
March 04, 2016
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Introduction

Service Providers today, such as Verizon, 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.

Verizon Business SIP Trunk 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 Microsoft Skype for Business Server 2015 and Verizon network, Cisco Unified Border Element (Cisco UBE 11.5.0) 15.6.1.S [IOS-XE 3.17] can be used. The Cisco Unified Border Element (Cisco UBE) 15.6.1.S provides demarcation, security and inter-working and session control services for Microsoft Skype for Business Server 2015 connected to Verizon IP network.

This document assumes the reader is knowledgeable with the terminology and configuration of Microsoft Skype for Business Server 2015. Only configuration settings specifically required for Verizon 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 Microsoft Skype for Business Server 2015 and Cisco Unified Border Element (Cisco UBE 11.5.0) 15.6.1.S [IOS-XE 3.17] for connectivity to Verizon SIP Trunking service. The deployment model covered in this application note is Skype for Business Server 2015 to PSTN via Cisco Unified Border Element (Cisco UBE) 15.6.1.S.

- Testing was performed in accordance to Cisco 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 High Availability.

- The Cisco Unified Border Element (Cisco UBE) configuration detailed in this document is based on a lab environment with a simple dial-plan used to ensure proper interoperability between Verizon SIP network and Microsoft Skype for Business Server. 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 Microsoft Skype for Business Server to interoperate to Verizon SIP Trunking network.
**Network Topology**

The network topology includes the Microsoft Skype for Business Server Enterprise Edition and 2 Lync clients. Cisco UBE published as a PSTN gateway in the Skype for Business Server topology. Verizon was used as the service provider with a SIP trunk to the Cisco UBE.

- 2 Cisco Unified Border Elements are used here for High Availability.
- SIP Trunk transport type used between Cisco Unified Border Element and Microsoft Skype for Business is either TCP or TLS. Configuration for each trunk type is explained in the Dial-peer section of this document.

**IP-PBX - CUBE Trunk Settings:**

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Media Bypass</td>
<td>OFF</td>
</tr>
<tr>
<td>REFER Support</td>
<td>ON</td>
</tr>
<tr>
<td>Session Timer</td>
<td>ON</td>
</tr>
<tr>
<td>Early Media support with PRACK</td>
<td>ON</td>
</tr>
</tbody>
</table>
System Components

Hardware Requirements
- Cisco UBE on Cisco ISR 4431 router
- Generic Server for Skype for Business

Software Requirements
- Cisco UBE IOS CUBE-Version: 11.5.0, SW-Version: 15.6.1.S, XE- 3.17.00.S
- Microsoft Skype for Business Server 2015- Version: 6.0.9319.0
- Microsoft Skype for Business Client – Version 15.0.4797.1000

Features

Features Supported
- Incoming and outgoing off-net calls using G711
- International Calls and digit manipulations
- Call Conference
- CPE Voice Mail support
- Call hold & Resume with and without MoH
- Unattended and Attended Call transfer
- Call forward (all and no answer)
- DTMF (RFC2833)
- IP-PBX Calling number privacy
- High Availability
- Early Media support

Features Not Supported
- G729 is not Supported by Skype for Business
- Fax (G.711 and T.38)
- Blind Call transfer
- Call forward on Busy
Caveats

- Skype for Business does not support G729 so g711 is preferred.
- Trunk from Skype for Business to CUBE is tested with both TCP / RTP and TLS / SRTP.
- Ring back tone is not heard in off-net phone when the Skype user transfers the call.
- Caller ID updates are not observed on attended call transfer scenarios.
- Testing is done with only one IP PBX.
- Skype for Business requires 3rd party FAX for FAX support
- The Media Bypass is turned off in trunk between Skype for business and CUBE
- Workaround is done for SRTP negotiation between Skype for Business and CUBE
- Workaround is done for Diversion header manipulations for Call Forward and call Transfer
- RTCP Sender reports are not sent from CUBE to Skype unless ITSP sends it to CUBE.
- CUBE has an added dial-peer for Skype for Business REFER support to work
- SIP Global Binding should be done with the interface pointing to Skype for Business
Configuration

Configuring Cisco Unified Border Element

Network Interface
The IP address used are for illustration only, the actual IP address can vary. The Active/Standby pair share the same virtual IP address and continually exchange status messages.

Figure 2 High Availability topology
CUBE 1:

interface GigabitEthernet0/0/0
  ip address 10.64.4.19 255.255.0.0
  media-type rj45
  negotiation auto
!

interface GigabitEthernet0/0/1
  no ip address
  shutdown
  media-type rj45
  negotiation auto
!

interface GigabitEthernet0/0/2
  description LAN interface
  ip address 10.80.22.74 255.255.255.0
  media-type rj45
  negotiation auto
  redundancy rii 1
  redundancy group 1 ip 10.80.22.100 exclusive
!

interface GigabitEthernet0/0/3
  description WAN interface
  ip address 192.65.79.114 255.255.255.224
  media-type rj45
  negotiation auto
  redundancy rii 2
  redundancy group 1 ip 192.65.79.124 exclusive
!
CUBE 2:

! interface GigabitEthernet0/0/0
    ip address 10.64.4.20 255.255.0.0
    media-type rj45
    negotiation auto

! interface GigabitEthernet0/0/1
    no ip address
    shutdown
    media-type rj45
    negotiation auto

! interface GigabitEthernet0/0/2
    ip address 10.80.22.75 255.255.255.0
    media-type rj45
    negotiation auto
    redundancy rii 1
    redundancy group 1 ip 10.80.22.100 exclusive

! interface GigabitEthernet0/0/3
    description Wan Interface
    ip address 192.65.79.115 255.255.255.224
    media-type rj45
    negotiation auto
    redundancy rii 2
    redundancy group 1 ip 192.65.79.122 exclusive

