Microsoft Skype for Business Server [v6.0.9319.0] to Verizon Business SIP Trunk via Cisco Unified Border Element 11.5.2 [v16.3.1a, ISR4431/XE- 16.3.1a]

February 03, 2017
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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 11.5.2) IOS-XE 16.03.1a can be used. The Cisco Unified Border Element (Cisco UBE) 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 11.5.2) IOS-XE version 16.03.1a 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 11.5.2) IOS-XE version 16.03.1a.

- 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 using it FQDN. 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 TLS with SRTP.

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

<table>
<thead>
<tr>
<th>Setting</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Media Bypass</td>
<td>ON</td>
</tr>
<tr>
<td>Encryption Support</td>
<td>ON</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>
<tr>
<td>Transport from CUBE to Skype for Business</td>
<td>TLS with SRTP</td>
</tr>
<tr>
<td>Transport from CUBE to Service Provider</td>
<td>UDP with RTP</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: 11.5.2, SW-Version: 16.03.1a, XE- 16.03.1a
- Microsoft Skype for Business Server 2015- Version: 6.0.9319.0
- Microsoft Skype for Business Client – Version 15.0.4867.1001

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 using TLS with 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 ON in trunk between Skype for business and CUBE. So the media flows directly from Skype to CUBE and vice versa.
- 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
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.

![High Availability topology](image)

**Figure 2** High Availability topology

CUBE 1:
```
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 192.65.79.140 255.255.255.128
  media-type rj45
  negotiation auto
  redundancy rii 2
  redundancy group 1 ip 192.65.79.180 exclusive
!
CUBE 2:

! 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 WAN
  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 192.65.79.141 255.255.255.128
  media-type rj45
  negotiation auto
  redundancy rii 2
  redundancy group 1 ip 192.65.79.180 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 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
  sip-profiles 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>
</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
Dial peers

**Dial-peer to Skype for Business using 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:fe01.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 options-keepalive
    voice-class sip session refresh
    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-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 +1...........
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 copy-list 1
voice-class sip session refresh
voice-class sip bind control source-interface GigabitEthernet0/0/2
voice-class sip bind media source-interface GigabitEthernet0/0/2
voice-class sip requiri-passing
dtmf-relay rtp-nce
rtcp keepalive
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 srtp-auth sha1-80 sha1-32
voice-class sip localhost dns:isr4k.skypelabsk.local:5061
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
rtcp keepalive
srtp
no vad
!

Dial-peer to Verizon using UDP and RTP:
!
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 ipv4:152.188.29.149:5072
session transport udp
voice-class sip rel1xx supported "100rel"
voice-class sip asserted-id pai
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-nte
rtcp keepalive
codec g711ulaw
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 sip rel1xx supported "100rel"
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-n-te
rtcp keepalive
codec g711ulaw
no vad
!
Configuration example
The following configuration snippet contains a sample configuration of Cisco UBE with all parameters mentioned previously.

Active Cisco UBE:

! version 16.3
service timestamps debug datetime msec
service timestamps log datetime msec
no platform punt-keepalive disable-kernel-core
!
hostname ISR4KR1
!
boot-start-marker
boot system flash isr4400-universalk9.16.03.01a.SPA.bin
boot-end-marker
!
!
vrf definition Mgmt-intf
!
address-family ipv4
exit-address-family
!
address-family ipv6
exit-address-family
!
logging buffered 9999999
no logging console
enable password *****
no aaa new-model

ipc zone default
association 1
no shutdown

ip host dc.skypelabsk.local 10.64.4.10
ip host fe01.skypelabsk.local 10.64.4.11
ip host isr4k.skypelabsk.local 10.80.22.75
ip name-server 10.64.4.10

no ip domain lookup

subscriber templating

multilink bundle-name authenticated

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

!  
