# Cisco CallManager Express (CME) SIP中继配置示例

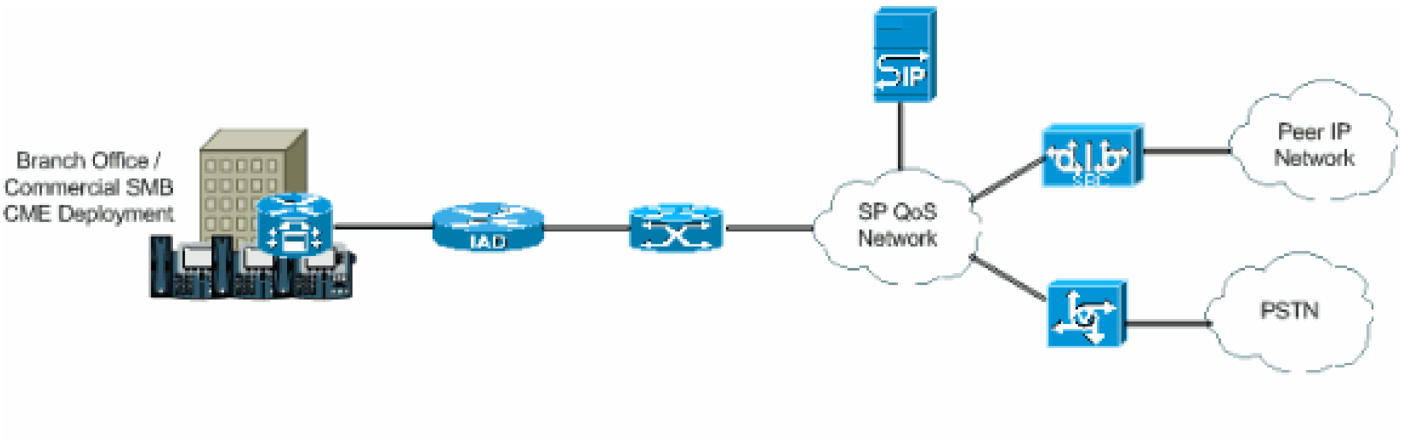
## Contents



## Introduction

目前，电信行业正在经历从建立已久的交换和传输技术向基于 IP 的传输和边缘设备的转变。IP 通信革命已经开始在中小型企业中产生巨大的商业影响。这些中小企业意识到使用IP是非常高效的，因为IP能使用在单个网络的语音、视频和数据功能，而不是使用三独立的专用网络。图1显示趋向往IP Trunking的IP电话配置。

**图1 - IP电话系统**




IP PBX在语音技术的事务开始占优势，并且TDM PBX不再是主时钟源作为去在两个语音网络之间的交叉。TDM PBX的使用方法近两三年来减少了，并且使用IP PBX成为在IP LAN和WAN的有利的投资。为了连接到PSTN， PBX需要某类Trunking例如TDM (T1/E1)或模拟线路。使用Trunk的这些类型IP PBX能访问PSTN，但是需要转换IP语音流量成传统PSTN，能有时导致连续的转换从IP域到TDM域的媒体网关。这些连续的转换增加网关的维修费用，增量潜伏期，并且减少语音质量。

为了避免这些问题， IP PBX使用协议会话开始和管理，最突出哪些是会话初始化协议(SIP)。本文在SIP Trunking和Cisco CallManager Express (CME)提供一个说明使用Inbound和Outbound呼叫的， SIP Trunking和一种配置应用有CME的一个基于IP的电话系统。

# Prerequisites

## Requirements

尝试进行此配置之前，请确保满足以下要求：

- 安装CME版本4.1
- Cisco IOS软件版本12.4(11)XJ或IOS 12.4(6th)T的镜像在路由器
- NM-CUE模块安装有提示版本2.3.4

## Components Used

本文档中的信息基于以下软件和硬件版本：

- 在Cisco IOS Software Release 12.4(11)XJ的Cisco 3825 Router
- 在Cisco IOS Software Release 12.4的Cisco Catalyst 3550 Switch
- Cisco IP 7960 电话
- Cisco CallManager Express 4.1
- Cisco Unity Express 2.3.4

The information in this document was created from the devices in a specific lab environment.All of the devices used in this document started with a cleared (default) configuration.If your network is live, make sure that you understand the potential impact of any command.