!
Global Cisco UBE settings

In order to enable Cisco UBE IP2IP SBC functionality, following command has to be entered:

```
voice service voip
  no ip address trusted authenticate
  rtcp keepalive
  address-hiding
  mode border-element license capacity 20
  allow-connections sip to sip
  redundancy-group 1
  no supplementary-service sip moved-temporarily
  no supplementary-service sip refer
  supplementary-service media-renegotiate
  sip
    bind control source-interface GigabitEthernet0/0/2
    bind media source-interface GigabitEthernet0/0/2
    session refresh
    header-passing
    referto-passing
    conn-reuse
    sip-profiles inbound
    sip-profiles 248 inbound
```

**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>redundancy-group 1</td>
<td>Enable High Availability for the VoIP service</td>
</tr>
<tr>
<td>rtcp keepalive</td>
<td>Enables the CUBE to send rtcp keepalive packets for the session keepalive</td>
</tr>
<tr>
<td>Bind control and media</td>
<td>SIP Global binding is done with the interface pointing to Skype for Business to avoid TCP socket issue</td>
</tr>
</tbody>
</table>
Codecs

G711 is used as the preferred codec for this testing as G729 is not supported in Skype for Business.

**voice class codec 2**

codec preference 1 g711ulaw
codec preference 2 g729r8 bytes 30
codec preference 3 g726r32
codec preference 4 g729br8

SIP Profiles

SIP Profile 1 is used in the dial-peer pointing towards Verizon. Here the Diversion header is inserted by copying the uri from “REFERRED-BY” header and then truncating the preceding “+1”. SIP Profile 248 is used to invalidate the ‘a’ attribute of the SDP that has MKI. As the CUBE does not support crypto with MKI we have to apply this profile for the SRTP calls to work.

**voice class sip-profiles 1**

request INVITE peer-header sip REFERRED-BY copy "sip:(.*)@" u01

request INVITE sip-header Diversion add "Diversion: <sip:sip-uri@192.65.79.122>"

request INVITE sip-header Diversion modify "sip:(.*)@" "sip:\u01@"

request INVITE sip-header Diversion modify "sip:+1" "sip:

request REFER sip-header Referred-By remove

**voice class sip-profiles 248**

request ANY sdp-header Audio-Attribute modify
"a=(.*)AES_CM_128_HMAC_SHA1_(.*) inline(.*)\|(.*):(.*)" "a=\1CRYPTO_UNKNOWN inline\3\4:\5"

response ANY sdp-header Audio-Attribute modify
"a=(.*)AES_CM_128_HMAC_SHA1_(.*) inline(.*)\|(.*):(.*)" "a=\1CRYPTO_UNKNOWN inline\3\4:\5"
Dial peer

Dial-peer to Skype for Business for TCP with RTP:

! 

dial-peer voice 107 voip
  description "Incoming PSTN to IP-PBX - IP-PBX facing side"
  translation-profile outgoing e164
  destination-pattern 719....... 
  session protocol sipv2
  session target dns:lync1.sfblabis.local:5060
  session transport tcp
  incoming uri request FQDNsfb
  voice-class codec 2
  voice-class sip localhost dns:isr4k.sfblabis.local:5060
  voice-class sip call-route url
  voice-class sip options-keepalive
  voice-class sip bind control source-interface GigabitEthernet0/0/2
  dtmf-relay rtp-n-te
  refer consume
  no vad
!

dial-peer voice 105 voip
  description incoming from IP-PBX
  session protocol sipv2
  session transport tcp
  incoming called-number 921424259..
  voice-class codec 2
  voice-class sip localhost dns:isr4k.sfblabis.local:5060
  voice-class sip copy-list 1
  voice-class sip bind control source-interface GigabitEthernet0/0/2
voice-class sip bind media source-interface GigabitEthernet0/0/2
dtmf-relay rtp-nce
no vad
!
dial-peer voice 786 voip
description TEST for REFER
translation-profile outgoing e164
session protocol sipv2
session target sip-uri
session transport tcp
destination uri SFB
voice-class codec 2
voice-class sip localhost dns:isr4k.skypelabsk.local:5060
voice-class sip profiles 248
voice-class sip bind control source-interface GigabitEthernet0/0/2
voice-class sip bind media source-interface GigabitEthernet0/0/2
voice-class sip referto-passing
voice-class sip requiri-passing
dtmf-relay rtp-nce
no vad

Dial-peer to Skype for Business for TLS with SRTP:

dial-peer voice 107 voip
description "Incoming PSTN to IP-PBX - IP-PBX facing side"
translation-profile outgoing e164
destination-pattern 719....... 
session protocol sipv2
session target dns:fe.skypelabsk.local:5067
session transport tcp tls
incoming uri request FQDNsfb
voice-class codec 2
voice-class sip srtp-auth sha1-80 sha1-32
voice-class sip localhost dns:isr4k.skypelabsk.local:5061
voice-class sip asserted-id pai
voice-class sip call-route url
voice-class sip profiles 248
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/0/2
voice-class sip bind media source-interface GigabitEthernet0/0/2
voice-class sip refer-to-passing
dtmf-relay rtp-n-te
refer consume
rtcp keepalive
srtp
no vad
!
dial-peer voice 105 voip
description incoming from IP-PBX
session protocol sipv2
session transport tcp tls
incoming called-number 9..........incoming uri request SFB
voice-class codec 2
voice-class sip srtp-auth sha1-80 sha1-32
voice-class sip localhost dns:isr4k.skypelabsk.local:5061
no voice-class sip asserted-id
voice-class sip call-route url
no voice-class sip srtp negotiate
voice-class sip bind control source-interface GigabitEthernet0/0/2
voice-class sip bind media source-interface GigabitEthernet0/0/2
voice-class sip requiro-passing
dtmf-relay rtp-n-te
srtp
no vad
! 
dial-peer voice 786 voip
description TEST for REFER
translation-profile outgoing e164
session protocol sipv2
session target sip-uri
session transport tcp tls
destination uri SFB
voice-class codec 2
voice-class sip srtpt-auth sha1-80 sha1-32
voice-class sip localhost dns:isr4k.skypelabsk.local:5061
voice-class sip profiles 248
voice-class sip bind control source-interface GigabitEthernet0/0/2
voice-class sip bind media source-interface GigabitEthernet0/0/2
voice-class sip referto-passing
voice-class sip requri-passing
dtmf-relay rtp-n-te
srtp
no vad