crypto pki certificate chain SFBCA  
certificate 5000000004E4EBA4EF5C77B52D6000000000004E
656C6162 736A2C44 433D6C6F 63616C3F 634A4365 72746966 69636174 653F6261
73653F6F 626A6563 74436C61 73733D63 65727469 66696361 74696F6E 41757468
6F726974 79303D60 092B0601 04018237 15070430 30E0626 2B060104 01823715
0887C882 5984899B 76828187 3780D082 7386C9D7 0F6B879C B26A82DB CD030201
6402010A 301D0603 551D2504 16301406 082B0601 05050703 0106082B 06010505
07030230 2706092B 06010401 8237150A 041A3018 30A0608 2B060105 05070301
300A0608 2B060105 05070302 30D0609 2A864886 F70D0101 05050003 82010100
9D150FAD E8D1791C 26C471AC 4561F763 9E8E29FB 340A7CDD 17D8A0A6 0917ED39
652A39C8 86129758 8C293DB 77E6962D 035C58A1 9B0E5A2B C825A940 91DA6F06
FD93CB3 3EB56D2C D2DA8B8E 109B0CA9D E83CEAC2 D2DA1C37 6E59BAF9 7EAA6C6E
9B78F61B C8B38D79 EED2C302 9CAD5A55 A968492D 2DE39F9C 9675CF62 366966CB
9EEB06C9 F5244953 782F3925 849C6358 8099CA6D FA26727E 5158646A B6C67EDD
052FA89B 1B58C7B1 3A969ED3 07AAE3EB C6576CE9 D67530F8 A9B6D940 0B8C341
209CEBD1 68893E64 FBDEA75 4BC14FBE 05F0115A A604C026 055D752E D64D6DC5
FF8B550 3F135130 EEA7CE13 BE75F9E8 9948A6FA 320888A8 B1C04CE1 DB9A3F91
quit
certificate ca 2C0BFAFACCBD24A1420DEF837B9FBC8F
3082037B 30820263 A0030201 0202102C 0BFAFACC BD24A142 0DEF837B 9FBC8F30
0D06092A 864886F7 0D010105 05003050 31153013 060A0992 268993F2 2C640119
16056C6F 63616C31 1A301806 0A992268 8993F22C 64011916 0A736B79 70656C61
62736A31 1B301906 03550403 1312736B 7970656C 6162736A 2D444330 312D4341
301E170D 31353036 30393138 35383534 5A170D32 30303630 39313930 3B35325A
30503115 3013060A 09922689 93F22C64 01191605 6C6F6361 6C311A30 18060A09
92268993 F22C6401 19160A73 6B797065 6C616273 6A311B30 19060355 04031312
736B7970 656C6162 736A2D44 4330312D 43413082 0122300D 06092A86 4886F70D
01010105 00038201 00F00382 010A0282 010100A4 65F31045 74F51718 C32E37E1
7EE69305 B93C6A07 7DEA2BAD EB854545 2C2A4569 AF0CF2A7 DD525288 7FA9F0BB
7F7BD9DFE E05CA19 9C205F8E E3C913E7 753B5A88 A289CE4B B184E265 EEEED984
BFA4FE27 AC778CD4 5D76A2EF 43F2DB10 12470BF8 0807EC2F EA64D0DC 386B38EC

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0A46454A A456DBD8 FB0AE0B5 B9AFD285 38F818C9 8E3BDC39 47CE2905 378A9128
1836C101 7C368DB9 8509281F 6F12920A 971257DD CCC23BE9 92860C8C CD47B52C
17887B9F A20B2995 FA26D0F9 6C34B64D 6726B76E 85AEA675 C61141CF 3382836E
6392C6EE 66F62BAB 2E72A77B 24A7A14E C34A7439 F7C460D3 DA0FB17C 9D9DC25A
DAFB62EE 850CF72F FC069549 773D503B 44D3B102 03010001 A351304F 300B0603
551D0F04 04030201 86300F06 03551D13 0101FF04 05300301 01FF301D 0603551D
0E041604 14D17C5C D92CCBD2 2D8BFC99 ABAF07D1 944522C0 18301006 092B0601
04018237 15010403 02010030 0D06092A 864886F7 0D010105 05000382 01010030
7404A96F 9B790586 F4A8D827 5CB8BD58 E692E9CF 2C7F9D897 768E7F85 FF1A717D
B2214917 5B2C8417 00787E4F EA21E1D3 F8B303DB CBB8A955 7314B955 959B47F3
48FBD08F 79038EE7 AC560B06 4612D894 B01E573B 4D76A02F D084C90C 2D289F8A
A49C8A68 F2CA91F5 B440384A BB459345 99A901F1 241A8DF4 6A227D48 C902B806
A9F658BA 76086857 3E1DA3E 4CA78EE9 D2255E06 92E464C9 18764495 F753EB53
5A60EB7F 4A8585D7 B32A3563 AE8F90F7 E52D3B46 47F25409 4DAF6214 808D6F2C
FA79E6FF AA11F81E 14C8D6F9 B5DCB6C DBD9216C D6557FF9 D0D0F83F E0F0E004
33974FF B212328 49740D12 E96A1CB 626BBCBC 8E786743 305F0DFB 3F3883

quit

cts logging verbose
!
voice rtp send-receive
!
voice service voip

no ip address trusted authenticate
rtpc keepalive
address-hiding
mode border-element license capacity 20
allow-connections sip to sip
redundancy-group 1
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
  midcall-signaling passthru
  sip-profiles inbound

voice class uri SFB sip
  host FE01.skypelabsk.local

voice class uri FQDNsfb sip
  host 10.80.22.75

voice class uri PSTN sip
  host 152.188.29.149
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 1
request INVITE sip-header Diversion modify "sip:+1" "sip:"
request INVITE sdp-header Audio-Attribute add "a=sendrecv"
!
