Integrating Microsoft Skype for Business Server [v6.0.9319.0] with Windstream SIP Trunk via Cisco Unified Border Element v12.0 [IOS-XE 16.06.02]

May 9, 2018
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Introduction

Service Providers today, such as Windstream Communications, 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.

Windstream 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 (SfB) and Windstream network, Cisco Unified Border Element (v12.0.0) ISR 4321/K9 running IOS-XE 16.6.2 can be used. The Cisco Unified Border Element 16.6.2 provides demarcation, security, interworking and session control services for Microsoft Skype for Business Server 2015 connected to Windstream 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 Windstream 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 Microsoft Skype for Business Server 2015, and Cisco UBE (v12.0.0) on ISR 4321/K9 [IOS-XE – 16.6.2] for connectivity to Windstream SIP Trunking service. The deployment model covered in this application note is CPE (Microsoft Skype for Business Server 2015) to PSTN (Windstream) via Cisco UBE v12.0 [IOS-XE] 16.6.2.

- 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 UBE configuration detailed in this document is based on a lab environment with a simple dial-plan used to ensure proper interoperability between Windstream SIP network and Cisco Unified Communications. 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 Skype for Business Server to interoperate to Windstream SIP Trunking network.
Network Topology

Figure 1 Network Topology

- The network topology includes the Skype for Business Server 2015 Standard Edition, Exchange Server integrated, Cisco Fax gateway and 2 SfB Clients. Cisco UBE is added as a PSTN gateway in the Skype for Business Server topology using its FQDN. Windstream Communications is used as the service provider with SIP trunk to the Cisco UBE using the WAN Virtual IP Address.
- 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.
- SIP Trunk transport type used between Cisco Unified Border Element and Windstream is UDP with RTP.
- Early media support with PRACK is enabled at Cisco UBE.

Skype for Business Server 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>
</tbody>
</table>
System Components

Hardware Requirements
- Cisco UBE on Cisco ISR 4321 router
- Generic Server for Skype for Business
- Cisco 2901 with FXS ports and Analog Fax machine

Software Requirements
- CUBE-Version: 12.0 running IOS-XE 16.6.2
- Microsoft Skype for Business Server 2015- Version: 6.0.9319.281
- Microsoft Skype for Business Client – Version 15.0.4893.1000
- Cisco IOS v15.7.3 for the fax gateway

Features

Features Supported
- Incoming and outgoing off-net calls using G711ulaw
- International Calls and digit manipulations
- Call Conference
- CPE Voice Mail support
- Call hold & Resume with and without MoH
- Unattended and Attended Call transfer
- DTMF (RFC2833)
- Fax (G711 Pass-through)
- IP-PBX Calling number privacy
- High Availability
- Early Media Support

Features Not Supported
- G729 codec is not Supported by Skype for Business
- T.38 Fax is not Supported by Skype for Business
- Skype for Business does not support Blind Call transfer
- Skype for Business does not support Call forward on Busy
- Windstream does not support Fax at Super G3 Speed
Caveats

- For basic inbound call from PSTN to SfB user, CUBE receives forked 183 response with SDP from SfB since media bypass is enabled at Skype server. CUBE handles forked 183 responses by sending UPDATE/SDP on the inbound call leg towards PSTN to renegotiate the media offer.
- Loop back calls via trunk are not successful. When call is being answered by SfB user2, originator’s (SfB user1) call state is in calling state. CUBE receives forked 183 w/SDP responses in the fourth call leg from SfB user2 (Outbound from CUBE to SfB). CUBE sends UPDATE w/SDP to SfB user1 in the first call leg (Inbound to CUBE from SfB). SfB rejects UPDATE w/SDP with ‘491 call does not exist’ since SfB does not support UPDATE w/SDP causing the first call to be in incomplete state and the call cannot be established.
- Call Forward test cases (PSTN user1 to PBX user who forwards the call to PSTN user2) are not successful. When call is being forwarded from Skype for Business user to PSTN user2, originator (PSTN user1) hears garbled audio. The issue is confirmed to be a limitation with CUBE for handling SRTP to RTP media interworking when 183 w/SDP signaling forking is involved. A future IOS software update will resolve the issue.
- Skype for Business does not support G729 voice codec. All testing is performed with g711ulaw codec.
- Ring back tone is not heard in off-net phone when the Skype user transfers the call.
- Caller ID is not updated on attended and unattended transfer scenarios.
- Only one IP PBX used for the testing.
- The Cisco UBE HA tested is layer 2 box to box redundancy.
- In HA Redundancy mode, the Primary Cisco UBE will not take over the Primary/Active role after a reboot/network outage.
- Media Bypass is enabled on the trunk between Skype for Business and CUBE. So the media flows directly from Skype clients to CUBE and vice versa.
- RTCP packets are not generated by the CUBE towards Skype for Business unless ITSP sends it to CUBE.
- Skype for Business requires 3rd party FAX for FAX support. Fax clients connected to a Cisco voice gateway are used for the test.
- Skype for Business does not support Tone on Hold.
- CUBE requires an additional dial-peer to handle REFER sent by Skype for Business.
- Work around has been done in CUBE for outbound private call to manipulate P-Asserted id (PAI) with ITSP pilot number. Skype sends privacy=id with PAI header, which has no user part present in. Therefore, CUBE restricts the caller details in the outgoing requests towards ITSP. Hence no caller information is available in incoming requests causing Windstream to reject the call.
- 0+10 digit calls reaches at Windstream Operator Services.
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
Cisco UBE 1:

interface GigabitEthernet0/0/0
 description WindstreamCube WAN
 ip address 192.xx.xx.xx 255.255.255.128
 media-type rj45
 negotiation auto
 redundancy rii 15
 redundancy group 1 ip 192.xx.xx.xx exclusive
!