Dial-peer to Verizon:

dial-peer voice 400 voip
description "Outgoing To PSTN"
translation-profile outgoing pstn
destination-pattern .T
rtp payload-type comfort-noise 13
session protocol sipv2
session target ipv4:63.87.147.30:5072
session transport udp
voice-class codec udp
voice-class sip rel1xx disable
voice-class sip privacy-policy passthru
voice-class sip profiles 1
voice-class sip session refresh
voice-class sip bind control source-interface GigabitEthernet0/0/3
dtmf-relay rtp-n te
rtcp keepalive
srtp fallback
no vad

! 
dial-peer voice 201 voip
description incoming from PSTN
translate-outgoing called 25
session protocol sipv2
session transport udp
incoming called-number 719....... 
voice-class codec 2 
voice-class sip rel1xx disable
voice-class sip profiles 2 inbound
voice-class sip session refresh
voice-class sip bind control source-interface GigabitEthernet0/0/3
dtmf-relay rtp-n te
srtp fallback
no vad
Configuration example

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

Active Cisco UBE:

CISCO_4K_ROUTER1#sh run

Building configuration...

Current configuration : 15142 bytes

! Last configuration change at 08:00:33 UTC Sat Feb 13 2016 by cisco

! version 15.6

service timestamps debug datetime msec localtime year
service timestamps log datetime msec localtime year
service sequence-numbers
no platform punt-keepalive disable-kernel-core

! hostname CISCO_4K_ROUTER1

! boot-start-marker

boot system flash bootflash:isr4400-universalk9.03.17.00.S.156-1.S-std.SPA.bin

boot-end-marker

!

vrf definition Mgmt-intf

!

address-family ipv4

exit-address-family
!  
address-family ipv6  
exit-address-family  
!  
vrf definition mgmt-intf  
!  
logging queue-limit 500000  
logging queue-limit trap 500000  
logging buffered 9999999  
logging rate-limit 10000  
no logging console  
no logging monitor  
enable secret 5 $1$EWC0$6QpPfaUZ/L04rhx3qQlj7/  
!  
aaa new-model  
!  
!  
aaa session-id common  
!  
ipc zone default  
 association 1  
 no shutdown  
!  
ip host dc.skypelabsk.local 10.64.3.230  
ip host fe.skypelabsk.local 10.64.3.231  
ip host isr4k.skypelabsk.local 10.80.22.74  

