Sprint SIP Trunking: Connecting Cisco Unified Communications Manager 7.1(2) via the Cisco Unified Border Element 1.2 using SIP

August 17, 2009

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Introduction

This document explains key concepts for Sprint SIP Trunking (SST) service. SST is a means for enterprise customer to leverage their investment in GMPLS while reducing LEC charges. Sprint SIP Trunking (SST) uses Session Initiation Protocol, which enables enterprises to migrated from legacy based telecommunications equipment to Unified Communications and leverage the flexibility and scalability of IP.

This document assumes the reader is knowledgeable with the terminology and configuration of CUCM Cisco Unified Communications Manager as well as CUBE Cisco Unified Border Element. CUBE is available either through Sprint Managed Network Services (MNS) or it can be a customer owned and operated unit. This document provides a high-level overview of the options available with SST. These options primarily include CODEC selection. Additionally it will detail the configuration steps required in CUCM to integrate with SST and enable proper functionality.

- This application note describes how to configure a Cisco Unified Communications Manager (CUCM) with a Cisco Unified Border Element (CUBE) for connectivity to Sprint’s SIP trunking service. The deployment model covered in this application note is CPE (CUCM 7.1.2 / CUBE) bi-directional calls between “PSTN” via Sprint Trunking Service (SST).

- Testing was performed in accordance to Sprint’s test plan and all features were verified. Key features verified are: Basic Call, DNIS translations, Codec Negotiation, Transfer Connect, Intra-site Transfers, Intra-site Conferencing, Fax using T.38 (G3 only)

- The Cisco Unified Border Element configuration detailed in this document is based on a lab environment with a simple dial-plan used to ensure proper interoperability between Sprint SIP network and Cisco Unified Communications. The configuration described in this document details the important commands to have enabled for interoperability to be successful; care must be taken, by the network administrator deploying CUBE in a production environment, to ensure these commands are set per each dial-peer required to interoperate to Sprint SIP Trunking service.

- The results and configuration examples in this application note are based on Cisco Unified Communications Manager 6.1 and Cisco Unified Border Element 1.2 testing. CUCM 7.1 with CUBE 1.2 are designed to meet the same results as the tested versions.
Network Topology

Figure 1. Basic Call Setup
System Components

Hardware Components

- Cisco IOS ISR running CUBE 1.2 (IOS image version 12.4(20)T3 or later)
  - Cisco Unified Border Element is an integrated Cisco IOS Software application that runs on various IOS platforms, follow the link for more details: [http://www.cisco.com/go/cube](http://www.cisco.com/go/cube)
- Cisco MCS 7800 Series server (Cisco Unified Communications Manager)
- Cisco IP Phones (The topology diagram shows 7960 and 7961, but any Cisco IP phone model can be used) Tested with both SCCP and SIP phone software.
- Cisco IOS Gateway (only needed if Fax, analog phones or TDM systems are to be interconnected). This component may be a H323, SIP or MGCP gateway, the protocol is optional and the choice is left up to the customer’s network design.

Software Requirements

- Cisco Unified CM 7.1.2.20000-2. Additional versions may be supported, check with your Sprint account team.
- CUBE version 1.2 (IOS version 12.4(20)T3 or later)
- Cisco GW IOS Release: 12.4 or later
Features

Features Supported

- Basic Call using G.729 or G.711ulaw
- Calling Party Number Presentation and Restriction
- Fax using T.38
- Incoming DNIS Translation and Routing
- CUBE: Configured for media flow-around
- CUCM SIP trunk configured for Delay-Offer, the initial SIP INVITE from CUCM will be without SDP. The SIP trunk is configured without an MTP.
- RPID and PAI, PAI is only supported on CUCM 7.x and higher

Features Not Supported

- MOH and TOH (Tone On Hold) will not be heard on calls that terminate on a PSTN phone when it is put on Hold or the call is Xfer.
- Fax Passthru

Caveats

- When using G.729 between Sprint and Cisco Unified Border Element/Cisco Unified Communications Manager SIP trunk it is required to configure a Conference Bridge (CFB) resource on CUBE in order for Cisco Unified Communications Manager IP phone to initiate a three-way conference between G729 media end-points. See configuration section for details.
- For DTMF digit passing using RFC2833 you must set a payload-type value of 101 for "nte" (network telephone-events DTMF) on CUBE dial-peer pointing towards Cisco Unified Communications Manager. See configuration section for details. 101 is the default value for much of Cisco equipment including CUBE, IOS Voice Gateways and CUCM.
- Cisco Unified Communications Manager will only handle one codec type per SIP trunk. Even though the SST SIP INVITE proposes a list of codecs (e.g. G729, G711) the UCM SIP trunk will only allow the codec value it has been configured for. See configuration section for details.
Configuration