## Conventions

有关文档规则的详细信息，请参阅 Cisco 技术提示规则。

# SIP协议

SIP是基于的ASCII，应用层能使用建立，维持和终止在两个或多个终端之间的呼叫的控制协议。SIP迅速地涌现了作为用于IP通信的标准协议，因为它是除语音之外，能使用视频会话和即时消息的多媒体协议。并且， SIP能处理会议会话和广播，以及一对一的会话。SIP有巨大可能性在变换和开发方式人民沟通。为此， Cisco有并且继续在采取领导的播放一的重要的角色创建提出SIP和其申请IP通信标准的新技术。

SIP Trunk类似于电话线路，除了SIP Trunk使用IP网络，不是PSTN。另外， SIP Trunk允许语音和数据收敛在普通的所有IP连接上。使用SIP Trunk，为了访问IP网络，是必要的配置在服务提供商被做，以及在用户侧。用户需求设置和配置CME，是PBX将足够解释SIP信号并且顺利地通过数据流。服务提供商需要配置SIP代理服务器。然而， SIP Trunk比正常PSTN Trunk是复杂化设立。原因是用户由设备供应商面对在处理SIP的不同的解释和实施的挑战，提供安全，管理服务质量(QoS)，启用网络地址转换(NAT)和防火墙穿越，和保证载波级别服务可靠性和连续性。

这些点描述SIP Trunk为什么变得很明显在中小企业：

- 快速和容易的配置
- 被改进的网络利用率容量
- 在统一和降低电话费用的可能性
- 经济直接拨入(DID)
- 企业连续性

# CME SIP Trunk技术支持

Cisco CME是集成直接地Cisco IOS软件的IP电话解决方案。CME允许中小企业配置语音、数据和视频在单个平台。IP电话网络是简单设置，因为CME在单个路由器运行，提供企业的一个PBX功能。所以，使用单个融合的解决方案以最低费用，通过使用CME，中小企业能提供IP电话和数据路由。

## SIP Trunk的DTMF中继

当发布了， CME开始支持SIP Trunking CME 3.1。然而，当SIP电话呼叫SCCP电话或设法访问语音邮件，一些问题存在了。问题是SCCP电话被连接到CME要求使用带外DTMF中继传输DTMF (位)在VoIP连接间，并且SIP电话使用在波段之内传输。DTMF失真存在了在两个设备之间。当发布了CME 3.2，技术支持被添加了到DTMF中继。从SCCP的DTMF位能转换成在波段之内DTMF中继机制通过RFC2833或通知方法。

CME当前支持SIP的DTMF互联网络此列表对SIP呼叫：

- 通知<--->请从12.4(4)T通知
- RFC2833 <--->请从12.4(4)T通知
- 通知<---> RFC2833从12.4(4)T
- 同带信号传输G711 <--->从12.4(11)T **[Requires Transcoder]**

CME当前支持SIP的此DTMF互联网络对SCCP呼叫：

- 带外SCCP - SIP通知/RFC2833从12.4(4)T

## 编码技术支持和转码

要考虑的另一个重要方面，当您设置SIP Trunk时是支持的编码。编码表示信号的脉冲代码调制示例在语音频率。SIP Trunk支持这些编码：G.711和G.729。然而，对于不同的功能例如Cisco Unity Express (CUE)和Music on Hold (MOH)，支持仅编码G.711。这意味着使用SIP Trunk使用编码G.729的语音呼叫不能访问提示，除非转码器存在允许语音流压缩和解压匹配提示功能。MOH能也使用编码G.729保存带宽，但是编码不提供足够的质量MOH流。这归结于事实G.729为语音优化。所以，您必须强制MOH使用G.711。

## 今后呼叫

当呼叫在SIP Trunk进来并且转送(CFNA/CFB/CFA)时，然后默认行为是为了CME能发送302"临时地被移动的" SIP消息到服务提供商(SP)代理。联系报头的用户部分在302消息的也许需要被转换反射SP代理能路由的DID的。应该修改联系报头的主机部分在302消息的反射记录(AOR)的地址使用**host-registrar** CLI在SIP UA下和**b2bua** CLI在去的VoIP拨号对等体下到提示。