interface GigabitEthernet0/0/1
 description WindstreamCube LAN
 ip address 10.80.18.48 255.255.255.0
 negotiation auto
 redundancy rii 16
 redundancy group 1 ip 10.80.18.50 exclusive
!

interface GigabitEthernet0/1/0
 description WindstreamCube HA Interface
 ip address 10.70.50.100 255.255.255.0
 negotiation auto
!

interface GigabitEthernet0
 vrf forwarding Mgmt-intf
 no ip address
 shutdown
 negotiation auto
!
Cisco UBE 2:

interface GigabitEthernet0/0/0
  description WindstreamCube WAN
  ip address 192.xx.xx.xx 255.255.255.128
  media-type rj45
  negotiation auto
  redundancy rii 15
  redundancy group 1 ip 192.xx.xx.xx exclusive

interface GigabitEthernet0/0/1
  description WindstreamCube LAN
  ip address 10.80.18.49 255.255.255.0
  negotiation auto
  redundancy rii 16
  redundancy group 1 ip 10.80.18.50 exclusive

interface GigabitEthernet0/1/0
  description WindstreamCube HA Interface
  ip address 10.70.50.110 255.255.255.0
  negotiation auto

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

!
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
  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 GigabitEthernet0/0/0
    bind media source-interface GigabitEthernet0/0/0
    session refresh
    privacy pstn
    conn-reuse
    midcall-signaling passthru
```

**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>no supplementary-service sip refer</td>
<td>Prevents the router from forwarding REFER message to the destination for call transfers. The router instead attempts to initiate a hairpin call to the new target.</td>
</tr>
</tbody>
</table>
Codecs

G711ulaw codec is used in this testing, since Skype for Business does not support G729 codec.

voice class codec 1
  codec preference 1 g711ulaw
  codec preference 2 g711alaw

Dial peer

Dial-peers to Skype for Business using TLS with SRTP:

dial-peer voice 500 voip
  description ** Outbound Call From SFB to PSTN - at CUBE LAN interface **
  session protocol sipv2
  session transport tcp tls
  incoming uri via sfb
  voice-class codec 1
  voice-class sip localhost dns:isr4k.sfblabsm.local:5061
  voice-class sip asserted-id pai
  voice-class sip call-route url
  voice-class sip copy-list 1
  voice-class sip bind control source-interface GigabitEthernet0/0/1
  voice-class sip bind media source-interface GigabitEthernet0/0/1
  voice-class sip refer-to-passing
dtmf-relay rtp-n-te
  srtp
  no vad
!

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dial-peer voice 610 voip

description ** Inbound Call From PSTN to SFB - at CUBE LAN interface **

translation-profile outgoing E164dialing

destination-pattern 469341....

session protocol sipv2

session target dns:lync.sfblabsm.local:5067

session transport tcp tls

voice-class codec 1

voice-class sip localhost dns:isr4k.sfblabsm.local:5061

voice-class sip asserted-id pai

voice-class sip call-route url

voice-class sip options-ping 60

voice-class sip profiles 103

voice-class sip options-keepalive

voice-class sip bind control source-interface GigabitEthernet0/0/1

voice-class sip bind media source-interface GigabitEthernet0/0/1

voice-class sip referto passing

dtmf-relay rtp-nte

srtp

no vad

!
dial-peer voice 700 voip
  description ** For REFER handling - at CUBE LAN interface **
  translation-profile outgoing E164dialing
  session protocol sipv2
  session target dns:lync.sfblabsm.local:5067
  session transport tcp tls
  destination uri sfbfqdn
  voice-class codec 1
  voice-class sip localhost dns:isr4k.sfblabsm.local:5061
  voice-class sip bind control source-interface GigabitEthernet0/0/1
  voice-class sip bind media source-interface GigabitEthernet0/0/1
  dtmf-relay rtp-nte
  rtcp keepalive
  srtp
  no vad
  !
Dial-peers to Windstream using UDP and RTP:

dial-peer voice 510 voip
description ** Outbound Call From SFB to PSTN - at CUBE WAN interface **
translation-profile outgoing non-e164pstncall
destination-pattern .T
session protocol sipv2
session target ipv4:64.xx.xx.xx
session transport udp
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip options-ping 60
voice-class sip profiles 104
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nte
no vad
!
dial-peer voice 600 voip
description ** Inbound Call FROM PSTN to SFB - at CUBE WAN interface **
session protocol sipv2
session transport udp
incoming called-number 469341....
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip options-ping 60
voice-class sip profiles 102
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nte
no vad
!
dial-peer voice 710 voip
  description ** Loop Back Call via Trunk - at CUBE WAN interface **
  translation-profile outgoing non-e164pstncall
  huntstop
  destination-pattern 1469341....
  session protocol sipv2
  session target ipv4:64.XX.XX.XX
  session transport udp
  voice-class codec 1
  voice-class sip asserted-id pai
  voice-class sip profiles 104
  voice-class sip options-keepalive
  voice-class sip bind control source-interface GigabitEthernet0/0/0
  voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nte
no vad
!
**Configuration example**

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

*Active Cisco UBE:*

```
windstreamCube1#sh run
Building configuration...