Configuring Cisco Unified Border Element (CUBE)

Critical commands are marked bold with footnote and description at bottom of the page

version 12.4
service tcp-keepalives-in
service tcp-keepalives-out
service timestamps debug datetime msec localtime
service timestamps log datetime msec localtime
service password-encryption
!
hostname cube
!
boot-start-marker
boot system flash flash:c3825-adventerprisek9 IvS-mz.124-20.T3.bin
boot-end-marker
!
logging message-counter syslog
logging queue-limit trap 10000
logging buffered 2000000
logging rate-limit 1000
no logging console
enable secret cisco
!
aaa new-model
!
!
aaa authentication login default group tacacs+ enable
!
!
aaa session-id common
clock timezone EST -5
clock summer-time EDT recurring
!
ip source-route
ip cef
!
!
!
!
no ip domain lookup
ip domain name smsl.mrn
no ipv6 cef
!
multilink bundle-name authenticated
!
!
!
voice-card 0
dspfarm
dsp services dspfarm
!
!
!
voice service voip
allow-connections sip to sip
no supplementary-service h450.2
no supplementary-service h450.3
supplementary-service h450.12
fax protocol 138 is-redundancy 0 hs-redundancy 0 fallback none
sip
asserted-id pai
privacy pstn
midcall-signaling passthru
!
!
voice class codec 1
codec preference 1 g729a
!
voice class codec 2
codec preference 2 g711ulaw
!
voice class sip-profiles 100
request INVITE sip-header From modify "((.*)@(.*)\>)" "\1sprint\2"
request REINVITE sip-header From modify "((.*)@(.*)\>)" "\1sprint\2"
request INVITE sip-header P-Asserted-Identity modify "((.*)@(.*)\>)" "\1sprint\2"
request REINVITE sip-header P-Asserted-Identity modify "((.*)@(.*)\>)" "\1sprint\2"
request INVITE sip-header Diversion modify "((.*)@(.*)\>)" "\1sprint\2"
request REINVITE sip-header Diversion modify "((.*)@(.*)\>)" "\1sprint\2"
!
!
voice translation-rule 10
rule 1 /.*\((.*\d\d\d\d)/ \1/
!
!
voice translation-rule 778
rule 1 /^778/ //
!
!
voice translation-profile Last10
translate called 10
!
voice translation-profile Strip-778
translate called 778
!
!
archive
log config
hidekeys
!
!
ip telnet source-interface GigabitEthernet0/0
no ip ftp passive
ip ftp source-interface GigabitEthernet0/0
ip tftp source-interface GigabitEthernet0/0
ip ssh version 2
!
!
interface GigabitEthernet0/0
description LAN Interface
ip address x.x.x.x 255.255.255.224
load-interval 30
duplex full
speed 1000
media-type sfp
negotiation auto
!
interface GigabitEthernet0/1
description Not Used
no ip address
shutdown
duplex auto
speed auto
media-type rj45
!
!
ip forward-protocol nd
ip route 0.0.0.0 0.0.0.0 x.x.x.1
no ip http server
no ip http secure-server
!
!
ip tacacs source-interface GigabitEthernet0/0
!
logging trap debugging
logging source-interface GigabitEthernet0/0
access-list 98 remark ACL controlling SNMP access
access-list 98 permit x.x.x.x 0.0.0.255
access-list 98 permit x.x.x.x 0.0.0.255
access-list 99 remark ACL controlling VTY access
access-list 99 permit x.x.x.x 0.0.0.255
access-list 99 permit x.x.x.x 0.0.0.255
access-list 99 permit x.x.x.x 0.0.3.255
access-list 99 permit 10.101.0.0 0.0.255.255
access-list 99 permit 10.77.124.0 0.0.3.255
snmp-server community snmp-comm-string RW 98
snmp-server ifindex persist
snmp-server trap-source GigabitEthernet0/0
snmp-server location Rm 2A711, bldg Ops1, Reston VA
snmp-server enable traps snmp authentication linkdown linkup coldstart warmstart
snmp-server enable traps vrrp
snmp-server enable traps gateway
snmp-server enable traps eigrp
snmp-server enable traps xgcp
snmp-server enable traps flash-insertion
snmp-server enable traps flash-removal
snmp-server enable traps envmon
snmp-server enable traps icxdsu
snmp-server enable traps isdn call-information
snmp-server enable traps isdn layer2
snmp-server enable traps isdn chan-not-avail
snmp-server enable traps isdn ietf
snmp-server enable traps ds0-busyout
snmp-server enable traps ds1-loopback
snmp-server enable traps ethernet cfm cc mep-up mep-down cross-connect loop config
snmp-server enable traps ethernet cfm crosscheck mep-missing mep-unknown service-up
snmp-server enable traps license
snmp-server enable traps aaa_server
snmp-server enable traps atm subif
snmp-server enable traps bgp
snmp-server enable traps cef resource-failure peer-state-change peer-fib-state-change inconsistency
snmp-server enable traps memory bufferpeak
snmp-server enable traps cnpd
snmp-server enable traps config-copy
snmp-server enable traps config
snmp-server enable traps config-ctid
snmp-server enable traps dial
snmp-server enable traps dsp card-status
snmp-server enable traps dsp oper-state
snmp-server enable traps entity
snmp-server enable traps fru-ctrl
snmp-server enable traps resource-policy
snmp-server enable traps frame-relay multilink bundle-mismatch
snmp-server enable traps frame-relay
snmp-server enable traps frame-relay subif
snmp-server enable traps hsrp
snmp-server enable traps ipmulticast
snmp-server enable traps msdp
snmp-server enable traps mvpn
snmp-server enable traps ospf state-change
snmp-server enable traps ospf errors
snmp-server enable traps ospf lsa
snmp-server enable traps ospf cisco-specific state-change nssa-trans-change
snmp-server enable traps ospf cisco-specific state-change shamlink interface-old
snmp-server enable traps ospf cisco-specific state-change shamlink neighbor
snmp-server enable traps ospf cisco-specific errors
snmp-server enable traps ospf cisco-specific retransmit
snmp-server enable traps ospf cisco-specific lsa
snmp-server enable traps pim invalid-pim-message
snmp-server enable traps pppoe
snmp-server enable traps cpu threshold
snmp-server enable traps rsvp
snmp-server enable traps ipslia
snmp-server enable traps syslog
snmp-server enable traps l2tun session
snmp-server enable traps l2tun pseudowire status
snmp-server enable traps vtp
snmp-server enable traps event-manager
snmp-server enable traps ccme
snmp-server enable traps srst
snmp-server enable traps voice
snmp-server enable traps dnsis
snmp-server host x.x.x.x mnstraps
snmp-server host x.x.x.x mnstraps
snmp-server host x.x.x.x voyence config
!
!
tacacs-server host x.x.x.x
tacacs-server host x.x.x.x
!
control-plane
!
call threshold global total-calls low 1 high 50
!
!
sccp local GigabitEthernet0/0
sccp ccm x.x.x.11 identifier 11 priority 2 version 6.0
sccp ccm x.x.x.12 identifier 12 priority 1 version 6.0