no ip domain lookup
subscriber templating

multilink bundle-name authenticated

crypto pki trustpoint SFBCA

enrollment terminal

fqdn isr4k.skypelabsk.local

subject-name CN=isr4k.skypelabsk.local

revocation-check none

rsakeypair SFBCAKey


crypto pki certificate chain SFBCA

certificate 5400000017CCECC867005D14ED0000000000017

30820525 3082040D A0030201 02021354 00000017 CCECC867 005D14ED 00000000
0017300D 06092A86 4886F70D 01010505 00304E31 15301306 0A099226 8993F22C
64011916 056C6F63 616C311A 3018060A 09922689 93F22C64 0119160A 736B7970
656C6162 736B3119 30170603 55040313 10736B79 70656C61 62736B2D 44432D43
41301E17 0D313531 32303933 39313030 385A170D 31373132 30383139 31303038
5A302131 1F301D06 30550403 13166973 72346B2E 736B7970 656C6162 736B2E6C
6F63616C 30820122 300D0609 2A864886 F70D0101 01050003 82010F00 3082010A
02820101 00B8F496 C7301835 BD6448F1 A6826B11 F46E75E2 141D6B2F 3794F139
639C8760 6AC90F46 14E4E7D0 74A33B52 CC7B77A7 55815F6A 10E767CB 45968A94
6DB83F89 B3709C83 36F1E44D FED6A78C 447C6068 DD86E04E 1637BE4C F9CD2938
5A91B593 B840ADE8 E1EFE131 B2089BB0 3F83AE3D 2CD878FC B9286508 212CE087
D7A69F12 81DAB2F2 AD31CCFF 50932BE5 710BB7B5 6D584CC2 783941BC BC9F5AF7
B00408A8 81186776 D72F62E3 0032155A A0ED0853 9CEFB562 B1AE12FC EC855041
A8DF0638 3C1E7EC3 259288C5 75FE9004 1D8E6956 DAE0D1C 898AFA1A 175614BB
quit
certificate ca 4776AE3591B921AF45CB8415421DF2DD
30820377 3082025F A0030201 02021047 76AE3591 B921AF45 CB841542 1DF2DD30
0D06092A 864886F7 0D010105 0500304E 31153013 060A0992 268993F2 26640119
16056C6F 63616C31 1A301806 0A099226 8993F22C 64011916 0A736B79 70656C61
62736B31 19301706 03550403 1310736B 7970656C 6162736B 2D44432D 4341301E
170D3135 30323034 31363536 31385A17 0D323030 32303431 37303631 375A304E
31153013 060A0992 268993F2 26640119 16056C6F 63616C31 1A301806 0A099226
8993F22C 64011916 0A736B79 70656C61 62736B31 19301706 03550403 1310736B
7970656C 6162736B 2D44432D 43413082 0122300D 06092A86 4886F70D 01010105
00038201 0F003082 010A0282 010100B1 55EF48A2 FC6A1500 5314D0CC F2D9381E
991FF2FB 012EF82F 15F9DC2E C1C94F15 77AA4117 3529D9BD C0197CDA 74A963A5
41D7B081 3E1615DB 893C7693 7AEE8D88 0EC572CD 87791C64 86A5D2EB 5BF3A54C
DD63C4CE BEC3D834 E77A7E9 F0E66DF6 729D612C B649ADFA 7837C660 0D932964
4F481344 B3A51479 3F6F9D98 D7488E2D 6BC145EE 6A6FA2CF 90C01AC6 E2C19D91
D4873656 A7625FB9 4FFCB793 09E7F825 68108464 A8C94B1A 5F0E4D9B B54711BD
162FF78C 09952D8F 94B1318A 77E37F00 E32EB1CB 63915804 844DAD2C E26FC898
3117F86F E78A0AE7 3F5FCA67 919C9BA4 B5D3C05E 74656D18 6AA19F39 1431A8E5
E10D6F75 5A6A77A6 A50C3C7E CE4B4B02 03010001 A351304F 300B0653 551DF040
04030201 86300F06 03551D13 0101F004 05300301 01FF301D 0603551D 0E416046
14590DF8 7D6E08A3 424ABC7A 85FAFD4E 0F4BDDA5 B8301006 092B0601 04018237
15010403 02010030 0D06092A 864886F7 0D010105 05000382 01010052 961F4C96
4BE0ECEE DF1E49EF 98C1E483 D0AA05A5 955DB3C6 200ACC85 FFD7E8D9 8182430E
8C22880F 3B4D71A6 F5848ED5 0B5F6A49 17A53F20 A7D45CF 4796C2E9 298C1B28
1584B717 13435874 4D31C21F 5F26535D D685E41A 74399A1E AC5D3A05 BCD484EF
F37BD773 51B23F3B A1928226 1088D276 F4583E6E DB3B2D74 4FD66E21 00031A4C
1EAC877E D5246E96 8323FFC5 DD3F2C77 82364246 7411AC35 24D89BDE 9DEFD183
1BCFA3A F53AD01E 0225D825 0A094649 840DD901 EBE87F00 2F44CE0E DFF19546
quit
cs logging verbose
!
voice service voip
  no ip address trusted authenticate
  rtcp keepalive
  address-hiding
  mode border-element license capacity 20
  allow-connections sip to sip
  redundancy-group 1
  no supplementary-service sip moved-temporarily
  no supplementary-service sip refer
  supplementary-service media-renegotiate
  sip
    bind control source-interface GigabitEthernet0/0/2
    bind media source-interface GigabitEthernet0/0/2
    session refresh
    header-passing
    refer-to-passing
    conn-reuse
    midcall-signaling passthru media-change
    sip-profiles inbound
    sip-profiles 248 inbound
  !
voice class uri PSTN sip
  host ipv4:63.87.147.30
voice class uri SFB sip
    host fe.skypelabslsk.local

voice class uri FQDNsfb sip
    host 10.80.22.74

voice class codec 2
    codec preference 1 g711ulaw
    codec preference 2 g729r8 bytes 30
    codec preference 3 g726r32
    codec preference 4 g729br8

voice class sip-profiles 2
    request ANY sip-header Allow-Header modify ",PRACK" ""

voice class sip-profiles 203

voice class sip-profiles 248
    request ANY sdp-header Audio-Attribute modify
        "a=(.*)AES_CM_128_HMAC_SHA1_(.*) inline(.*)\|(.*)\|(.*)" "a=\"CRYPTO_UNKNOWN inline\3\4:5"
    response ANY sdp-header Audio-Attribute modify
        "a=(.*)AES_CM_128_HMAC_SHA1_(.*) inline(.*)\|(.*)\|(.*)" "a=\"CRYPTO_UNKNOWN inline\3\4:5"
voice class sip-profiles 1

request INVITE sip-header Referred-By remove
request INVITE sip-header Diversion remove
request INVITE sip-header Diversion add "Diversion: <sip:7193779214@192.65.79.124>"
request INVITE sip-header P-Asserted-Identity remove

voice class sip-copylist 1
sip-header REFERRED-BY

voice translation-rule 1
rule 2 /\(^..............\)$\)/ +1\1/

voice translation-rule 2
rule 1 /7193779226/ /2255/
rule 2 /\(^..............\)$\)/ +1\1/
rule 3 /\(^..............\)$\)/ +1\1/

voice translation-rule 3
rule 1 /\^+1\(..............\)$\)/ /1/1/
rule 3 /\^9\(..............\)$\)/ /1\1/
rule 4 /\^9\(01191..............\)$\)/ /1/1/

voice translation-rule 4
rule 1 /\^+1\(..............\)$\)/ /1/1/
voice translation-profile e164
translate calling 1
translate called 2
!
voice translation-profile pstn
translate calling 4
translate called 3
!
media service