version 16.6
service timestamps debug datetime msec
service timestamps log datetime msec
platform qfp utilization monitor load 80
no platform punt-keepalive disable-kernel-core
!
hostname WindstreamCube1
!
boot-start-marker
boot system flash isr4300-universalk9.16.06.02.SPA.bin
boot-end-marker
!
vrf definition Mgmt-intf
!
address-family ipv4
exit-address-family
!
address-family ipv6
exit-address-family
!
no logging queue-limit
```
logging buffered 10000000
no logging rate-limit
no logging monitor
enable secret 5 $1$9JRy$HYSGC.AnbGxxxx
!
aaa new-model
!
aaa session-id common
!
ip host lync.sfblabsm.local 172.16.29.48
ip name-server 172.16.29.47
!
subscriber templating
!
multilink bundle-name authenticated
!
crypto pki trustpoint sfbc
    enrollment terminal
    fqdn isr4k.sfblabsm.local
    subject-name CN=isr4k.sfblabsm.local
    revocation-check none
    rsakeypair isr4k
!
crypto pki certificate chain sfbc
    certificate 100000005B8D119292D66E8481000000000005B
        30820517 308203FF A0030201 02021310 0000005B 8D119292 D66E8481 00000000
        005B300D 06092A86 4886F70D 01010505 00304A31 15301306 0A099226 8993F22C
        64011916 056C6F63 616C3118 30160609 09922689 93F22C64 01191608 7366626C
        6162736D 31173015 06035504 03130E73 66626C61 62736D2D 44432D43 41301E17
0FCB4BF0 3AE90D2D FDA9E232 9380D7D4 398849D5 620708A7 DAD6A694 0D39E26C
21A91E6A 3730ABFE DE1F2054 608DD9AE E699C280 3BFFCC9B DED22AE6 205871F5
D2D66BB2 001B778E 1051F6AF A528E9B5 922502DD BAD9C937 E888D62D F863F633
80E1F489 75DD553F D36293C0 3B0C59B7 7AA5F852 F9B57CFC 28048E47 DB2AB8FF
86AAD514 DB8E6F66 F7382355 78EDA6B1 34928C97 C9E9A758 D5C91764 603841D7
B9EB9B0F 4512C175 1BE57480 DED66027 C90D1EAD 93E2CE7D 4F0EE3E2 A30DF2D5
0327B9CA 280912C7 7E118A0C 7A87718C 11065D2C A0A3FD13 D21E20DB C45B3272
ADF2ED01 1BF6D584 307C87F2 3BCC7D36 97E9C512 79270C4A F2459D F2459D
quit
certificate ca 67101D7CE9C812B140EFC231D1BD207A
3082036F 30820257 A0030201 00210267 101D7CE9 C812B140 EFC231D1 BD207A30
0D06092A 864886F7 0D010105 0500304A 31153013 060A0992 268993F2 2C640119
16056C6F 63616C31 18301060 0A099226 8993F22C 64011916 08736662 6C616273
6D311730 15060355 0403130E 7366626C 6162736D 2D44432D 4341301E 170D3135
30363136 30373131 34305A17 0D323030 36313630 37323133 395A304A 31153013
060A0992 268993F2 2C640119 16056C6F 63616C31 18301060 0A099226 8993F22C
64011916 08736662 6C616273 6D311730 15060355 0403130E 7366626C 6162736D
2D44432D 43413082 0122300D 06092A86 4886F70D 01010105 00038201 0F003082
010A0282 01010097 8E8BF74A 05562140 6AEDDABC 13679FF0 50BABA47 F2753452
FDACFF61 ECF08926 F9B57B2E C647671F 3669F0C2 5E31AAAD 89424E49 8DCA1A56
11D47211 AD66C0F9 F3BD5627 DB5ABD22 7BA530CD B367D88B 44680003 3A6A87DA
7F6A75C6 2CF708B0 52EE82C7 66D0A1FD 56AD5881 631862F0 38622546 0F3C0934
89AFA9FF 3ECEED21 3A9EC80B 8499E216 C68851F9 3F5B6F42 75F05F52 A4C59D5B
487A72DB B27AE5F4 DB89193A B0F707EB E7C537F7 EDD7EC69 B4D8BA64 4D7F512F
C522CFAB AC813224 4B9B11CD A516E181 29C255A7 795F6151 3269B541 CC93ED55
FC3A9698 4B9CA9F7 EBOE1035 09B0940A 53C94601 0EC478A8 B695E9DC 62A2C34E
282E0538 DE030702 03010001 A351304F 30080603 551D0F04 04030201 86300F06
03551D13 0101FF04 05300301 01FF301D 0603551D 0E041604 1493C1CD 4B1D92D2
D6B70355 1FDFC2AA 62D94FE4 30301006 0920B061 04018237 15010443 02010030
quit
!

voice service voip
no ip address trusted authenticate
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 GigabitEthernet0/0/0
bind media source-interface GigabitEthernet0/0/0
session refresh
conn-reuse
midcall-signaling passthru
pass-thru headers unsupp
!
!
voice class uri sfb sip
  host 172.16.29.48
!
voice class uri sfbfqdn sip
  host lync.sfblabsm.local

voice class codec 1
  codec preference 1 g711ulaw
  codec preference 2 g711alaw
!

voice class sip-profiles 101
  request REGISTER sip-header From modify "<sip:(.*)@(.*)>" "<sip:\1@192.XX.XX.XX;otg=SIPTBS469341XXXX>
!

voice class sip-profiles 102
  request INVITE sip-header Diversion modify "sip:\+1" "sip:
  response ANY sip-header From modify "<sip:(.*)@(.*)>" "<sip:\1;otg=SIPTBS469341XXXX>2"
  request ANY sip-header From modify "<sip:(.*)@(.*)>" "<sip:\1@192.XX.XX.XX;otg=SIPTBS469341XXXX>
!

voice class sip-profiles 103
  request ANY sip-header From modify "<sip:(.*)@(.*)>" "<sip:\1@isr4k.sfblabsm.local>
!

voice class sip-profiles 104
  request INVITE peer-header sip Referred-By copy "<sip:(.*)@" u01
  request INVITE sip-header P-Asserted-Identity modify "<(.*)@(.*)>" "<sip:\u01\2>"
  request INVITE sip-header P-Asserted-Identity modify "sip:\+1" "sip:
  request INVITE sip-header From copy "<sip:(.*)>" u02
request INVITE sip-header P-Asserted-Identity modify "<sip:@(.*)>"
"<sip://02>"

request ANY sip-header From modify "<sip:(.*)@(.*)>"
"<sip:\1@192.xx.xx.xx;otg=SIPTBS469341xxxxxx>"

response ANY sip-header From modify "<sip:(.*@.*)>"
"<sip:\1;otg=SIPTBS469341xxxxxxx2>"

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

voice class sip-copylist 1
voice translation-rule 1
voice translation-profile E164dialing

translate called 1

