.10.1
!
tftp-server flash:P0xxxxxxxxxx.bin
tftp-server flash:SampleAudioSource.wav
tftp-server flash:ToneOnHold.ulaw.wav
tftp-server flash:moh_g711u_music.wav
tftp-server flash:music-on-hold.au
tftp-server flash:dkern9971.100609R2-9-3-1-33.sebn
tftp-server flash:kern9971.9-3-1-33.sebn
tftp-server flash:rootfs9971.9-3-1-33.sebn
tftp-server flash:sboot9971.031610R1-9-3-1-33.sebn
tftp-server flash:sip9971.9-3-1-33.loads
tftp-server flash:skern9971.022809R2-9-3-1-33.sebn
tftp-server flash:kern2.78xx.12-0-1-11.sbn
tftp-server flash:kern78xx.12-0-1-11.sbn
tftp-server flash:rootfs2.78xx.12-0-1-11.sbn
tftp-server flash:rootfs78xx.12-0-1-11.sbn
tftp-server flash:sboot2.78xx.12-0-1-11.sbn
tftp-server flash:sboot78xx.12-0-1-11.sbn
tftp-server flash:sip78xx.12-0-1-11.loads
tftp-server flash:apps75.9-4-2ES26.sbn
tftp-server flash:cnu75.9-4-2ES26.sbn
tftp-server flash:cvm75sccp.9-4-2ES26.sbn
tftp-server flash:dsp75.9-4-2ES26.sbn
tftp-server flash:jar75sccp.9-4-2ES26.sbn
tftp-server flash:SCCP75.9-4-2SR3-1S.loads
tftp-server flash:term75.default.loads
tftp-server flash:RedHotChiliPeppers.wav
tftp-server flash:groove.wav
!
!
control-plane
!
```

Note: Testing was conducted in tekVizion labs

```
!
voice-port 0/1/0
 cptone IN
 station-id number 4806539365
 caller-id enable
!
voice-port 0/1/1
!
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
!
telephony-service
 conference transfer-pattern
 max-ephones 20
 max-dn 20
 ip source-address 10.80.10.11 port 2000
 load 7975 SCCP75.9-4-2SR3-1S
 time-zone 8
 voicemail 7777[18]
 max-conferences 8 gain -6
 call-forward pattern .T
 moh enable-g711 "music-on-hold.au"[19]
 transfer-system full-consult
 transfer-pattern .T
 create cnf-files version-stamp Jan 01 2002 00:00:00
!
dspfarm profile 1 transcode universal[20]
 codec g711ulaw
 maximum sessions 12
 associate application CUBE
!
dial-peer voice 101 voip
 description inbound from CableONE
 translation-profile incoming to_ext
 session protocol sipv2
 session transport udp
 incoming called-number 480653....
 voice-class codec 1
 no vad
```

---

[18] 7777 - Voice mail pilot number used in this example
[19] This is Music on Hold file configured in this example
[20] Transcoding profile used in this example

Note: Testing was conducted in tekVizion labs

```
!
dial-peer voice 100 voip
 description Outbound Call to CableONE
 translation-profile outgoing To-10Dig
 destination-pattern .T
 session protocol sipv2
 session target ipv4:10.64.4.132:5060[21]
 session transport udp
 voice-class codec 1[22]
 dtmf-relay rtp-nte
 no vad
!
dial-peer voice 102 voip[23]
 description towards CUC
 destination-pattern 7777
 session protocol sipv2
 session target ipv4:10.71.4.15:5060[24]
 session transport udp
 voice-class codec 1
 dtmf-relay rtp-nte
 no vad
!
dial-peer voice 104 pots
 preference 2
 service session
 destination-pattern 9365
 no digit-strip
 port 0/1/0
!
!
sip-ua
 mwi-server ipv4:10.71.4.15 expires 3600 port 5060 transport udp unsolicited
!
ephone-dn  1  dual-line
 number 8485
 label 8485 for CableONE
 name CableONE Phone
 call-forward busy 12142425962
 call-forward noan 7777 timeout 20
 mwi sip
 moh-group 1
!
```

---

[21] This command sets the SIP server target for outgoing SIP calls

[22] This command assigns the voice class codec setting to this dial-peer

[23] Dial peer used to route the calls to Cisco unity connection server for Voicemail access

[24] This command sets the Cisco unity connection server target for voicemail calls

Note: Testing was conducted in tekVizion labs