license accept end user agreement
license boot level appxk9
license boot level uck9
license boot level securityk9
!
spanning-tree extend system-id
!
redundancy
mode none
application redundancy
group 1
  name voice-b2bha
  priority 100 failover threshold 75
timers delay 30 reload 60
control GigabitEthernet0/0/0 protocol 1
data GigabitEthernet0/0/0
track 1 shutdown
track 2 shutdown
! vlan internal allocation policy ascending
!
track 1 interface GigabitEthernet0/0/2 line-protocol
track 2 interface GigabitEthernet0/0/3 line-protocol
!
translation-rule 25
  rule 1 7193779214 18563330430
!
interface GigabitEthernet0/0/0
  ip address 10.64.4.19 255.255.0.0
  media-type rj45
  negotiation auto
!
interface GigabitEthernet0/0/1
  no ip address
  shutdown
  media-type rj45
  negotiation auto
!
interface GigabitEthernet0/0/2
  description LAN interface
  ip address 10.80.22.74 255.255.255.0
  media-type rj45
  negotiation auto
  redundancy rii 1
  redundancy group 1 ip 10.80.22.100 exclusive
!
interface GigabitEthernet0/0/3
  description WAN interface
  ip address 192.65.79.114 255.255.255.224
  media-type rj45
  negotiation auto
  redundancy rii 2
  redundancy group 1 ip 192.65.79.124 exclusive
!
interface GigabitEthernet0
  vrf forwarding Mgmt-intf
  no ip address
  shutdown
  negotiation auto
!
interface Vlan1
  no ip address
  shutdown
!
ip forward-protocol nd
ip ftp username admin
ip ftp password admin
no ip http server
no ip http secure-server
ip tftp source-interface GigabitEthernet0
ip route 0.0.0.0 0.0.0.0 192.65.79.97
ip route 10.64.0.0 255.255.0.0 10.80.22.1
ip route 172.16.2.0 255.255.255.0 10.80.22.1
ip route 172.16.29.0 255.255.255.0 10.80.22.1
ip route 172.16.31.0 255.255.255.0 10.80.22.1
ip ssh version 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
!
sccp ccm 10.64.1.72 identifier 1 version 4.0
!
sccp ccm group 1
    associate ccm 1 priority 1
    associate profile 1 register XCODE123456
    keepalive retries 1
    keepalive timeout 10
    switchover method immediate
    switchback method immediate
!
sccp ccm group 2
    associate ccm 1 priority 1
    associate profile 2 register SCCPTLS
!
telephony-service
max-conferences 8 gain -6
transfer-system full-consult
!

dspfarm profile 2 transcode universal
associate application SCCP
shutdown
!
dspfarm profile 1 transcode universal
associate application SCCP
shutdown
!
dial-peer voice 107 voip
  description "Incoming PSTN to IP-PBX - IP-PBX facing side"
  translation-profile outgoing e164
  destination-pattern 719....... 
  session protocol sipv2
  session target dns:fe.skypelabsk.local:5067
  session transport tcp tls
  incoming uri request FQDNsfb
  voice-class codec 2
  voice-class sip srtp-auth sha1-80 sha1-32
  voice-class sip localhost dns:isr4k.skypelabsk.local:5061
  voice-class sip asserted-id pai
  voice-class sip call-route url
  voice-class sip profiles 248
  voice-class sip options-keepalive

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voice-class sip bind control source-interface GigabitEthernet0/0/2
voice-class sip bind media source-interface GigabitEthernet0/0/2
voice-class sip referto-passing
dtmf-relay rtp-npe
refer consume
rtcp consume
srtp keepalive
srtp
no vad!

dial-peer voice 105 voip
description incoming from IP-PBX
session protocol sipv2
session transport tcp tls
incoming called-number 9........
incoming uri request SFB
voice-class codec 2
voice-class sip srtp-auth sha-80 sha-32
voice-class sip localhost dns:isr4k.skypelabsk.local:5061
no voice-class sip asserted-id
voice-class sip call-route url
no voice-class sip srtpe negotiate
voice-class sip bind control source-interface GigabitEthernet0/0/2
voice-class sip bind media source-interface GigabitEthernet0/0/2
voice-class sip requri-passing
dtmf-relay rtp-npe
srtp
no vad!

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dial-peer voice 400 voip
description "Outgoing To PSTN"
translation-profile outgoing pstn
destination-pattern .T
rtp payload-type comfort-noise 13
session protocol sipv2
session target ipv4:63.87.147.30:5072
session transport udp
voice-class codec 2
voice-class sip rel1xx disable
voice-class sip privacy-policy passthru
voice-class sip profiles 1
voice-class sip session refresh
voice-class sip bind control source-interface GigabitEthernet0/0/3
voice-class sip bind media source-interface GigabitEthernet0/0/3
dtmf-relay rtp-nre
rtcp keepalive
srtp fallback
no vad
!
dial-peer voice 201 voip
description incoming from PSTN
translate-outgoing called 25
session protocol sipv2
session transport udp
incoming called-number 719.......
voice-class codec 2
voice-class sip rel1xx disable
voice-class sip profiles 2 inbound
voice-class sip session refresh
voice-class sip bind control source-interface GigabitEthernet0/0/3
dtmf-relay rtp-nce
srtp fallback
no vad
!
dial-peer voice 786 voip
description TEST for REFER
translation-profile outgoing e164
session protocol sipv2
session target sip-uri
session transport tcp tls
destination uri SFB
voice-class codec 2
voice-class sip srtsp-auth shal-80 shal-32
voice-class sip localhost dns:isr4k.skypelabsk.local:5061
voice-class sip profiles 248
voice-class sip bind control source-interface GigabitEthernet0/0/2
voice-class sip bind media source-interface GigabitEthernet0/0/2
voice-class sip refer-to-passing
voice-class sip requiri-passing
dtmf-relay rtp-nce
srtp
no vad
!
!
gateway
  media-inactivity-criteria all
  timer receive-rtcp 5
  timer receive-rtp 86400
!
sip-ua
  no remote-party-id
  timers options 1000
  connection-reuse
  crypto signaling default trustpoint SFBCA
!
line con 0
  logging synchronous
  stopbits 1
line aux 0
  stopbits 1
line vty 0
  session-timeout 90
  exec-timeout 960 0
  no activation-character
  logging synchronous
  transport preferred ssh
  transport input all
  stopbits 1
line vty 1 4
  exec-timeout 960 0
  logging synchronous
  transport input all
end

CISCO_4K_ROUTER1#
Standby Cisco UBE:

! Last configuration change at 19:45:02 UTC Fri Feb 12 2016
!
version 15.6
service timestamps debug datetime msec
service timestamps log datetime msec
no platform punt-keepalive disable-kernel-core
!
hostname CISCO_4K_ROUTER2
!
boot-start-marker
boot system flash bootflash:isr4400-universalk9.03.17.00.S.156-1.S-
std.SPA.bin
boot-end-marker
!
!
vrf definition Mgmt-intf
!
address-family ipv4
exit-address-family
!
address-family ipv6
exit-address-family
!
enable password cisco
!
aaa new-model
!
aaa session-id common
!