sccp ccm group 7
description transcoding resource for CUCM 7x
bind interface GigabitEthernet0/0
associate ccm 12 priority 1
associate ccm 11 priority 2
associate profile 11 register rl-cube-xcoder
associate profile 111 register res-lab-cubemtp

dsdpf farmhouse profile 11 transcode
description transcoding profile for CUCM 7.x
codec g711ulaw
codec g711alaw
codec g729ar8
codec g729abr8
codec g729r8
maximum sessions 8
associate application SCCP

ndsdpf farmhouse profile 111 mtp
codec g711ulaw
maximum sessions software 8
associate application SCCP


dial-peer voice 101 voip
description SST outbound DP to CUCM (pairs with 104)
translation-profile outgoing Last10
preference 1
destination-pattern ^258[2-9][2-9].......
media flow-around
voice-class codec 1
voice-class sip asserted-id pai
session protocol sipv2
session target ipv4:x.x.x.12
session transport udp
dtmf-relay rtp-nte
no vad


dial-peer voice 102 voip
description SST outbound DP to CUCM (pairs with 104)
translation-profile outgoing Last10
preference 2
destination-pattern ^258[2-9][2-9].......
media flow-around
voice-class codec 1
voice-class sip asserted-id pai
session protocol sipv2
session target ipv4:x.x.x.11
dtmf-relay rtp-nte
no vad
! dial-peer voice 103 voip
description SST outbound DP to CUCM (pairs with 104)
translation-profile outgoing Last10
preference 2
destination-pattern ^258[2-9]..[2-9]......$
media flow-around
voice-class codec 1
voice-class sip asserted-id pai
session protocol sipv2
session target ipv4:x.x.x.35
dtmf-relay rtp-nte
no vad
!
dial-peer voice 104 voip
description SST inbound DP for 258 + 10 Digit Dialing (pairs with 101,102,103)
media flow-around
voice-class codec 2
session protocol sipv2
incoming called-number ^258[2-9]..[2-9]......$
dtmf-relay rtp-nte
no vad
!
dial-peer voice 201 voip
description SST outbound DP (pairs with 204)
translation-profile outgoing Strip-778
preference 1
destination-pattern ^778
media flow-around
voice-class codec 2
voice-class sip asserted-id pai
voice-class sip profiles 100
session protocol sipv2
session target ipv4:xxx.xxx.xxx.xxx
session transport udp
dtmf-relay rtp-nte
no vad
!
dial-peer voice 202 voip
description SST outbound DP(pairs with 204)
translation-profile outgoing Strip-778
preference 2
destination-pattern ^778
media flow-around
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip profiles 764
session protocol sipv2
session target ipv4:xxx.xxx.xxx.xxx
dtmf-relay rtp-nte
no vad
!
dial-peer voice 204 voip
description SST inbound DP from CUCM (pairs with 201,202)
media flow-around
voice-class codec 1
session protocol sipv2
incoming called-number ^778
dtmf-relay rtp-nte
no vad