```
!
ephone-dn  2  dual-line
 number 8487
 label 8487 for CableONE
 name CableONE Phone
 mwi sip
 moh-group 1
!
!
ephone  1
 mac-address 0008.3031.EB46
 username "8485" password 1234
 type 7975
 mwi-line 1
 button  1:1
!
ephone  2
 mac-address 0008.3031.F49B
 type 7975
 mwi-line 1
 button  1:2
!
line con 0
 exec-timeout 0 0
 password 7 06120A2A7A1F13485519
 login
 transport input none
 stopbits 1
line aux 0
 stopbits 1
line vty 0 4
 password 7 105A0C123346085A5C0A
 login local
 transport input telnet
!
wsma agent exec
!
wsma agent config
!
wsma agent filesys
!
wsma agent notify
!
!
end

CableONEcme#
```

Note: Testing was conducted in tekVizion labs

Note: Testing was conducted in tekVizion labs

Cisco Unity Connection

Version Details

Cisco Unity Connection User Configuration

Navigation: Cisco Unity Connection → Users → Users

Set Alias*= 8485. This is used for this example.

Set First Name = 8485 is used to identify this User.

Set Last Name* = 8485. This is used for this example

Set Display Name= 8485. This is used in this example.

Set SMTP Address =8485. This is used in this example.

Set Extension = 8485. This is used in this example.

Set Phone System= CME. This is used in this example.

All other values are default.

Cisco Unity Connection User Configuration (Continued)

All values are default.

Note: Testing was conducted in tekVizion labs

Cisco Unity Connection Administration
For Cisco Unified Communications Solutions

Navigation Cisco Unity Connection Administration ▼ Go

administrator | Search Documentation | About | Sign Out

Cisco Unity Connection

- Users
  - Users
  - Import Users
  - Synch Users
- Class of Service
- Templates
- Contacts
- Distribution Lists
- Call Management
- Message Storage
- Networking
- Unified Messaging
- Video
- Dial Plan
- System Settings
- Telephony Integrations
- Tools

Class of Service   Voice Mail User COS ▼
Active Schedule   Weekdays ▼ View

☐ Set for Self-enrollment at Next Sign-In
☑ List in Directory
☑ Send Non-Delivery Receipts on Failed Message Delivery
☐ Skip PIN When Calling From a Known Extension
☐ Use Short Calendar Caching Poll Interval

Recorded Name

8485
Number or URI   30   Volume   1x Speed

Location

Address
Building
City
State
Postal Code
Country   United States ▼
☑ Use System Default Time Zone
Time Zone   (GMT-06:00) America/Chicago ▼
Language   ⦿ Use System Default Language
                ◯ English(United States) ▼
Department
Manager
Billing ID
Corporate Email Address

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Cisco Unity Connection Telephony Integration

Navigation: Telephony Integrations → Phone system

Set System Name* = CME. This Name used for this example.

Port Group

Navigation: Telephony Integration → Port Group.

Set Display Name* = CME-1. This Name used for this example.

Check Register with SIP server.

Navigation Path: Telephony Integration→ Port Group → Edit → Servers.

Note: Testing was conducted in tekVizion labs

Port

Set Port Name = CME-1-001. This Name used for this example.

Phone System = CME.

Port Group = CME-1.

Server = unity-unity2.lab.tekvizion.com. This Name used for this example.

Note: Testing was conducted in tekVizion labs

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Acronyms

SIP	Session Initiation Protocol
MGCP	Media Gateway Control Protocol
SCCP	Skinny Client Control Protocol
Cisco Unified CME	Cisco Unified Communications Manager Express
SP	Service Provider
PSTN	Public switched telephone network
DTMF	Dual Tone Multi-Frequency
CUC	Cisco Unity Connection
VOIP	Voice Over Internet Protocol
IP	Internet Protocol
TDM	Time-division multiplexing
BTN	Billed (or Billing) Telephone Number
CODEC	Coder-Decoder (in this document a device used to digitize and un digitize voice signals)

Note: Testing was conducted in tekVizion labs

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