license udi pid ISR4321/K9 sn FDO1922OMQ8
diagnostic bootup level minimal
spanning-tree extend system-id

username cisco password 0 *****
redundancy
mode none
application redundancy
group 1
  name b2bHAwindstream
  priority 100 failover threshold 75
  timers delay 30 reload 60
  control GigabitEthernet0/1/0 protocol 1
  data GigabitEthernet0/1/0
  track 1 shutdown
  track 2 shutdown
!
track 1 interface GigabitEthernet0/0/0 line-protocol
track 2 interface GigabitEthernet0/0/1 line-protocol
!
interface GigabitEthernet0/0/0
  description WindstreamCube WAN
  ip address 192.XX.XX.XX 255.255.255.128
  media-type rj45
  negotiation auto
  redundancy rii 15
  redundancy group 1 ip 192.XX.XX.XX exclusive
!
interface GigabitEthernet0/0/1
  description WindstreamCube LAN
  ip address 10.80.18.48 255.255.255.0
  negotiation auto
  redundancy rii 16
  redundancy group 1 ip 10.80.18.50 exclusive
interface GigabitEthernet0/1/0

description WindstreamCube HA Interface

ip address 10.70.50.100 255.255.255.0

negotiation auto

interface GigabitEthernet0

vrf forwarding Mgmt-intf

no ip address

shutdown

negotiation auto

ip forward-protocol nd

no ip http server

no ip http secure-server

ip route 0.0.0.0 0.0.0.0 192.xx.xx.xx

ip route 10.64.0.0 255.255.0.0 10.80.18.1

ip route 10.70.50.0 255.255.255.0 10.80.18.1

ip route 172.16.24.0 255.255.248.0 10.80.18.1

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 500 voip
description ** Outbound Call From SFB to PSTN - at CUBE LAN interface **
session protocol sipv2
session transport tcp tls
incoming uri via sfb
voice-class codec 1
voice-class sip localhost dns:isr4k.sfblabsm.local:5061
voice-class sip asserted-id pai
voice-class sip call-route url
voice-class sip copy-list 1
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
voice-class sip referto-passing
dtmf-relay rtp-nte
srtp
no vad
dial-peer voice 510 voip
description ** Outbound Call From SFB to PSTN – at CUBE WAN interface **
translation-profile outgoing non-e164pstncall
destination-pattern .T
session protocol sipv2
session target ipv4:64.xx.xx.xx
session transport udp
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip options-ping 60
voice-class sip profiles 104
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-n-te
no vad
!
dial-peer voice 600 voip
description ** Inbound Call FROM PSTN to SFB – at CUBE WAN interface **
session protocol sipv2
session transport udp
incoming called-number 469341....
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip options-ping 60
voice-class sip profiles 102
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-n-te
no vad
!

dial-peer voice 610 voip
description ** Inbound Call From PSTN to SFB - at CUBE LAN interface **
translation-profile outgoing E164dialing
destination-pattern 469341...
session protocol sipv2
session target dns:lync.sfblabsm.local:5067
session transport tcp tls
voice-class codec 1
voice-class sip localhost dns:isr4k.sfblabsm.local:5061
voice-class sip asserted-id pai
voice-class sip call-route url
voice-class sip options-ping 60
voice-class sip profiles 103
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
voice-class sip refereo-passing
dtmf-relay rtp-nate
srtp
no vad
!

no vad
!
dial-peer voice 700 voip
  description ** For REFER handling - at CUBE LAN interface **
  translation-profile outgoing E164dialing
  session protocol sipv2
  session target dns:lync.sfblabsm.local:5067
  session transport tcp tls
  destination uri sfbfqdn
  voice-class codec 1
  voice-class sip localhost dns:isr4k.sfblabsm.local:5061
  voice-class sip bind control source-interface GigabitEthernet0/0/1
  voice-class sip bind media source-interface GigabitEthernet0/0/1
  dtmf-relay rtp-npe
  rtcp keepalive
  srtp
  no vad
!

dial-peer voice 710 voip
  description ** Loop Back Call via Trunk - at CUBE WAN interface **
  translation-profile outgoing non-e164pstncall
  huntstop
  destination-pattern 1469341....
  session protocol sipv2
  session target ipv4:64.XX.XX.XX
  session transport udp
  voice-class codec 1
  voice-class sip asserted-id pai
  voice-class sip profiles 104
  voice-class sip options-keepalive
  voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nce
no vad
!
gateway
timer receive-rtp 1200
!
sip-ua
credentials number 469341xxxx username 469341xxxx password 7 065259781F1A5841544146 realm mcleodusa
authentication username 469341xxxx password 7 03500D5255B70141F5F4D realm mcleodusa
no remote-party-id
registrar ipv4:64.xx.xx.xx:5060 expires 60
connection-reuse
crypto signaling default trustpoint sfbca
!
line con 0
password *******
transport input none
stopbits 1
line aux 0
password *******
stopbits 1
line vty 0 4
exec-timeout 0 0
password *******
transport preferred telnet
!
wsma agent exec
wsma agent config

wsma agent filesys

wsma agent notify

End
Standby Cisco UBE:

windstreamCube2#sh run
!
version 16.6
service timestamps debug datetime msec localtime
service timestamps log datetime msec localtime
service password-encryption
service internal
service sequence-numbers