!  
!  
!
gateway
timer receive-rtp 600
!
sip-ua
no remote-party-id
retry invite 2^9
reason-header override
!
!
!
gatekeeper
shutdown
!
!
call-manager-fallback
  max-conferences 4 gain -6
  transfer-system full-consult
!

.banner motd □
**************************************************************************
WARNING TO UNAUTHORIZED USERS: This system is for the use of authorized users only. Individuals using this computer system without authority, or in excess of their authority, are subject to having all of their activities on this system monitored and recorded by system personnel. In the course of monitoring individuals improperly using this system, or in the course of system maintenance, the activities of authorized users may be monitored. Anyone using this system expressly consents to such monitoring and is advised that if such monitoring reveals possible evidence of criminal activity, system personnel may provide evidence of such monitoring to law enforcement officials.
**************************************************************************

 alias exec srb show run | begin
 alias exec sri show run | include
 alias exec sli show logging | include
 alias exec slb show logging | begin
 alias exec rate show interface g0/0 | i rate
 alias exec caip show call active voice brief | include ^ IP
 alias exec chip show call history voice brief | include ^ IP
 alias exec chpid show call history voice brief | include pid
 alias exec capid show call active voice brief | include pid
 alias exec ch show call history voice brief
 alias exec cl clear logging
 alias exec ca show call active voice brief | begin ^Telephony
 alias exec dsp show dsfarm dsp active
!
 line con 0
 exec-timeout 0 0
 line aux 0
exec-timeout 11 0
line vty 0 4
access-class 99 in
eexec-timeout 0 0
transport input ssh
transport output ssh
!
exception data-corruption buffer truncate
scheduler allocate 20000 1000
ntp source GigabitEthernet0/0
ntp server x.x.x.x prefer
end

1 Specifies T.38 Fax Protocol, Sprint Trunking Service only support T.38
2 Supports PAI Asserted Identify to Sprint SIP Provider
3 Required for PAI
4 Specifies G.729 CODEC – Required so subsequent reINVITEs following a hold or xfer reference the proper CODEC in the SIP SDPs.
5 Specifies G.711 CODEC – Required so subsequent reINVITEs following a hold or xfer reference the proper CODEC in the SIP SDPs.
6 Required for PAI support
7 Media Flow Around configured locally on the dial-peers. Can also be configured globally under voice service voip – sip – media flow around, do not configure both locally and globally.
8 Assigned the SIP header profile locally on the dial-peer.
9 Limits the number of INVITEs sent to a dial-peer before failing over to another dial-peer if no response. Setting to 2 limits failover to approx 3 seconds, the default is 6. This make failover occur much more slowly, on the order of 30 seconds.

Sprint SIP Trunking Basic Call Flow:

With Sprint SIP Trunking (SST) the enterprises DID range will now be “pointed” to the SIP data center(s) instead of the legacy LEC trunks. Calls into the enterprise will then traverse the GMPLS circuit via the SIP trunk. Calls from IP desk phones will like wise use the SIP Trunk and GMPLS circuit to reach PSTN destinations. The Cisco Unified Communications Manager section details the configurations steps that will be required to create the SIP trunk and redirect calls via the SIP trunk.

Cisco Unified Border Element:

Cisco’s Unified Border Element is integrated with CUCM via SIP. It also provides a point of demarcation between the Enterprises voice network and Sprints. Cisco Unified Border Element can be provided and managed by Sprint Managed Network Services or be purchase and managed by the customer. For large enterprises that may require additional CUBE’s for load balancing, Sprint Custom Solutions Group will be engaged. Customers may also provide their own CUBE(s) in addition to Sprint CUBE if they require additional call capacity or redundancy.

CODEC Options:

Sprint SIP Trunking (SST) support both G.711ulaw and G.729r8 CODEC’s. For optimum voice quality Sprint suggest that G.711ulaw be used for all calls. For maximum bandwidth savings G.729r8 should be used. Additionally many enterprises require a mix of CODEC supported. For example a customer support number would typically use G.711 but for branch offices over a WAN link G.729 would be used. SST can be configured to support both simultaneously. Talk with your Sprint account team to determine what is best for your environment.