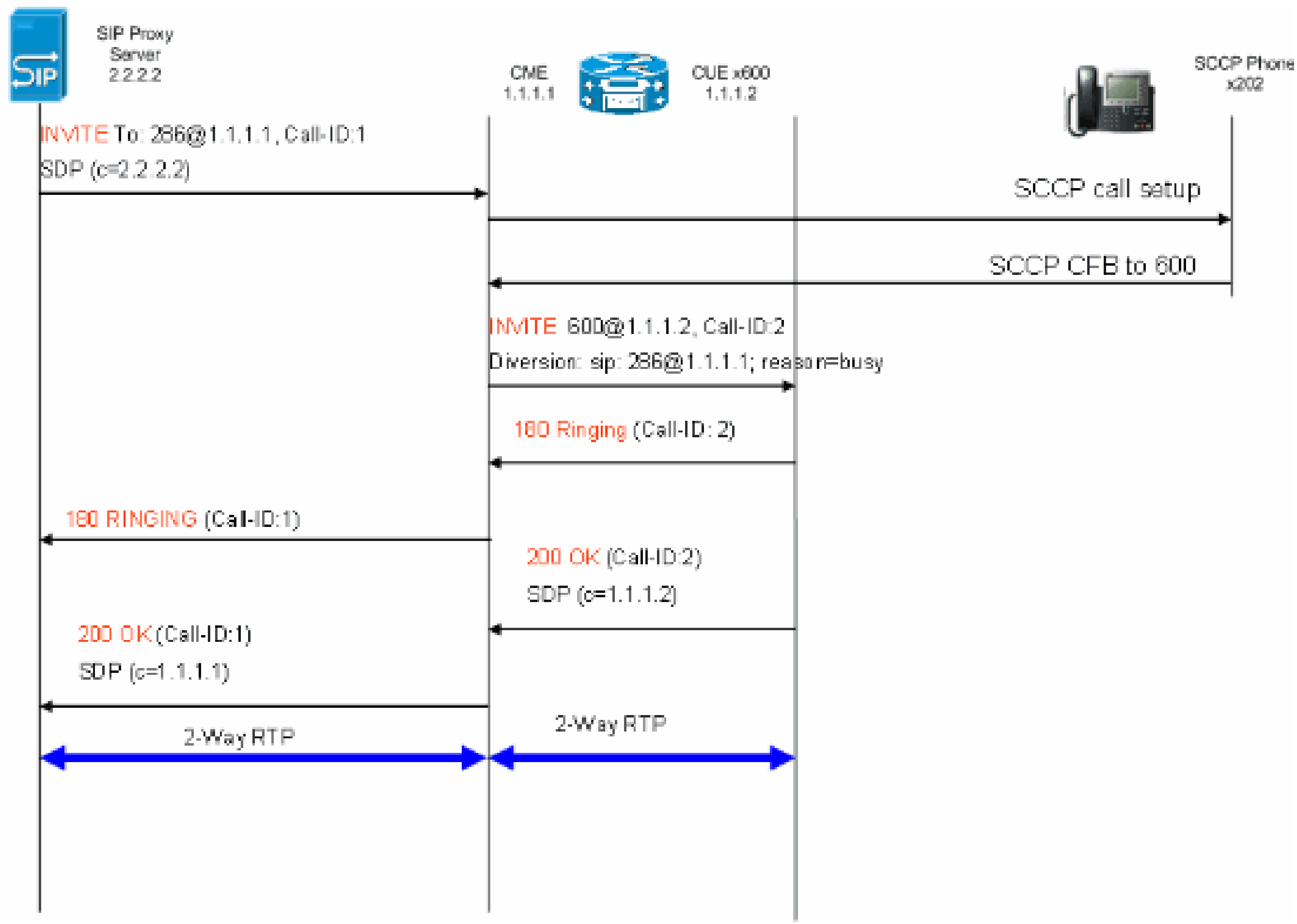
一些SIP代理也许不支持此。如果那样，您然后需要添加此：

```
Router(config)#voice service voip
```

```
Router(conf-voi-serv)#no supplementary-service sip moved-temporarily
```