platform qfp utilization monitor load 80
no platform punt-keepalive disable-kernel-core
!
hostname windstreamCube2
!
boot-start-marker
boot system flash isr4300-universalk9.16.06.02.SPA.bin
boot-end-marker
!
vrf definition Mgmt-intf
!
address-family ipv4
exit-address-family
!
address-family ipv6
exit-address-family
!
logging queue-limit 1000000000
logging buffered 30000000
logging rate-limit 10000
no logging console
logging monitor notifications
enable secret 5 $1$aEKr$TYXs2kVsHhvpeqnj4bdYn.
!
no aaa new-model
!
ip host lync.sfblabsm.local 172.16.29.48
ip name-server 172.16.29.47
!
subscriber templating
!
multilink bundle-name authenticated
!
crypto pki trustpoint sfbca
  enrollment terminal
  fqdn isr4k.sfblabsm.local
  subject-name CN=isr4k.sfblabsm.local
  revocation-check none
  rsakeypair isr4k
!
!
crypto pki certificate chain sfbca
  certificate 100000005AF2FC9CD1FACC82030000000000005A
    30820517 308203FF A0030201 02021310 0000005A F2FC9CD1 FACC8203 00000000
    005A300D 06092A86 4886F70D 01010505 00304A31 15301306 0A099226 8993F22C
    64011916 056C6F63 616C3118 3016060A 09922689 93F22C64 01191608 7366626C
    6162756D 31173015 06035504 03130E73 66626C61 62736D2D 44432D43 41301E17
    0D313B30 31396315 35343730 325A170D 32303031 31363135 34373032 5A301F31
    1D301B06 03550403 13146973 72346B2E 7366626C 6162756D 2E6C6F63 616C3082
ECE1A73B 8F44EA4B 9D7EDB84 B BBCBBBF0 DF78D0CB AC5AE8B1C A1F45352 07C481B5 EFD36AC7 24B902F7 DCCA6103 2A53ECD8 5A10365D 23EF2CCB 43A8975A 1E0D275D 880317E7 7BE2824F 043B88A4 E37A0983 1DBFA255 F93EB613 3063BBC5 2CAC89A7 E4A74805 EF7886B5 099F8C08 452A2238 F8E0144C 09E9010B 3B2761D1 83B7992A 665C8DBB 64CF243C 7C673562 B28443B3 A13B0E83 AD5B0B28 9B6817D6 1C09ABD3 D945C521 752F236E 32B5CA7 C216BF04 6AD54907 91A4D912 11A083

quit
certificate ca 67101D7CE9C812B140EFC231D1BD207A

3082036F 30820257 A0030201 02021067 101D7CE9 C812B140 EFC231D1 BD207A30 0D06092A 864886F7 0D010105 0500304A 31153013 060A0992 268993F2 2C640119 16056C6F 63616C31 18301606 0A099226 8993F22C 64011916 08736662 6C616273 6D311730 15060355 0403130E 7366626C 6162736D 2D44432D 4341301E 170D3135 30363136 30373131 34305A17 0D323030 36313630 37323133 395A304A 31153013 060A0992 268993F2 2C640119 16056C6F 63616C31 18301606 0A099226 8993F22C 64011916 08736662 6C616273 6D311730 15060355 0403130E 7366626C 6162736D 2D44432D 43413082 0122300D 06092A86 4886F70D 01010105 00038201 0F003082 010A0282 01010097 8E8BF74A 05562140 6AEDDABC 13679FF0 50BABA47 F2753452 FDACFF61 ECF08926 F9B57BAE C647671F 3669FC0D 5E31AADA 8942E449 8DC2A156 11D47211 AD66C0F9 F3BD5627 DB5ABDB7 7BA530CD B367D88B 44680003 3A6A87DA 7F6A75C6 2CF70880 52E882C7 66D0A1F0 56AD5881 631862F0 38622546 0F3C0934 89AFA9FF 3ECEED21 3A9EC8DB 8499E216 C68851F9 3F5B6F42 75F05F52 A4C5D95B 487A72DB B27AE5F4 DB89193A B0F707EB E7C537F7 EDD7EC69 B4D8BA64 4D7F512F C522CFAB AC813224 489B11CD A516E181 29C255A7 795F6151 3269B541 CC93E655 FC3A9698 4BCAE97F EB0E1035 09B0940A 53C94601 0EC47A8A B695E9DC 62A23C4E 282E0538 DE030702 03010001 A351304F 300B0603 551D0F04 04030201 86300F06 03551D13 0101FF04 05300301 01FF301D 0603551D 0E041604 1493C1CD 4B1D9D22 D6B70355 1FDFC2AA 62D94FE4 30301006 092B0601 04018237 15010403 2010030 0D06092A 864886F7 0D010105 05000382 01010064 3935BF00 067E8C7C FC8E0076 77D0FB8C 0846697C 3A1A39A0 3A3387C5 18D4D9D3 596F540D 659F27A2 26063A21
quit
!
!
voice service voip
   no ip address trusted authenticate
   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 GigabitEthernet0/0/0
      bind media source-interface GigabitEthernet0/0/0
   session refresh
   privacy pstn
   conn-reuse
   midcall-signaling passthru
   pass-thru headers unsupp
!
!
voice class uri sfb sip
  host 172.16.29.48

voice class uri sfbfqdn sip
  host lync.sfblabsm.local

voice class codec 1
  codec preference 1 g711ulaw
  codec preference 2 g711alaw

voice class sip-profiles 101
  request REGISTER sip-header From modify "<sip:(.*)@(.*)>"
  "<sip:\l@192.XX.XXXX;otg=SIPTBS469341XXXX>"

voice class sip-profiles 102
  request INVITE sip-header Diversion modify "sip:\+1" "sip:"
  response ANY sip-header From modify "<sip:(.*)@(.*)>">
  "<sip:\l@192.XX.XXXX;otg=SIPTBS469341XXXX>"

voice class sip-profiles 103
  request ANY sip-header From modify "<sip:(.*)@(.*)>"
  "<sip:\l@isr4k.sfblabsm.local>"

voice class sip-profiles 104
  request INVITE peer-header sip Referred-By copy "<sip:(.*)@ u01
  request INVITE sip-header P-Asserted-Identity modify "<(.*)@(.*)>"
  "<sip:\u01@
  request INVITE sip-header P-Asserted-Identity modify "sip:\+1" "sip:"
  request INVITE sip-header From copy "<sip:(.*)@ u02
request INVITE sip-header P-Asserted-Identity modify "<sip:@(.*>)""<sip:u02>"

response ANY sip-header From modify "<sip:(.*@.*)(>. withhold"
"<sip:\1.otg=SIPTBS469341XXXX\2"

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

request ANY sip-header From modify "<sip:(.*@.*)(.*)@.*)>" "<sip:\1@192.XX.XX.XX;otg=SIPTBS469341XXXX>

request INVITE sip-header P-Asserted-Identity modify "<sip:anonymous@(.*)>" "<sip:469341XXXX@192.XX.XX.XX>"