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ipc zone default
association 1
  no shutdown

ip host dc.skypelabsk.local 10.64.3.230
ip host fe.skypelabsk.local 10.64.3.231
ip host isr4k.skypelabsk.local 10.80.22.100
ip name-server 8.8.8.8

no ip domain lookup

ipv6 unicast-routing

subscriber templating

multilink bundle-name authenticated

crypto pki trustpoint SFBCA
  enrollment terminal
  fqdn isr4k.skypelabsk.local
  subject-name CN=isr4k.skypelabsk.local
  revocation-check none
  rsakeypair SFBCAKey


crypto pki certificate chain SFBCA
  certificate 540000000251CFe0A478A81E0A80000000000025
    30820525 3082040D A0030201 02021354 00000025 1CFe0A47 8A81E0A8 00000000
    0025300D 06092A86 4886F70D 01010505 00304E31 15301306 0A099226 8993F22C
    64011916 056C6F63 616C311A 3018060A 09922689 93F22C64 0119160A 736B7970
    656C6162 736B3119 30170603 55040313 10736B79 70656C61 62736B2D 44432D43
    41301317 0D313630 32313231 37343331 335A170D 31383032 31313137 34333133
1CF5801A A0A5FFA8 7CC07F79 91578935 60F5A032 064C281C 48ED81BE 0B827DDC 92204AFF 1B5A5918 FAADBA6E A2D73E0D B7279E26 70B3C67C 8E12CDCA 121AF195 CAFB1C60 7B4E067F 26E99686 CFD80B03 808424C6 356C65ED D892D3E0 481A65F3 769E5D1F CD29158C C97D25C2 31144DD4 851B4DEC B8DB86C1 DB134961 86ADF7B7D 09713AD0 3FC87F28 3E

quit
certificate ca 4776AE3591B921AF45CB8415421DF2DD
30820377 3082025F A0030201 02021047 76AE3591 B921AF45 CB841542 1DF2DD30 0D06092A 864886F7 0D010105 0500304E 31153013 060A0992 268999F2 2C640119 16056C6F 63616C31 1A301806 0A099226 8993F22C 64011916 0A736B79 70656C61 62736B31 19031706 03550403 1310736B 7970656C 6162736B 2D44432D 4341301E 170D3135 30323034 31363536 31385A17 0D323030 32303431 37303631 375A304E 31153013 060A0992 268999F2 2C640119 16056C6F 63616C31 1A301806 0A099226 8993F22C 64011916 0A736B79 70656C61 62736B31 19031706 03550403 1310736B 7970656C 6162736B 2D44432D 43413018 0122300D 06092A86 4886F70D 01010105 00038201 00030820 0101001B 55EF48A2 FC6A1500 5314D0CC F2D9381E 991FF2FB 012EF82F 15F9DCE2 C1C94F15 77AA4A17 3529D9B0 C0197CDA 74A963A5 4D7B081 3E1615DB 893C7693 7AEEB838 0EC572CD 87791C64 86A5D2EB 5BF3A54C DD63C4CE BEC30834 E77AA7E9 F0E66DF6 729D612C B649ADFA 7837C660 0D932964 4F481344 B3A51479 3F6F9D98 D7488E2D 6BC145EE 6A6FA2CF 90C01AC6 E2C19D91 D4873656 A7625FB9 4FCCB793 09E7F825 68108464 A8C94B1A 5F0E4D9B B54711BD 162FF78C 09952D8F 94B1318A 77E37F00 E32EB1CB 63915804 844DAD2C E26FC898 3117F86F E78A0AE7 3F5FCA67 919C9BA4 B5D3C05E 74656D18 6AA19F39 1431A8E5 E10DF75 5A6A77A6 A50C3CE7 CE4B4B02 03010001 A351304F 30BB0603 551D0F04 04030201 86300F06 03551D13 0101FF04 05300301 01FF301D 0603551D 0E041604 14590DF8 7D6E08A3 4244BC7A 85FAFD4E 0F4BDDA5 B8301006 092B0601 04018237 15010403 02010030 0D06092A 864886F7 0D010105 05000382 01010052 961FC96 4BE0EECE DF1E49EF 98C1E4B3 D0A05A5 955DB3C6 200ACC5B FFD7E8D9 8182430E 8C22B880F 3B4D71A6 F5848ED5 08F56A49 17A53F20 A75D45CF 4796C2E9 298C1B28 1584B717 13435874 4D31C21F 5F26535D D685E4A1 74399A1E AC5D3A05 BCD484EF F37BD773 51B23F3B A1928226 1088D276 F45833E6 DB3B2D74 4FD66E21 00031A4C 1E3C877E D5246E96 83233FCC D53F2C77 82364246 7411AC35 24D898BE 9DEFD183

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quit

crts logging verbose
!