当302消息是失效的时，图2显示CME系统的工作情况。

**图2 -呼叫转发忙(CFB)流同302停用的消息**




此方法在CME将允许302个SIP消息的发夹呼叫的转发。也需要以上是否有有没有DID的映射的某一

扩展，因为SP代理也许不会路由这样呼叫。如果禁用3xx回应，**呼叫号码发起者**可以用于保留原始主叫方的呼叫方ID。

## 呼叫转移

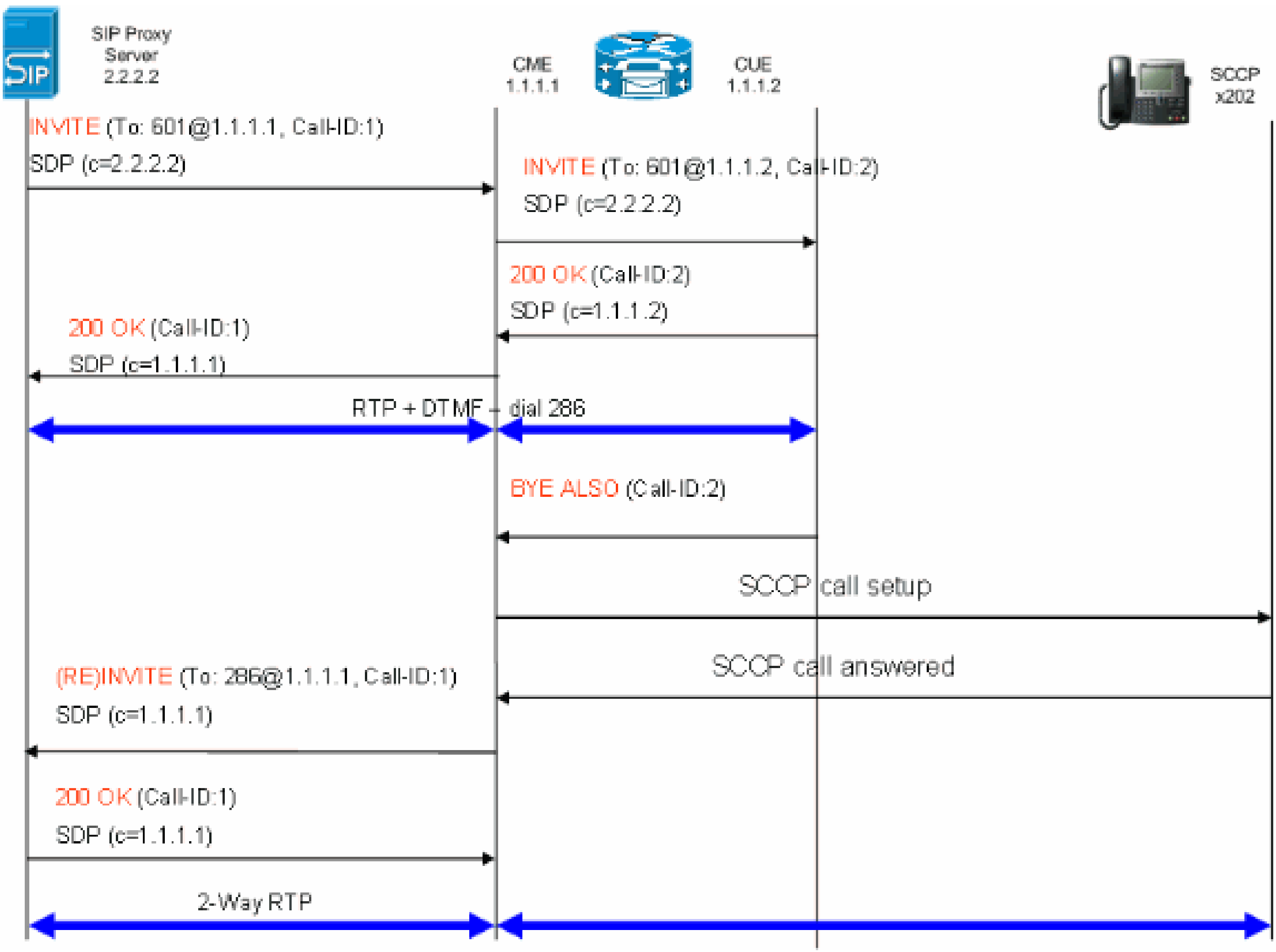
当呼叫在SIP Trunk进来到SCCP电话或提示AutoAttendant (AA)和调用，默认情况下CME将发送SIP参考消息SP代理。多数SP代理服务器不支持参考方法。需要配置这为了强制CME到两隧道间的本地交换呼叫：

```
Router(config)#voice service voip
```

```
Router(conf-voi-serv)#no supplementary-service sip refer
```

图3显示同参考方法停用的CME系统的工作情况。

**图3 -与REFER的转移禁用**




如果SIP代理支持REFER，必须转换是指的用户部分和参考由为SP代理了解的DID的。是指的主机部分和参考由字段必须是默认情况下SP代理能路由到的IP地址或DNS (这在CME 4.1)发生。

## 呼叫暂挂

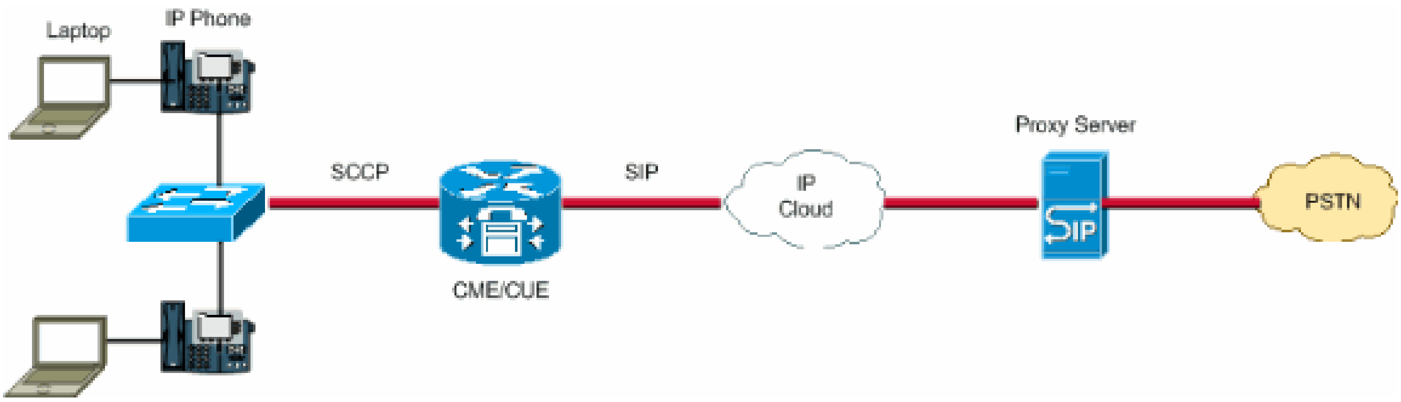
如果SCCP电话暂挂中发出从PSTN的呼叫，本地CME更改媒体。SIP信息在SIP Trunk没有传送。Music on Hold将被演奏给在根据CME配置的SIP Trunk间的用户。