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 2 /\^19\(\.........\$\)/ /\1\1/
rule 3 /\^9\(\.........\$\)/ /\1/
rule 4 /\^9\(01191\.........\$\)/ /\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 FOC1844988
license boot level appxk9
license boot level uck9
license boot level securityk9
!
diagnostic bootup level minimal
spanning-tree extend system-id
!
username cisco privilege 15 secret 5 xxxx
!
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 192.65.79.140 255.255.255.128
  media-type rj45
  negotiation auto
  redundancy rii 2
redundancy group 1 ip 192.65.79.180 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 0.0.0.0 192.65.79.129
ip route 10.64.0.0 255.255.0.0 10.64.1.1
ip route 10.80.10.0 255.255.255.0 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
dsfpfarm profile 1 conference security

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:fe01.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 options-keepalive
voice-class sip session refresh
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-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 +1...........
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 copy-list 1
voice-class sip session refresh
voice-class sip bind control source-interface GigabitEthernet0/0/2
voice-class sip bind media source-interface GigabitEthernet0/0/2
voice-class sip requiri-passing
dtmf-relay rtp-nte
rtcp keepalive
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 ipv4:152.188.29.149:5072
session transport udp
voice-class sip rel1xx supported "100rel"
voice-class sip asserted-id pai
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-nate
rtcp keepalive
codec g711ulaw
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 sip rel1xx supported "100rel"
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-nate
rtcp keepalive
codec g711ulaw
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 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 requri passing
dtmf-relay rtp-nte
rtcp keepalive
srtp
no vad
!
!
!
sip-ua
  no remote-party-id
retry invite 1
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
  no activation-character
  logging synchronous
  login local
  transport preferred ssh
  transport input all
  stopbits 1
line vty 1
  exec-timeout 960 0
  logging synchronous
  login
  transport input all
line vty 2 4
  exec-timeout 960 0
  logging synchronous
  login
  transport input all
!
end
Standby Cisco UBE:

version 16.3
service timestamps debug datetime msec
service timestamps log datetime msec
no platform punt-keepalive disable-kernel-core
!
hostname ISR4KR2
!
boot-start-marker
boot system flash isr4400-universalk9.16.03.01a.SPA.bin
boot-end-marker
!
!
vrf definition Mgmt-intf
!
address-family ipv4
exit-address-family
!
address-family ipv6
exit-address-family
!
no aaa new-model
!
ip host dc.skypelabsk.local 10.64.4.10
ip host fe01.skypelabsk.local 10.64.4.11
ip host isr4k.skypelabsk.local 10.80.22.75
ip name-server 10.64.4.10
subscriber templating
!
multilink bundle-name authenticated
!
crypto pki trustpoint SFBCA
enrollment terminal
serial-number
fqdn isr4k.skypelabsk.local
subject-name CN=isr4k.skypelabsk.local
revocation-check none
rsakeypair SFBCAKey
!