As the type of CODEC used has a direct correlation to the bandwidth available on the GMPLS WAN circuit. Talk to your Sprint account team for proper sizing of your circuit needs based on your call volume, dial plan and CODEC selection.

Before Making any Changes on your production CUCM

BEFORE IMPLEMENTING ANY OF THE SST CHANGES ON A PRODUCTION CISCO UNIFIED COMMUNICATIONS MANAGER SYSTEM IT IS HIGHLY SUGGESTED THAT A SEPARATE NON-PRODUCTION “PILOT” CLUSTER BE CONFIGURED USING THE EXACT CONFIGURATION OF THE INTENDED PRODUCTION SYSTEM. IMPLEMENT THE SST CHANGES ON THE PILOT SYSTEM FIRST TO UNDERSTAND ANY IMPACT THE SST CHANGES MAY HAVE. BECAUSE OF THE NUMEROUS WAYS IN WHICH A CUCM SYSTEM CAN BE CONFIGURED IT IS IMPOSSIBLE TO FORSEE ALL IMPACT THAT THE SST CHANGES MAY
HAVE. REFER TO THE APPROPRIATE CISCO DOCUMENTATION FOR YOUR VERSION OF CUCM. BEFORE IMPLEMENTING THE SST CHANGES ON A PRODUCTION CUCM CLUSTER MAKE A FULL BACK OF THE CURRENT CUCM CONFIGURATION.
CUCM Configuration Steps For Implementing SST

When configuring CUCM for SST there are a number of configuration steps required in order for SST to properly function. This section will detail the various steps required to configure you CUCM to interoperate properly with SST.

Communications Manager Software Version

Sprint SIP Trunking (SST) has been certified to work with CUCM version 7.1(2) and higher. Contact your Sprint account team to determine if your version is supported.

Communications Manager Service Parameters

For SST to interoperate with CUCM all Service Parameters are left as default, except for the below listed:

<table>
<thead>
<tr>
<th>Clusterwide Parameters (Device – SIP)</th>
<th></th>
</tr>
</thead>
<tbody>
<tr>
<td>Retry Count for SIP Invite</td>
<td></td>
</tr>
<tr>
<td>This determines the number of INVITE CUCM sends out before trying another route. With the default setting of “6” CUCM can take up to 30 seconds to try another route. Changing the value to “2” allows for the failover to another route to occur much more quickly.</td>
<td></td>
</tr>
<tr>
<td>Default</td>
<td>6</td>
</tr>
<tr>
<td>Configured Value</td>
<td>2</td>
</tr>
</tbody>
</table>

1. Service Parameter Changes
   - System -> Service Parameter -> Server -> Cisco CallManager -> Clusterwide Parameters (Device – SIP) Change Retry Count for SIP Invite - 2

Configuring the SIP Trunk

The SIP trunk between Cisco Unified Communications Manager and the Sprint Managed Cisco Unified Border Element is required to enable SST to function

2. Configure a SIP Profile
   - Device -> Device Settings -> SIP Profile
   - Copy the existing Standard SIP Profile
   - Name the New SIP Profile – SST SIP Profile or equivalent.
   - There are no changes made to the setting between the original Standard SIP Profile and the SST SIP Profile. It is best practices to make copies of individual components when configuring the SIP trunk.
   - Save the configuration.

3. Configure a SIP Trunk Security Profile
   - System -> Security Profile -> SIP Trunk Security Profile
   - Copy the existing Non Secure SIP Trunk Profile
   - Name the New SIP Trunk Profile – SST Non Secure SIP Trunk Profile or equivalent.
   - Change the Outgoing Transport Type to be only UDP
   - Make sure all other Check Boxes are unchecked except for “Accept Out-of-Dialog REFER.”
   - Save the configuration.

4. Configure a Region(s)
System -> Region -> Add New
If your creating a G.711 Only SST Implementation name it “SST-SIP-Trunk-Region-G.711” or equivalent
If you’re creating a G.729 Only SST Implementation name it “SST-SIP-Trunk-Region-G.729” or equivalent
If you’ll be doing a mix of both, then you will need to create both Regions.
For a G.711 region configure G.711 for use between your existing Regions.
For a G.729 region configure G.729 for use between your existing Regions.
If you will be implementing both then ensure that in addition to the Regions that you have sufficient transcoder resources to allow calls between both regions. Refer to the Cisco documentation for configuring and adding transcoders to your CUCM environment.
Save the configuration.