!
voice class sip-copylist 1

sip-header REFERRED-BY

!

voice translation-rule 1

rule 1 /469341xxxx/ /xxxx/

rule 2 /\(^{.........}$\)/ /+1\1/

!

voice translation-rule 2

rule 1 /\+1\(.........\)$/ /\1/

!

voice translation-profile E164dialing

translate called 1

!

voice translation-profile non-e164pstncall

translate calling 2

!

license udi pid ISR4321/K9 sn FDO19220MW3
license boot level appxk9
license boot level uck9
diagnostic bootup level minimal
spanning-tree extend system-id
!
username cisco privilege 15 password 7 13111xxx97B7B2A
!
redundancy
mode none
application redundancy
group 1
  name b2bHAwindstream
  priority 150 failover threshold 75
timers delay 30 reload 60
  control GigabitEthernet0/1/0 protocol 1
data GigabitEthernet0/1/0
  track 1 shutdown
  track 2 shutdown
!
track 1 interface GigabitEthernet0/0/0 line-protocol
track 2 interface GigabitEthernet0/0/1 line-protocol
!
interface GigabitEthernet0/0/0
  description WindstreamCube WAN
  ip address 192.xx.xx.xx 255.255.255.128
  media-type rj45
  negotiation auto
  redundancy rii 15
  redundancy group 1 ip 192.xx.xx.xx exclusive
!
interface GigabitEthernet0/0/1
description WindstreamCube LAN
ip address 10.80.18.49 255.255.255.0
negotiation auto
redundancy rii 16
redundancy group 1 ip 10.80.18.50 exclusive
!
interface GigabitEthernet0/1/0
description WindstreamCube HA Interface
ip address 10.70.50.110 255.255.255.0
negotiation auto
!
interface GigabitEthernet0
vrf forwarding Mgmt-intf
no ip address
shutdown
negotiation auto
!
ip forward-protocol nd
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.xx.xx.xx
ip route 10.64.0.0 255.255.0.0 10.80.18.1
ip route 10.70.50.0 255.255.255.0 10.80.18.1
ip route 172.16.24.0 255.255.248.0 10.80.18.1
!
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
!
telephony-service
  max-conferences 8 gain -6
  transfer-system full-consult
!
!
dial-peer voice 500 voip
description ** Outbound Call From SFB to PSTN - at CUBE LAN interface **
session protocol sipv2
session transport tcp tls
incoming uri via sfb
voice-class codec 1
voice-class sip localhost dns:isr4k.sfblabsm.local:5061
voice-class sip asserted-id pai
voice-class sip call-route url
voice-class sip copy-list 1
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
voice-class sip referto-passing
dtmf-relay rtp-nnte
srtp
no vad
!
dial-peer voice 510 voip

description ** Outbound Call From SFB to PSTN - at CUBE WAN interface **
translation-profile outgoing non-e164pstncall
destination-pattern .T
session protocol sipv2
session target ipv4:64.xx.xx.xx
session transport udp
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip options-ping 60
voice-class sip profiles 104
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nte
fax-relay ecm disable
fax protocol pass-through g711ulaw
no vad
!
dial-peer voice 600 voip

description ** Inbound Call FROM PSTN to SFB - at CUBE WAN interface **
session protocol sipv2
session transport udp
incoming called-number 469341....
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip options-ping 60
voice-class sip profiles 102
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-n-te

no vad

!

dial-peer voice 610 voip
description ** Inbound Call From PSTN to SFB - at CUBE LAN interface **
translation-profile outgoing E164dialing
destination-pattern 469341....
session protocol sipv2
session target dns:lync.sfblabsm.local:5067
session transport tcp tls
voice-class codec 1
voice-class sip localhost dns:isr4k.sfblabsm.local:5061
voice-class sip asserted-id pai
voice-class sip call-route url
voice-class sip options-ping 60
voice-class sip profiles 103
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
voice-class sip referto-passing
dtmf-relay rtp-n-te
srtp

no vad

!

dial-peer voice 700 voip
description ** For REFER handling - at CUBE LAN interface **
translation-profile outgoing E164dialing
session protocol sipv2
session target dns:lync.sfblabsm.local:5067
session transport tcp tls
destination uri sfbfqdn
voice-class codec 1
voice-class sip localhost dns:isr4k.sfblabsm.local:5061
voice-class sip bind control source-interface GigabitEthernet0/0/1
voice-class sip bind media source-interface GigabitEthernet0/0/1
dtmf-relay rtp-nfe
rtcp keepalive
srtp
no vad
!
dial-peer voice 710 voip
description ** Loop Back Call via Trunk - at CUBE WAN interface **
translation-profile outgoing non-e164pstncall
huntstop
destination-pattern 1469341....
session protocol sipv2
session target ipv4:64.XX.XX.XX
session transport udp
voice-class codec 1
voice-class sip asserted-id pai
voice-class sip profiles 104
voice-class sip options-keepalive
voice-class sip bind control source-interface GigabitEthernet0/0/0
voice-class sip bind media source-interface GigabitEthernet0/0/0
dtmf-relay rtp-nfe
no vad
!
!
gateway
timer receive-rtp 1200
!
sip-ua
credentials number 469341xxxx username 469341xxxx password 7
124d534e415f5d5c7b7d70 realm mcleodusa
authentication username 469341xxxx password 7 124d534e415f5d5c7b7d70 realm mcleodusa