# Configure

本部分提供有关如何配置本文档所述功能的信息。

## Network Diagram

本文档使用以下网络设置：




## 配置

这些配置元素提供要求的步骤的分级显示用SIP Trunk配置您的CME ：

- 基础设施元素：接口、TFTP和DHCP服务， NTP等等。
- Telephony-service ：Enable (event)在CME平台的IOS “PBX”呼叫控制包括电话管理的元素
- 的Ephone Ephone dns ：定义IP电话和他们的电话号码
- 拨号计划：拨号点，扩展，语音转换规则
- IOS SIP配置：Enable (event) SIP、电话注册与SIP代理，呼叫路由在Trunk等等。
- 语音邮件技术支持：Cisco Unity Express
- 交换机Catalyst配置：IP地址、接口等等。

这是必要的完整的配置配置与SIP Trunk的一个CME系统：

### 路由器- CME配置

```
!
AUSNML-3825-01#show run
Building configuration...

Current configuration : 8634 bytes
!
version 12.4
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname AUSNML-3825-01
!
boot-start-marker
boot-end-marker
!
```

```
enable secret 5 $1$vBU1$MCMG1rXM5ejME8Wap6W0H1
!
no aaa new-model
clock timezone central -8
clock summer-time central recurring
ip cef
!
!--- DHCP Configuration --- ip dhcp pool Voice network 172.22.100.0 255.255.255.0 option 150 ip 172.22.
default-router 172.22.100.1 ! ip dhcp pool Data network 172.22.101.0 255.255.255.0 option 150 ip 172.22
default-router 172.22.101.1 ! ! ip domain name cisco.com ip name-server 205.152.0.20 multilink bundle-n
authenticated ! voice-card 0 no dspfarm ! ! ! ! !--- Voice Class and Service VoIP Configuration --- voi
service voip allow-connections sip to sip no supplementary-service sip moved-temporarily !---Disable 30
sending no supplementary-service sip refer !---Disable REFER sending sip registrar server expires max 3
min 3600 localhost dns:domain.test.com ! ! voice class codec 1 codec preference 1 g711ulaw ! ! ! ! ! !
! ! !--- Voice Translation Rules --- voice translation-rule 1 rule 1 /5123781291/ /601/ !--- An inbound
for AA pilot "601 rule 2 /5123781290/ /600/ !--- An inbound rule for the voicemail pilot "600" ! voice
translation-rule 2 rule 1 /^911$/ /911/ !--- An outbound rule to allow "911" rule 2 /^9\(.*\)/ /\1/ !--
outbound rule to strip "9" from PSTN calls ! voice translation-rule 3 rule 1 /^.*/ /5123781291/ !--- An
outbound rule to change calling-number CLID to a
!--- "main" number ! voice translation-rule 4 rule 1 /^9(.......)$/ /512\1/ !--- An outbound rule to ad
areacode for local calls rule 2 /600/ /5123788000/ !--- An outbound rule to present the voicemail pilot
extension as DID rule 3 /601/ /5123788001/ !--- An outbound rule to present the AA pilot extension as D
rule 4 /^2(..)$/ /51237812\1/ !--- An outbound rule to support transfers and call-forwards rule 5 /^9(.
/\1/ !--- An outbound rule to strip "9" from "9+" transfers and call-forwards ! ! voice translation-pro
CUE_Voicemail/AutoAttendant !--- Applied to the inbound dial-peers for CUE translate called 1 ! voice
translation-profile PSTN_CallForwarding !--- Applied to CUE dial-peers translate redirect-target 4 tran
redirect-called 4 ! voice translation-profile PSTN_Outgoing !--- Applied to all outbound dial-peers tra
calling 3 translate called 2 translate redirect-target 4 translate redirect-called 4 ! ! ! ! ! ! ! vlan
internal allocation policy ascending ! ! ! ! !--- Internet Connection Configuration --- interface
GigabitEthernet0/0 no ip address duplex auto speed auto media-type rj45 no keepalive ! interface
GigabitEthernet0/0.1 encapsulation dot1Q 1 native ip address 172.22.1.71 255.255.255.0 ! interface
GigabitEthernet0/0.20 encapsulation dot1Q 20 ip address 172.22.101.1 255.255.255.0 ! interface
GigabitEthernet0/0.100 encapsulation dot1Q 100 ip address 172.22.100.1 255.255.255.0 ! interface
GigabitEthernet0/1 no ip address shutdown duplex auto speed auto media-type rj45 no keepalive ! interfa
Service-Engine1/0 ip unnumbered GigabitEthernet0/0.1 service-module ip address 172.22.1.253 255.255.255
service-module ip default-gateway 172.22.1.71 ! ip route 0.0.0.0 0.0.0.0 172.22.1.1 ip route 172.22.1.2
255.255.255.255 Service-Engine1/0 ! ! ip http server no ip http secure-server ! ! ! !--- TFTP Server
Configuration --- tftp-server flash:P0030702T023.bin tftp-server flash:P0030702T023.loads tftp-server
flash:P0030702T023.sb2 tftp-server flash:P0030702T023.sbn ! control-plane ! ! ! ! ! ! ! !--- SIP Trunk
Configuration --- dial-peer voice 1 voip description **Incoming Call from SIP Trunk** translation-profi
incoming CUE_Voicemail/AutoAttendant voice-class codec 1 voice-class sip dtmf-relay force rtp-nte sessi
protocol sipv2 session target sip-server incoming called-number .% dtmf-relay rtp-nte no vad ! ! ! dial
voice 2 voip description **Outgoing Call to SIP Trunk** translation-profile outgoing PSTN_Outgoing
destination-pattern 9........ voice-class codec 1 voice-class sip dtmf-relay force rtp-nte session prot
sipv2 session target sip-server dtmf-relay rtp-nte no vad ! ! ! dial-peer voice 3 voip description **Ou