voice service voip

no ip address trusted authenticate
rtcp keepalive
address-hiding
mode border-element license capacity 20
allow-connections sip to sip
redundancy-group 1
no supplementary-service sip moved-temporarily
no supplementary-service sip refer
supplementary-service media-renegotiate
sip
bind control source-interface GigabitEthernet0/0/2
bind media source-interface GigabitEthernet0/0/2
session refresh
header-passing
refto-passing
conn-reuse
midcall-signaling passthru media-change
sip-profiles inbound
sip-profiles 248 inbound
!
!
voice class uri PSTN sip
host ipv4:63.87.147.30
!
voice class uri SFB sip
  host fe.skypelabsk.local
!
voice class uri FQDNsfb sip
  host 10.80.22.74
voice class codec 2
  codec preference 1 g711ulaw
  codec preference 2 g729r8 bytes 30
  codec preference 3 g726r32
  codec preference 4 g729br8
!
voice class sip-profiles 248
  request ANY sip-header Audio-Attribute modify
  "a=(.*)AES_CM_128_HMAC_SHA1_(.*) inline(.*)\|(.*)\|(.*)\|(.*)" "a=\1CRYPTO_UNKNOWN inline\3\4:5"
  response ANY sip-header Audio-Attribute modify
  "a=(.*)AES_CM_128_HMAC_SHA1_(.*) inline(.*)\|(.*)\|(.*)\|(.*)" "a=\1CRYPTO_UNKNOWN inline\3\4:5"
!
voice class sip-profiles 2
  request ANY sip-header Allow-Header modify ",PRACK" ""
!
voice class sip-profiles 1
  request INVITE sip-header Referred-By remove
  request INVITE sip-header Diversion remove
  request INVITE sip-header Diversion add "Diversion: <sip:7193779214@192.65.79.124>"
  request INVITE sip-header P-Asserted-Identity remove
!
voice class sip-copylist 1
  sip-header REFERRED-BY
!
voice translation-rule 1
  rule 2 /\(^{................} $\)/ /+1\1/
voice translation-rule 2
rule 1 /7193779226/ /2255/
rule 2 /\(.*\)\)/ /+1\1/
rule 3 /\(.*\)\)/ /+1\1/

voice translation-rule 3
rule 1 /\+1\(.*\)\)/ /\1/
rule 3 /\9\(.*\)\)/ /\1\1/
rule 4 /\9\(01191.*\)\)/ /\1\1/

voice translation-rule 4
rule 1 /\+1\(.*\)\)/ /\1/

voice translation-profile e164
translate calling 1
translate called 2

voice translation-profile pstn
translate calling 4
translate called 3

license udi pid ISR4431/K9 sn FOC18232988
license boot level appxk9
license boot level uck9
license boot level securityk9

spanning-tree extend system-id

username cisco privilege 15 secret 5 $1$AGR7$e7pQx6UI0be3bzRbc01r81
redundancy
mode none
application redundancy
group 1
  name voice-b2bha
  priority 100 failover threshold 75
timers delay 30 reload 60
control GigabitEthernet0/0/0 protocol 1
data GigabitEthernet0/0/0
track 1 shutdown
track 2 shutdown
!
vlan internal allocation policy ascending
!
track 1 interface GigabitEthernet0/0/2 line-protocol
track 2 interface GigabitEthernet0/0/3 line-protocol
!
interface GigabitEthernet0/0/0
  ip address 10.64.4.20 255.255.0.0
  media-type rj45
  negotiation auto
!
interface GigabitEthernet0/0/1
  no ip address
  shutdown
  media-type rj45
  negotiation auto
!
interface GigabitEthernet0/0/2
   ip address 10.80.22.75 255.255.255.0
   media-type rj45
   negotiation auto
   redundancy rii 1
   redundancy group 1 ip 10.80.22.100 exclusive
!
interface GigabitEthernet0/0/3
   description Wan Interface
   ip address 192.65.79.115 255.255.255.224
   media-type rj45
   negotiation auto
   redundancy rii 2
   redundancy group 1 ip 192.65.79.124 exclusive
!
interface GigabitEthernet0
   vrf forwarding Mgmt-intf
   no ip address
   shutdown
   negotiation auto
!
interface Vlan1
   no ip address
   shutdown
!
ip forward-protocol nd
no ip http server
no ip http secure-server
ip route 0.0.0.0 0.0.0.0 192.65.79.97
ip route 10.64.0.0 255.255.0.0 10.80.22.1
ip route 10.80.22.0 255.255.255.0 10.80.22.1
ip route 172.16.0.0 255.255.0.0 10.80.22.1
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 107 voip
description "Incoming PSTN to IP-PBX - IP-PBX facing side"
translation-profile outgoing e164
destination-pattern 719....... 
session protocol sipv2
session target dns:fe.skypelabsk.local:5067
session transport tcp tls
incoming uri request FQDNsf
voice-class codec 2
voice-class sip srtp-auth sha1-80 sha1-32
voice-class sip localhost dns:isr4k.skypelabsk.local:5061
voice-class sip asserted-id pai
voice-class sip call-route url
voice-class sip profiles 248
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/0/2
voice-class sip bind media source-interface GigabitEthernet0/0/2
voice-class sip refer-to-passing
dtmf-relay rtp-nte
refer consume
rtcp keepalive
srtp
no vad
dial-peer voice 105 voip
description incoming from IP-PBX
session protocol sipv2
session transport tcp tls
incoming called-number 9..........
incoming uri request SFB
voice-class codec 2
voice-class sip srtp-auth sha1-80 sha1-32
voice-class sip localhost dns:isr4k.skypelabsk.local:5061
no voice-class sip asserted-id
voice-class sip call-route url
no voice-class sip srtp negotiate
voice-class sip bind control source-interface GigabitEthernet0/0/2
voice-class sip bind media source-interface GigabitEthernet0/0/2
voice-class sip requri-passing
dtmf-relay rtp-npe
srtp
no vad
!
dial-peer voice 400 voip
description "Outgoing To PSTN"
translation-profile outgoing pstn
destination-pattern .T
rtp payload-type comfort-noise 13
session protocol sipv2
session target ipv4:63.87.147.30:5072
session transport udp
voice-class codec 2
voice-class sip rel1xx disable
voice-class sip privacy-policy passthru
voice-class sip profiles 1
voice-class sip bind control source-interface GigabitEthernet0/0/3
voice-class sip bind media source-interface GigabitEthernet0/0/3
dtmf-relay rtp-nte
rtcp keepalive
srtp fallback
no vad
!
dial-peer voice 201 voip
description incoming from PSTN
translate-outgoing called 25
session protocol sipv2
session transport udp
incoming called-number 719....... 
voice-class codec 2
voice-class sip re1xx disable
voice-class sip profiles 2 inbound
voice-class sip bind control source-interface GigabitEthernet0/0/3
voice-class sip bind media source-interface GigabitEthernet0/0/3
dtmf-relay rtp-nte
srtp fallback
no vad
!
dial-peer voice 786 voip
description TEST for REFER
translation-profile outgoing e164
session protocol sipv2
session target sip-uri
session transport tcp tls
destination uri SFB
voice-class codec 2
voice-class sip srtp-auth sha1-80 sha1-32
voice-class sip localhost dns:isr4k.skypelabsk.local:5061
voice-class sip profiles 248
voice-class sip bind control source-interface GigabitEthernet0/0/2
voice-class sip bind media source-interface GigabitEthernet0/0/2
voice-class sip referto-passing
dtmf-relay rtp-nre
srtp
no vad
!
gateway
media-inactivity-criteria all
timer receive-rtcp 5
timer receive-rtp 86400
!
sip-ua
no remote-party-id
timers options 1000
connection-reuse
crypto signaling default trustpoint SFBCA
!
line con 0
  stopbits 1
line aux 0
  stopbits 1
line vty 0
  session-timeout 90
  exec-timeout 960 0
  password tekV1z10n
  no activation-character
  logging synchronous
  transport preferred ssh
  transport input all
stopbits 1
line vty 1 4
extec-timeout 960 0
password tekV1z10n
logging synchronous
transport input all
!
!
end
Configuring Microsoft Skype for Business Server