no remote-party-id
registrar ipv4:64.xx.xx.xx:5060 expires 60
connection-reuse
crypto signaling default trustpoint sfbca
!
!
line con 0
exec-timeout 0 0
login local
transport input none
stopbits 1
line aux 0
stopbits 1
line vty 0 4
exec-timeout 0 0
login local
transport input telnet
!
wsma agent exec
!
wsma agent config
!
wsma agent filesys
!
wsma agent notify
!
end
Configuring Microsoft Skype for Business Server

This section describes the necessary steps to configure SIP Trunk from Skype for Business Server to CUBE LAN interface.

**PSTN Gateway Configuration**

- Open Skype for Business Server 2015 Topology Builder and navigate to Shared Components.
- Right-click PSTN Gateways and from the pop-up menu, choose New IP/PSTN Gateway.

![Figure 3: Add new IP/PSTN Gateway in Skype for Business Topology Builder](image-url)

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• Configure PSTN Gateway with CUBE FQDN

Figure 4: Define the PSTN gateway
- Select “Enable IPv4” with use all configured IP Address.

![Define New IP/PSTN Gateway](image)

Figure 5: Define IP address

- Define trunk name and specify the Listening port for IP/PSTN gateway
Select the SIP Transport Protocol and Associated Mediation Server with its listening port as shown below and Click Finish.

Figure 6: Define the Root Trunk details
Newly configured “PSTN gateway” and “Trunk” will appear in the Topology Builder as shown below.

Figure 7: Configured PSTN Gateway to CUBE
For voice mail and fax support, two other PSTN gateways has configured in Topology Builder towards
Exchange Server and to Fax gateway.

Figure 8: Configured PSTN Gateways in SfB Topology Builder

- Next publish the Topology by selecting “Action → Topology → Publish...”
Figure 9: Publishing the Topology
Figure 10: Changes to the Topology published successfully
Voice Routing Configuration

This section describes the Route and Trunk configuration in Skype for Business Server and associating it with the IP/PSTN gateway (configured in Topology Builder).

- Open Skype for Business Server 2015 Control Panel. In the left navigation bar, click “Voice Routing” and then click “Dial Plan”.

![Figure 11: Voice Routing Menu](image)

- From the “Dial Plan” page, Click on “New” and select “Pool Dial Plan”.

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Figure 12: Adding New Dial Plan

- Select “Pool” for dial plan scope from “Select Service” dialog box. Here Registrar is selected.
Figure 13: Select the service for Dial Plan

- Specify the Name and configure required normalization rules for the dial plan.
On the “Dial Plan” page, click “Commit”, and then click “Commit all”.

Figure 14: Dial Plan with added Normalization Rule
Navigate to “Voice Policy” tab, click on “New” and select “User Policy” to add the new user Voice Policy as shown below.

Figure 15: Uncommitted Dial Plan

Figure 16: Committed Dial Plan Configuration successfully
Figure 17: Voice policy Window

- Specify name for “Voice Policy”, description (optional) and enable required features then scroll down to “Associated PSTN Usages”. Click on “New” in the Associated PSTN Usage box.
Figure 18: Adding a New Voice Policy
Figure 19: Associated PSTN Usages

- Specify the name for new “PSTN Usage” and click on “New” to add a new Route.
- Add a new route for 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 trunk to associate with newly created route.
Figure 21: New Voice Route details and Add Trunk to associate
Double click on the appropriate route (or) Click “OK” after selecting a route. Selected route will displayed as shown below. Then click on “OK” to apply the newly created PSTN Usage. This will roll back to the ‘Voice Policy’ page with the newly added PSTN Usage and Route.

![Select Trunk to associate with Route](image)

**Figure 22:** Select Trunk to associate with Route

Click “OK” to apply the new Voice Policy settings. Now the “Voice Policy” page will be shown with the newly added voice policy as “uncommitted”.

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Figure 23: Voice Policy configuration with trunk associated
Click on “Commit” and select the “Commit All” menu to commit the changes made.
Figure 25: Committed Voice Policy 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

![Skype for Business Server 2015 Control Panel](image)

Figure 26: New Pool Trunk

- Configure the trunk with the required parameters selected as shown below.
Figure 27: SIP Trunk to Cisco UBE
Figure 28: Uncommitted Trunk Configuration
• Repeat the above procedure for the trunks towards Fax gateway and to Exchange Server. Configured Trunk is as follows.
Figure 30: Trunk towards Fax Gateway
Figure 31: Trunk to Exchange Server
User Configuration

- Select the Skype Users who are intend to use the Enterprise Voice feature and fill the Windstream DID in the Line URI field. Select the newly added User Voice Policy in “Voice Policy” field.

Figure 32: Skype for Business User Configuration
Configuring Cisco Voice Gateway for Fax

Global Settings