Call to SIP Trunk** translation-profile outgoing PSTN_Outgoing destination-pattern 9[2-9]..[2-9]......
class codec 1 voice-class sip dtmf-relay force rtp-nte session protocol sipv2 session target sip-server
relay rtp-nte no vad ! ! ! dial-peer voice 4 voip description **Outgoing Call to SIP Trunk** translatio
profile outgoing PSTN_Outgoing destination-pattern 9[0-1][2-9]..[2-9]...... voice-class codec 1 voice-c
sip dtmf-relay force rtp-nte session protocol sipv2 session target sip-server dtmf-relay rtp-nte no vad
dial-peer voice 5 voip description **911 Outgoing Call to SIP Trunk** translation-profile outgoing
PSTN_Outgoing destination-pattern 911 voice-class codec 1 voice-class sip dtmf-relay force rtp-nte sess
protocol sipv2 session target sip-server dtmf-relay rtp-nte no vad ! ! ! dial-peer voice 6 voip descrip
**Emergency Outgoing Call to SIP Trunk** translation-profile outgoing PSTN_Outgoing destination-pattern
voice-class codec 1 voice-class sip dtmf-relay force rtp-nte session protocol sipv2 session target sip-
dtmf-relay rtp-nte no vad ! ! ! dial-peer voice 7 voip description **911/411 Outgoing Call to SIP Trunk
translation-profile outgoing PSTN_Outgoing destination-pattern 9[2-9]11 voice-class codec 1 voice-class
dtmf-relay force rtp-nte session protocol sipv2 session target sip-server dtmf-relay rtp-nte no vad ! !
dial-peer voice 8 voip description **International Outgoing Call to SIP Trunk** translation-profile out
PSTN_Outgoing destination-pattern 9011T voice-class codec 1 voice-class sip dtmf-relay force rtp-nte se
protocol sipv2 session target sip-server dtmf-relay rtp-nte no vad ! ! ! dial-peer voice 9 voip descrip
**Star Code to SIP Trunk** destination-pattern *.. voice-class codec 1 voice-class sip dtmf-relay force
nte session protocol sipv2 session target sip-server dtmf-relay rtp-nte no vad ! ! ! !--- Voicemail
Configuration --- dial-peer voice 10 voip description **CUE Voicemail** translation-profile outgoing
```

```
PSTN_CallForwarding destination-pattern 600 b2bua !--- Used by CME to send its IP address to SP proxy i
of CUE session protocol sipv2 session target ipv4:172.22.1.155 dtmf-relay sip-notify !--- This can also
RFC2833 going to CUE codec g711ulaw !--- CUE only supports G711ulaw as the codec no vad !--- With VAD
enabled, messages left on CUE could be blank or poor quality ! ! ! dial-peer voice 11 voip description
Auto Attendant** translation-profile outgoing PSTN_CallForwarding destination-pattern 601 b2bua session
protocol sipv2 session target ipv4:172.22.1.155 dtmf-relay sip-notify codec g711ulaw no vad ! ! !--- SI
Configuration --- sip-ua authentication username 5123781000 password 075A701E1D5E415447425B no remote-p
id retry invite 2 retry register 10 retry options 0 timers connect 100 registrar dns:domain.test.com ex
3600 sip-server dns:domain.test.com host-registrar ! ! !--- CME Telephony Service Configuration ---
telephony-service no auto-reg-ephone load 7960-7940 P0030702T023 max-ephones 168 max-dn 500 ip source-a
172.22.1.107 port 2000 calling-number initiator !--- Preserves the caller-id of a call when transferred
forwarded dialplan-pattern 1 51237812.. extension-length 3 extension-pattern 2.. no-reg voicemail 600 m
conferences 12 gain -6 call-forward pattern .T call-forward system redirecting-expanded !--- Enables
translation rule features for call-forwarding moh music-on-hold.au transfer-system full-consult dss tra
pattern 9.T secondary-dialtone 9 create cnf-files version-stamp Jan 01 2002 00:00:00 ! ! !--- Ephone an
Ephone-dn Configuration --- ephone-dn 11 dual-line number 201 secondary 5123781201 no-reg both !---"no-
both" means do not try to register either extension with SP SIP Proxy name John Smith call-forward busy
call-forward noan 600 timeout 15 ! ! ephone-dn 12 dual-line number 202 secondary 5123781202 no-reg both
Enrique Zurita call-forward busy 600 call-forward noan 600 timeout 15 ! ! ephone-dn 13 number 512378800
description **DID Number for Voicemail** ! ! ephone-dn 14 number 5123788001 description **DID Number fo
Attendant* ! ! ephone-dn 15 number 8000... no-reg primary mwi on ! ! ephone-dn 16 number 8001... no-reg
primary mwi off ! ! ephone 1 mac-address 0008.A371.28E9 type 7960 button 1:11 ! ! ! ephone 2 mac-addres
0008.A346.5C7F type 7960 button 1:12 ! ! ! ! line con 0 stopbits 1 line aux 0 stopbits 1 line 66 no
activation-character no exec transport preferred none transport input all transport output pad telnet r
lapb-ta mop udptn v120 ssh line vty 0 4 password ut69coe login ! scheduler allocate 20000 1000 ntp serv
172.22.1.107 ! end
```