crypto pki certificate chain SFBCA
certificate 5000000058116986E9168CF5B50000000000058
3082052D 30820415 A0030201 02021350 00000058 116986E9 168CF5B5 00000000
0058300D 06092A86 4886F70D 01010505 00305031 15301306 0A099226 8993F22C
64011916 056C6F63 616C311A 3018060A 09922689 93F22C64 0119160A 736B7970
656C6162 736A311B 30190603 55040313 12736B79 70656C61 62736A2D 44433031
2D434130 1E170D31 36313231 36313235 3234365A 170D3138 31323136 31323532
34365A30 21311F30 1D060355 04031316 69737234 6B2E736B 7970656C 6162736A
2E6C6F63 616C3082 0122300D 06092A86 4886F70D 01010105 00038201 0F003082
010A0282 01010091 D76CB7B3 C5ADE7DA F02EFC19 0FD4AA66 92745046 17568CEA
B848FE0C 1C172A75 C02C7C18 B0DD5C0E 784803C1 B620B0DD E2ED8CA5 C62559C4
034A63D6 738ED941 288DABCE 5EE7A384 0BD06F2C 936FB7BE C54FA75F 622A75F7
226160D6 53CC4921 824E6A11 D0F6AD11 4436A5AB 6E7D577F 988714F5 A4ACF09D
E5F3C96 F8FD10ED 9633F308 DD14608E 48874AF6 911A15F0 A844F194 C489C231
6B94517C 8F8A5647 33ABA177 8C03CA14 123F40B6 3D100108 1AA16B84 BF84FC04
0158843E 56B5B0B3 7640B596 B81D9382 9A723008 0CC693E3 385B9F6D 472867DE
FE83D390 7243586B 0739575C B951CB55 A20CDF9D 4D6B6058 21B34196 268EA3B9

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B198A486 34798D02 03010001 A382022D 30820229 300E0603 551D0F01 01FF0404
030205A0 301D0603 551D0E04 160414E0 1888ECF DAD3C021 DE3E86C1 9C70CBEC
79CB1E30 1F060355 D230418 30168014 D17C5C09 2CCBD22D 8BFC99AB AF07D194
4522C018 3081D206 03551D1F 0481CA30 81C73081 C4A081C1 A081BE86 81BB6C64
61703A2F 2F2F434E 3D736B79 70656C61 62736A2D 44433031 2D43412C 4343D44
4330312C 4343D43 44502C43 4E3D5075 626C6963 25323048 65792532 03563672
76696365 732C434E 3D536572 76696365 732C434E 3D436F66 66696777 72617469
6F6E2C44 3337D36B 7970656C 6162736A 2C44433D 6C6F6361 6C3F6365 72746966
69636174 65526576 6F636174 696F6E64 6973743F 62617365 3F6F626A 65637443
6C617373 3D63524C 44697374 72696275 74696F6E 506F696E 743081C9 06082B06
01050507 01010481 BC3081B9 3081B606 082B0601 05050730 028681A9 6C646170
3A2F2F2F 4343E3D7 6B797065 6C616273 6A2D4443 30312D43 412C434E 3D414941
2C4343E3 5076626C 59632532 304B6579 25323053 65727669 63657232 4343E3D5
65727669 63657232 4343E3D4 6F6E6669 67757261 74696F6E 2C44433D 736B7970
656C6162 736A2C44 433D6C6F 63616C3F 63414365 72746966 69636174 653F6261
73653F6F 626A6563 74436C61 73733D63 65727469 66696361 74696F6E 41757468
6F726974 79302106 0A01B060 04018237 14020414 1E120057 00650062 05300665
00720076 00650072 30130603 551D2504 0C300A06 082B0601 05050703 01300D06
092A8648 86F70D01 01050500 03820101 0007EEE9 D1F21E46 6A6B2E86 6032CDEA
C9746813 45171AD8 7C2973D E4144737 7D979487 8ECA5F6B 1B7E6B88 2868452C
A4615757 56B9C353 164690FE 43D91C59 0EFFBA24 2540352F 2C368E94 5D621FFA
596263B7 0DFDD2216 72C66564 4A54AD4D 75E2DEFF 29F28936 6F3C32F 4872C0C7
1EAEDAFD 22003118 B455E78C D312AC04 15F5B63E 937B43A5 43298950 EC89A3C
80F3FCE4 FF380232 C17C2C8 E363E557 769A60EE FE370196 CEE6F9F2 D82CF789
7FC8067F E483C801 5B8A8BF3 1DFCFC9E 20C548B7 C22E75E5 6BDA4396 6DA54855
6464EFE2 CB8F9728 30A42AF 0837F879 6AFB4E1B 004DF17D 497C4176 0CC57393
796AEAC8 3A98082E CFBFC06 34CB6D30 02
quit
certificate ca 2C0BFAFACCBBD24A1420DE837B9FBC8F

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voice service voip
ip address trusted list
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
redirect ip2ip
fax protocol pass-through g711ulaw
sip
  bind control source-interface GigabitEthernet0/0/2
  bind media source-interface GigabitEthernet0/0/2
  session refresh
  header-passing
  referto-passing
  registrar server
  conn-reuse
  midcall-signaling passthru media-change
  sip-profiles inbound

voice class uri SFB sip
  host FE01.skypelabsk.local

voice class uri FQDNsfb sip
  host 10.80.22.74
voice class uri PSTN sip
  host 152.188.29.149
!
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 1
  request INVITE sip-header Diversion modify "sip:+1" "sip:"
  request INVITE sdp-header Audio-Attribute add "a=sendrecv"
!
voice class sip-copylist 1
  sip-header REFERRED-BY
!
voice translation-rule 1
  rule 2 /\(^\.{0,2}\)$\)/ /+1\1/
!
voice translation-rule 2
  rule 1 /7193779226/ /2255/
  rule 2 /\(^\.{0,2}\)$\)/ /+1\1/
  rule 3 /\(^\.{0,2}\)$\)/ /+1\1/
!
voice translation-rule 3
  rule 1 /\^1\(...........\)$/ /\1/
  rule 2 /\^19\(...........\)$/ /\1\1/
  rule 3 /\^9\(...........\)$/ /\1/
  rule 4 /\^9\(01191...........\)$/ /\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
!