5. Configure a Device Pool(s)
System -> Device Pool -> Add New
For a G.711 only SST implementation make a G.711 Device Pool. Name it SST-Device-Pool-G.711 or equivalent.
For a G.729 only SST implementation make a G.729 Device Pool. Name it SST-Device-Pool-G.729 or equivalent.
For a mixed SST implementation you will have to configure two Device Pools.
Assign the appropriate region to the Device Pool.
Other setting such as Date/Time Group, Media Resource Group List etc… will have to be configured according to your CUCM environment.
Note that 0 members will initially be assigned to the Device Pool until you individually or via Bulk Administration add the IP phones to the Device Pool
Save the configuration.

6. Configure the SIP Trunk
Device -> Trunk – Add New
For Trunk Type Select SIP Trunk
The Device Protocol will automatically be populated with SIP, click Next
For a G.711 only SST implementation make a G.711 SIP Trunk, name it SST-SIP-Trunk-G.711 or equivalent.
For a G.729 only SST implementation make a G.729 SIP Trunk, name it SST-SIP-Trunk-G.729 or equivalent.
For a mixed SST implementation, you will have to configure two SIP Trunks.
Add the appropriate Device Pool that you previously configured in Step 5 above.
The following steps are the same for either CODEC selected.
Check the check box for “Use Device Pool Called Party Transformation CSS”
Check the check box for “Use Device Pool Calling Party Transformation CSS”
Check the check box “Redirecting Diversion Header Delivery – Outbound”.
Destination IP Address will be the Sprint Managed CUBE. Get this from your Sprint Account Team.
Add the SIP Trunking Security Profile previously configured in Step 3 above.
Add the SIP Profile previously configured in Step 2 above.
Save the configuration.

7. Configure a Route Group(s)
Call Routing -> Route/Hunt -> Route Group
Add New
For a G.711 Only SST implementation make a G.711 Route Group, name it SST-RG-G.711 or equivalent.
For a G.729 Only SST implementation make a G.729 Route Group, name it SST-RG-G.729 or equivalent.
For a mixed SST implementation, you will have to configure two Route Groups.
From the Available Devices select the appropriate SIP trunk previously configured in Step 6 above.
Save the configuration.

8. Configure a Route List(s)
Call Routing -> Route/Hunt -> Route List
Add New
For a G.711 Only SST implementation make a G.711 Route List, name it SST-RL-G.711 or equivalent.
For a G.729 Only SST implementation make a G.729 Route List, name it SST-RL-G.729 or equivalent.
Select the appropriate Cisco Unified Communications Manager Group
9. Configure Route Patterns
- Due to the complexity and individual needs of every enterprise, configuring any enterprise dial plan is subject to a number of variables.
- Please refer to the appropriate Cisco documentation for configuring Route Patterns.
- Each Route Pattern will need to point to its perspective Route List. You may need to have multiple route patterns for different dialing restrictions.
- If you are implementing a mix SST deployment you may have to have multiple duplicate Route Patterns in separate partitions point to different Route List depending on the CODEC being used.
- Engage your Sprint and Cisco account teams for additional information.

10. If you are going to implement SST with a single CODEC than existing partitions and Calling Search spaces are probably sufficient. Reconfiguring the Route Patterns / Route Groups to point to the proper SIP trunk maybe all that is required. If you will be implementing multiple CODEC’s for SST then you may have to duplicate existing partitions and Calling Search spaces. The below table is a typical dial plan that includes an Internal Partition for all IP phones, a Voicemail Partition and local, long distance and international partitions and Calling Search spaces. In order to make it easier to understand and trouble shoot we have named each partition and calling search space with the particular CODEC that the device originating or terminating the call will use. In actual practice you might be able to use your existing partitions and calling search space for one of the CODEC’s and simply copy those to support the other CODEC.
### Calling Search Space