### 路由器-提示配置

```
se-172-22-1-253#show run

Generating configuration:


clock timezone America/Chicago

hostname se-172-22-1-253

ip domain-name localdomain



groupname Administrators create
groupname Broadcasters create

!--- Users --- username Enrique create username John create username Enrique phonenumberE164 "512378120
username John phonenumberE164 "5123781201" username Enrique phonenumber "202" username John phonenumber
!--- AutoAttendant --- ccn application autoattendant description "**AutoAttendant**" enabled maxsession
script "aa.aef" parameter "busOpenPrompt" "AABusinessOpen.wav" parameter "operExtn" "601" parameter
"welcomePrompt" "AAWelcome.wav" parameter "disconnectAfterMenu" "false" parameter "busClosedPrompt"
"AABusinessClosed.wav" parameter "allowExternalTransfers" "false" parameter "holidayPrompt"
"AAHolidayPrompt.wav" parameter "businessSchedule" "systemschedule" parameter "MaxRetry" "3" end applic
!--- MWI --- ccn application ciscomwiapplication description "ciscomwiapplication" enabled maxsessions
script "setmwi.aef" parameter "CallControlGroupID" "0" parameter "strMWI_OFF_DN" "8001" parameter
"strMWI_ON_DN" "8000" end application !--- Voicemail --- ccn application voicemail description
"**Voicemail**" enabled maxsessions 4 script "voicebrowser.aef" parameter "uri"
"http://localhost/voicemail/vxmlscripts/login.vxml" parameter "logoutUri"
"http://localhost/voicemail/vxmlscripts/mbxLogout.jsp" end application !--- SIP --- ccn subsystem sip g
address "172.22.100.1" !--- Must match the "ip source-address" in telephony-service dtmf-relay sip-noti
sip outcall !--- Subscribe / Notify and Unsolicited Notify have not been tested transfer-mode blind bye
!--- Testing with REFER method on CUE has caused certain call flows to break end subsystem !--- Trigger
Phones --- ccn trigger sip phonenumber 600 application "voicemail" enabled maxsessions 4 end trigger cc
trigger sip phonenumber 601 application "autoattendant" enabled maxsessions 4 end trigger service phone
authentication end phone-authentication service voiceview enable end voiceview !--- Voicemail Mailboxes
voicemail default mailboxsize 21120 voicemail broadcast recording time 300 voicemail mailbox owner "Enr
```

```
size 300 description "**Enrique_Mailbox**" expiration time 10 messagesize 120 end mailbox voicemail mai
owner "John" size 300 description "**John'sMailbox**" expiration time 10 messagesize 120 end mailbox en
```