voice-card 0/1
  no watchdog
!

license udi pid ISR4431/K9 sn FOC18441KJL
license accept end user agreement
license boot level appxk9
license boot level uck9
license boot level securityk9
!
diagnostic bootup level minimal
spanning-tree extend system-id
!
username cisco privilege 15 secret 5 x xxxx
!
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 10.64.4.19 255.255.0.0
  media-type rj45
  negotiation auto
!
interface GigabitEthernet0/0/1
  no ip address
  shutdown
  media-type rj45
  negotiation auto
interface GigabitEthernet0/0/2
  description WAN
  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 192.65.79.141 255.255.255.128
  media-type rj45
  negotiation auto
  redundancy rii 2
  redundancy group 1 ip 192.65.79.180 exclusive

interface Service-Engine0/1/0

interface GigabitEthernet0
  vrf forwarding Mgmt-intf
  no ip address
  negotiation auto

interface Vlan1
  no ip address
  shutdown

ip forward-protocol nd
ip http server
no ip http secure-server
ip route 0.0.0.0 0.0.0.0 192.65.79.129
ip route 10.64.0.0 255.255.0.0 10.64.1.1
ip route 172.16.0.0 255.255.0.0 10.80.22.1
!
ipv6 access-list preauth_v6
   permit udp any any eq domain
   permit tcp any any eq domain
   permit icmp any any nd-ns
   permit icmp any any nd-na
   permit icmp any any router-solicitation
   permit icmp any any router-advertisement
   permit icmp any any redirect
   permit udp any eq 547 any eq 546
   permit udp any eq 546 any eq 547
   deny ipv6 any any
!
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:fe01.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 session refresh

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-nate

refer consume

rtp keepalive

srtp

no vad

!

! dial-peer voice 105 voip

description incoming from IP-PBX

session protocol sipv2

session transport tcp tls

incoming called-number +1.......... 

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 copy-list 1
voice-class sip session refresh
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-nce
rtcp keepalive
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 ipv4:152.188.29.149:5072
session transport udp
voice-class sip rel1xx supported "100rel"
voice-class sip asserted-id pai
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-npe
rtp keepalive
codec g711ulaw
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 sip rel1xx supported "100rel"
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-npe
rtp keepalive
codec g711ulaw
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 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-nite
rtcp keepalive
srtp
no vad
!
sip-ua
no remote-party-id
retry invite 1
timers options 1000
connection-reuse
crypto signaling default trustpoint SFBCA
!
line con 0
exec-timeout 0 0
logging synchronous
stopbits 1
line aux 0
privilege level 15
stopbits 1
line vty 0
exec-timeout 0 0
no activation-character
logging synchronous
login local
transport preferred ssh
transport input telnet ssh
line vty 1 4
  exec-timeout 0 0
  logging synchronous
  login local
  transport input telnet ssh
!
ntp server 10.10.10.5
!
!
!
!
!
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 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 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 FQDN of the CUBE](image)

• Select “Enable IPv4” with all configured IP Address.
• Please configure the port number, transport protocol and associated Mediation Server.
After completion, the newly created trunk will appear under the PSTN gateways with the associated Mediation Server as shown below.
Figure 7 PSTN Gateway to CUBE is added successfully using TLS

Figure 8 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 9 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.

![Voice Routing Configuration](image)

Figure 10 Voice Routing Configuration

- Select the needed features and click “New” as shown below to add a new PSTN Usage.
• Add a new PSTN Usage with name ‘PSTN_Usage_1’ and click on “New” to add a new Route.
Figure 12 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 13 New Voice Route details and Add Trunk to associate
Figure 14 Select Trunk to associate with Route

Figure 15 Commit changes to Voice Policy
Figure 16 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 17 New Pool Trunk

Figure 18 Select the Trunk Service
• Add the trunk with the corresponding requirements as shown below and associate the newly added PSTN Usage
Figure 20 Associate PSTN Usage to the Trunk
Figure 21 Commit the Changes to the Trunk
Figure 22 Trunk to CUBE added successfully

User Configuration
- Select the Skype Users who are intend to the 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 23 Skype User Configuration 1
Figure 24 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>
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<td>SIP</td>
<td>Session Initiation Protocol</td>
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</tbody>
</table>

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