<table>
<thead>
<tr>
<th>Calling Search Space</th>
<th>Partition (s)</th>
<th>Use</th>
</tr>
</thead>
<tbody>
<tr>
<td>SST-Internal-CSS-G711</td>
<td>SST-Internal-PT-G711, SST-Internal-PT-G729, SST-Voicemail-PT-G711</td>
<td>Allows IP phones in G711 region to talk to IP phones in the G729 region.</td>
</tr>
<tr>
<td>SST-Internal-CSS-G729</td>
<td>SST-Internal-PT-G729, SST-Internal-PT-G711, SST-Voicemail-PT-G729</td>
<td>Allows IP phones in G711 region to talk to IP phones in the G729 region.</td>
</tr>
<tr>
<td>SST-Local-CSS-G711</td>
<td>SST-Internal-PT-G711, SST-Internal-PT-G729, SST-Voicemail-PT-G711, SST-Local-PT-G711</td>
<td>Access for G711 region phones to Local Numbers</td>
</tr>
<tr>
<td>SST-Local-CSS-G729</td>
<td>SST-Internal-PT-G729, SST-Internal-PT-G711, SST-Voicemail-PT-G729, SST-Local-PT-G711</td>
<td>Access for G729 region phones to Local Numbers</td>
</tr>
<tr>
<td>SST-LD-CSS-G711</td>
<td>SST-Internal-PT-G711, SST-Internal-PT-G729, SST-Voicemail-PT-G711, SST-LD-PT-G711</td>
<td>Access for G711 region phones to Local and LD Numbers</td>
</tr>
<tr>
<td>SST-LD-CSS-G729</td>
<td>SST-Internal-PT-G729, SST-Internal-PT-G711, SST-Voicemail-PT-G729, SST-LD-PT-G711</td>
<td>Access for G729 region phones to Local and LD Numbers</td>
</tr>
<tr>
<td>SST-Intl-CSS-G711</td>
<td>SST-Internal-PT-G711, SST-Internal-PT-G729, SST-Voicemail-PT-G711, SST-LD-PT-G711, SST-Intl-PT-G711</td>
<td>Access for G711 region phones to Local, LD and International Numbers</td>
</tr>
<tr>
<td>SST-Intl-CSS-G729</td>
<td>SST-Internal-PT-G729, SST-Internal-PT-G711, SST-Voicemail-PT-G729, SST-LD-PT-G711, SST-Intl-PT-G729</td>
<td>Access for G729 region phones to Local, LD and International Numbers</td>
</tr>
</tbody>
</table>

Your dial-plan will probably be different.

**11. Adding IP Phones to Device Pools**

- Device -> Phone
- Select a phone from the list or Add New
- In the Device Pool drop down list select the appropriate Device Pool previously configured in Step 5 above.
- Save the configuration.

**Screen Shots Showing CUCM Configuration Changes**

**Service Parameter Configuration Screens**

System -> Service Parameters -> Server -> Cisco CallManager
Scroll down to the “Clusterwide Parameters (Device – SIP)” Section
<table>
<thead>
<tr>
<th>Parameter Description</th>
<th>Value 1</th>
<th>Value 2</th>
</tr>
</thead>
<tbody>
<tr>
<td>Example Parameter 1</td>
<td>Value 1</td>
<td>Value 2</td>
</tr>
<tr>
<td>Example Parameter 2</td>
<td>Value 1</td>
<td>Value 2</td>
</tr>
<tr>
<td>...</td>
<td>...</td>
<td>...</td>
</tr>
</tbody>
</table>

Note: The parameters mentioned in the table are just placeholders and do not represent real parameter names or values. The table structure and the way parameters are grouped together is not a standard format and might not correspond to the actual parameterization of any real-world software or application.
SIP Profile Configuration Screens

Device -> Device Settings -> SIP Profile (No Changes from Default)

Second Screen
### User Profile Configuration - Microsoft Internet Explorer

#### Cisco Unified CM Administration

For Cisco Unified Communications Solutions

**System → Call Routing → Media Resources → VoiceMail → Device → Application → User Management → Bulk Administration → Help →**

![Cisco Unified CM Configuration](https://example.com/cisco-unified-cm-config.png)

<table>
<thead>
<tr>
<th>Feature</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Call Pickup Group URI</td>
<td>cisco-service-upgr1</td>
</tr>
<tr>
<td>Meet Me Service URI</td>
<td>cisco-service-meetme</td>
</tr>
<tr>
<td>User Info</td>
<td>None</td>
</tr>
<tr>
<td>DNWFD On</td>
<td>Off</td>
</tr>
<tr>
<td>Call Hold Ring Bck</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 Ctrl</td>
<td>User</td>
</tr>
<tr>
<td>Ticket Level for 7440 and 7442</td>
<td>Disabled</td>
</tr>
<tr>
<td>Timer Answer Key Press (seconds)</td>
<td>30</td>
</tr>
<tr>
<td>Timer Subscribe Expire (seconds)</td>
<td>60</td>
</tr>
<tr>
<td>Maximum Redials</td>
<td>70</td>
</tr>
<tr>
<td>Off Hook-To Initial Digit Timer (milliseconds)</td>
<td>5000</td>
</tr>
<tr>
<td>Call Forward List</td>
<td>cisco-service-upgr1</td>
</tr>
<tr>
<td>Abbreviated Dial URI</td>
<td>cisco-service-upgr1</td>
</tr>
</tbody>
</table>

- **Conference Join Enabled**
- **RFC 3461 Hold**
- **Semi-Attended Transfer**
- **Enable VAD**
- **Stutter Message Waiting**
- **Call Stats**

---

**Trunk Specific Configuration**

- Resurse incoming request to new Trunk based on **Never**

### Saving Options

- Save
- Delete
- Copy
- Reset
- Apply Config
- Add New
SIP Trunk Security Profile Configuration Screens