Trunk configuration from Skype for Business to the LAN side of the CUBE.

**PSTN gateway configuration**

**Trunk to CUBE using TCP**

- Open Skype for Business Server 2015 Topology Builder and navigate to Shared Components → PSTN gateways → New IP/PSTN Gateway

![Image of Topology Builder](image-url)

Figure 3 Add new IP/PSTN Gateway in Skype for Business Topology Builder

- The FQDN of the CUBE is configured in the new SIP trunk wizard.
Figure 4 Define the FQDN of the CUBE
Select “Enable IPv4” with all configured IP Address.

Figure 5: IPv4/IPv6 address selection
Please configure the port number, transport protocol and associated Mediation Server.

Figure 6: Transport protocol and Port number
• After completion, the newly created trunk will appear under the PSTN gateways with the associated Mediation Server as shown below.

Figure 7 PSTN Gateway is added successfully using TCP

Figure 8 Trunk to CUBE is added successfully in TCP
Trunk to CUBE using TLS

- Open Skype for Business Server 2015 Topology Builder and navigate to Shared Components → PSTN gateways → New IP/PSTN Gateway

![Skype for Business Server 2015, Topology Builder](image)

Figure 9 Add new IP/PSTN Gateway in Skype for Business Topology Builder
The FQDN of the CUBE is configured in the new SIP trunk wizard.

Figure 10 FQDN of the CUBE
• Select “Enable IPv4” with all configured IP Address.

Figure 11: IPv4/IPv6 address selection
Please configure the port number, transport protocol and associated Mediation Server.

Figure 12: Transport protocol and Port number
After completion, the newly created trunk will appear under the PSTN gateways with the associated Mediation Server as shown below.

Figure 13 PSTN Gateway to CUBE is added successfully using TLS

Figure 14 Trunk to CUBE is added successfully using TLS
Publish the Skype for Business Topology

- Navigate to Action → Topology → Publish
- Follow the wizard to publish the changes in the topology

Figure 15 Publish the Skype for Business topology
Publishing wizard complete

Your topology was successfully published.

<table>
<thead>
<tr>
<th>Step</th>
<th>Status</th>
</tr>
</thead>
<tbody>
<tr>
<td>Publishing topology...</td>
<td>Success</td>
</tr>
<tr>
<td>Downloading topology...</td>
<td>Success</td>
</tr>
<tr>
<td>Downloading global simple URL settings...</td>
<td>Success</td>
</tr>
<tr>
<td>Updating role-based access control (RBAC) roles...</td>
<td>Success</td>
</tr>
<tr>
<td>Enabling topology...</td>
<td>Success</td>
</tr>
</tbody>
</table>

To close the wizard, click Finish.

Figure 16 Changes to the Topology published successfully
Voice Routing Configuration

Voice Policy

- Open Skype for Business Server 2015 Control Panel and select ‘Voice Routing’ and select ‘Voice Policy’. Click on “New” and select “User Policy” to add the new Voice Policy as shown below.

![Figure 17 Voice Routing Configuration](image-url)
- Select the needed features and click “New” as shown below to add a new PSTN Usage.

Figure 18 New User Voice Policy configuration
• Add a new PSTN Usage with name ‘PSTN_Usage_1’ and click on “New” to add a new Route.

Figure 19 New PSTN Usage Record configuration
- Add a new route pointing to CUBE. The Pattern to Match is set to “.*” which matches any dialed number from this Voice Policy.
- Under Associated Trunks, select “Add” to choose the CUBE and associate it with the trunk “Cisco ISR 4K”

Figure 20 New Voice Route details and Add Trunk to associate
Figure 21 Select Trunk to associate with Route

Figure 22 Commit changes to Voice Policy
Figure 23 Voice Policy added successfully
Trunk configuration

- In Skype for Business Server Control Panel, navigate to Voice Routing ➔ Trunk Configuration
- Select New ➔ Pool Trunk to add the trunk to CUBE

Figure 24 New Pool Trunk
Figure 25 Select the Trunk Service

- Add the trunk with the corresponding requirements as shown below and associate the newly added PSTN Usage
Figure 26 New Trunk Configuration
Figure 27 Associate PSTN Usage to the Trunk
Figure 28 Commit the Changes to the Trunk
Figure 29 Trunk to CUBE added successfully
User Configuration

- Select the Skype Users who are intend to use the Enterprise Voice feature and fill the Verizon DID in the Line URI field. Select the newly added User Voice Policy in ‘Voice Policy’ field.
Figure 31 Skype User Configuration 2
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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