**交换机配置**

```
se-172-22-1-253#show run

Generating configuration:


clock timezone America/Chicago

hostname se-172-22-1-253

ip domain-name localdomain



groupname Administrators create
groupname Broadcasters create

!--- Users --- username Enrique create username John create username Enrique phonenumberE164 "512378120
username John phonenumberE164 "5123781201" username Enrique phonenumber "202" username John phonenumber
!--- AutoAttendant --- ccn application autoattendant description "**AutoAttendant**" enabled maxsession
script "aa.aef" parameter "busOpenPrompt" "AABusinessOpen.wav" parameter "operExtn" "601" parameter
"welcomePrompt" "AAWelcome.wav" parameter "disconnectAfterMenu" "false" parameter "busClosedPrompt"
"AABusinessClosed.wav" parameter "allowExternalTransfers" "false" parameter "holidayPrompt"
"AAHolidayPrompt.wav" parameter "businessSchedule" "systemschedule" parameter "MaxRetry" "3" end applic
!--- MWI --- ccn application ciscomwiapplication description "ciscomwiapplication" enabled maxsessions
script "setmwi.aef" parameter "CallControlGroupID" "0" parameter "strMWI_OFF_DN" "8001" parameter
"strMWI_ON_DN" "8000" end application !--- Voicemail --- ccn application voicemail description
"**Voicemail**" enabled maxsessions 4 script "voicebrowser.aef" parameter "uri"
"http://localhost/voicemail/vxmlscripts/login.vxml" parameter "logoutUri"
"http://localhost/voicemail/vxmlscripts/mbxLogout.jsp" end application !--- SIP --- ccn subsystem sip g
address "172.22.100.1" !--- Must match the "ip source-address" in telephony-service dtmf-relay sip-noti
sip outcall !--- Subscribe / Notify and Unsolicited Notify have not been tested transfer-mode blind bye
!--- Testing with REFER method on CUE has caused certain call flows to break end subsystem !--- Trigger
Phones --- ccn trigger sip phonenumber 600 application "voicemail" enabled maxsessions 4 end trigger cc
trigger sip phonenumber 601 application "autoattendant" enabled maxsessions 4 end trigger service phone
authentication end phone-authentication service voiceview enable end voiceview !--- Voicemail Mailboxes
voicemail default mailboxsize 21120 voicemail broadcast recording time 300 voicemail mailbox owner "Enr
size 300 description "**Enrique_Mailbox**" expiration time 10 messagesize 120 end mailbox voicemail mai
owner "John" size 300 description "**John'sMailbox**" expiration time 10 messagesize 120 end mailbox en
```

# Verify

当前没有可用于此配置的验证过程。

# Troubleshoot

本部分提供的信息可用于对配置进行故障排除。

思科 CLI 分析器（仅适用于注册客户）支持某些 **show** 命令。请使用Cisco CLI分析器查看show命令输出分析。

**Note:** 使用 **debug** 命令之前，请参阅有关 Debug 命令的重要信息。

## 排除注册故障

排除在CME的SIP Trunk故障介入您使用排除故障IOS的SIP GW和CME排除故障的同样命令。请使用这些命令为了检查您的DN是否注册：

- **show sip-ua寄存器状态-**请使用此命令显示的E.164编号的状况SIP网关注册与一位外部主要的SIP管理员。
- **调试ccsip消息-** Enable (event)所有SIP SPI消息追踪，例如被交换在SIP用户代理客户端的那些(UAC)和接入服务器之间。

## 排除呼叫建立故障

排除的呼叫故障命令在SIP Trunk根本是作为您使用正常SIP GW和CME排除故障的相同的。

**显示**命令：

- **显示注册的ephone -**验证ephone注册。
- **表示voip RTP连接-**显示关于RTP已命名事件信息包的信息，例如caller-id编号、IP地址和端口本地和远程终点的。
- **show sip-ua呼叫-**显示活动UAC，并且用户代理关于SIP的服务器(UAS)信息呼叫。
- **show call active voice brief -**显示的语音呼叫或进展中的传真传输的激活呼叫信息。

调试指令：

- **调试ccsip消息-** Enable (event)所有SIP SPI消息追踪，例如被交换在SIP UAC和接入服务器之间的那些。
- **debug voip ccapi inout -**通过呼叫控制API跟踪执行路径。
- **调试语音转换-**检查转换规则的功能。
- **调试ephone详细资料*phone*> MAC地址<*mac* -**集选派Cisco IP电话的调试。
- **debug voip rtp会话已命名事件-**调试为实时传输协议(RTP)已命名事件信息包的Enable (event)。
- **debug sccp消息-**显示SCCP消息的顺序。

# Related Information

- [Cisco Unified Communications Manager Express系统管理员指南]
- [Cisco Unity Express 2.3安装和升级指南]
- [管理和监控Cisco Unified CallManager Express系统]
- [语音技术支持]
- [语音和统一通信产品支持]
- [Cisco IP 电话故障排除]
- [Technical Support & Documentation - Cisco Systems]