System -> Security Profile -> SIP Trunk Security Profile

![SIP Trunk Security Profile Configuration Screen]

- **SIP Trunk Security Profile Information**
  - Name: SIP Trunk Secure SIP Trunk Profile
  - Direction: Bidirectional
  - Device Security Mode: None Secure
  - Masking Explicit URI: No
  - Transport Type: CDC
  - Security Protocol: TLS
  - Message Digest Algorithm: MD5

- **Required Security Settings**
  - Accept Out-of-Dialog REMEX
  - Accept Distant Authentication
  - Accept Restricted Registration
  - Transmit Security Status

- **Additional Information**
  - Indicates required items.
Region Configuration Screens For Both G.711 and G.729 Regions

System -> Region (G.711 Region)

Region Configuration - Microsoft Internet Explorer

Region Relationships

<table>
<thead>
<tr>
<th>Region</th>
<th>Audio Codec</th>
<th>Voice Call-Bandwidth</th>
<th>ISDN Type</th>
</tr>
</thead>
<tbody>
<tr>
<td>G.711</td>
<td></td>
<td>0.64</td>
<td>Use System Default</td>
</tr>
</tbody>
</table>

Note: Region(s) not displayed
Use System Default Use System Default Use System Default

Modify Relationship to other Regions

[Icons for modifying relationships to other regions]

* Indicates required item.
**The Audio Codec selection determines bandwidth units. The G.711 and G.729 codecs both result in a maximum bandwidth of 64 kbps between regions and can be used interchangeably.
System -> Region (G.729 Region)

Device Pool Configuration Screens
System -> Device Pool -> (G.711 Device Pool)
System -&gt; Device Pool -&gt; (G.729 Device Pool)
### Service Pool Information
- **Device Pool:** SST-Device-Pool-0.7.09 (1 members*)

### Service Pool Settings
- **Device Pool Name:** SST-Device-Pool-0.7.09
- **Cisco Unified Communications Manager Group:** Default
- **Cisco Unified Communications Manager Zone:** None
- **Cisco Unified Communications Manager Backend:** None
- **Local Route Group:** Default

### Service Pool Related Information
- **Device Mobility Group:** None
- **Device Mobility Group:** None
- **Device Mobility Group:** None

### Important notes
- **Device Mobility Group:** None
- **Device Mobility Group:** None
- **Device Mobility Group:** None
SIP Trunk Configuration Screens - 1 of 3

Device -> Trunk (G.711 SIP Trunk)
Route Group Configuration Screens

Call Routing -> Route/Hunt -> Route Group (For G.711)

Call Routing -> Route/Hunt -> Route Group (For G.729)
Route List Configuration Screens

Call Routing -> Route/Hunt -> Route List (G.711)

Call Routing -> Route/Hunt -> Route List (G.729)
Adding IP Phones to Device Pools

Device -> Phone (G.711)

Device -> Phone (G.729)
## Acronyms

<table>
<thead>
<tr>
<th>Acronym</th>
<th>Definitions</th>
</tr>
</thead>
<tbody>
<tr>
<td>SIP</td>
<td>Session Initiation Protocol</td>
</tr>
<tr>
<td>MGCP</td>
<td>Media Gateway Control Protocol</td>
</tr>
<tr>
<td>SCCP</td>
<td>Skinny Client Control Protocol</td>
</tr>
<tr>
<td>CUCM</td>
<td>Cisco Unified Communications Manager</td>
</tr>
<tr>
<td>CUBE</td>
<td>Cisco Unified Border Element</td>
</tr>
</tbody>
</table>
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<td>BV</td>
<td>170 West Tasman Drive</td>
<td>Capital Tower</td>
</tr>
<tr>
<td>San Jose, CA 95134-1706</td>
<td>Haarlerbergpark</td>
<td>San Jose, CA 95134-1706</td>
<td>168 Robinson Road</td>
</tr>
<tr>
<td>USA</td>
<td>Haarlerbergweg 13-19</td>
<td>USA</td>
<td>#22-01 to #29-01</td>
</tr>
<tr>
<td></td>
<td>1101 CH Amsterdam</td>
<td><a href="http://www.cisco.com">www.cisco.com</a></td>
<td>Singapore 068912</td>
</tr>
<tr>
<td></td>
<td>The Netherlands</td>
<td>Tel: 408 526-7660</td>
<td><a href="http://www.cisco.com">www.cisco.com</a></td>
</tr>
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<td></td>
<td>www-europe.cisco.com</td>
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<td></td>
<td>Tel: 31 0 20 357 1000</td>
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<td>Fax: +65 317 7799</td>
</tr>
<tr>
<td></td>
<td>Fax: 31 0 20 357 1100</td>
<td></td>
<td></td>
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