```
voice service voip
  no ip address trusted authenticate
  address-hiding
  allow-connections sip to sip
  redirect ip2ip
  fax protocol pass-through g711ulaw
  no fax-relay sg3-to-g3
  sip
    rel1xx disable
    conn-reuse
    early-offer forced
    midcall-signaling passthru

!```

Codecs

```
G711ulaw is used for this testing.
voice class codec 1
  codec preference 1 g711ulaw
```
Dial peer

Outbound Dial-peer to Skype for Business:

dial-peer voice 5001 voip
description ** Windstream Outbound FAX **
translation-profile outgoing E164dialing
destination-pattern 9722657262
session protocol sipv2
session target dns:lync.sfblabsm.local:5067
session transport tcp tls
  voice-class sip srtp-auth sha1-80
voice-class sip localhost dns:gateway.sfblabsm.local:5061
voice-class sip options-keepalive
dtmf-relay rtp-nte
srtp
codec g711ulaw
fax-relay ecm disable
no vad
!

Inbound Dial-peer from Skype for Business:

dial-peer voice 5000 voip
description ** Windstream Inbound FAX **
session protocol sipv2
session transport tcp tls
incoming called-number +1469341xxxx
incoming uri to sfb
  voice-class sip srtp-auth sha1-80
dtmf-relay rtp-nte
srtp
codec g711ulaw
fax-relay ecm disable
no vad
POTS and Port Configuration:

Based on configured POTS destination pattern, gateway forwards the call to designated voice port.

dial-peer voice 50 pots
    service session
    destination-pattern +1469341XXXX
    no digit-strip
    port 0/0/1
    forward-digits all
    
voice-port 0/0/1
    no vad
    cptone IN
    station-id name WindstreamFax
    station-id number +1469341XXXX
    caller-id enable
    
!
Configuration example

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

User Access Verification

Username: cisco
Password:
FAX-GATEWAY1#sh run

version 15.7
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname FAX-GATEWAY1
!
boot-start-marker
boot system flash:c2900-universalk9-mz.SPA.157-3.M.bin
boot-end-marker
!
aqm-register-fnf
!
logging queue-limit 1000000000
logging buffered 10000000
logging rate-limit 10000
no logging console
enable secret 4 xxxxxx1Qcm7yYduKGZI
!
no aaa new-model
!
!
ip host lync.sfblabsm.local 172.16.29.48
ip name-server 172.16.29.47
ip cef
no ipv6 cef

multilink bundle-name authenticated

!  
stcapp feature access-code
!  
stcapp feature speed-dial
!  
ccts logging verbose
!
crypto pki trustpoint skypelab
    enrollment terminal
    fqdn faxgateway.sfblabsm.local
    subject-name CN=faxgateway.sfblabsm.local
    revocation-check none
    rsakeypair faxtest
!
crypto pki certificate chain skypelab
    certificate 10000000870DA687FA273CDCF6000000000087
        3082051C 30820404 A0030201 02021310 00000087 0DA687FA 273CDCF6 00000000
        0087300D 06092A86 4886F70D 01010505 00304A31 15301306 0A099226 8993F22C
        64011916 056C6F63 616C3118 3016060A 09922689 93F22C64 01191608 7366626C
        6162736D 31173015 06035504 03130E73 66626C61 62736D2D 44432D43 41301E17
        0D313830 33323131 34313431 335A170D 32303033 32303134 31343133 5A302431
        22302006 03550403 13196661 78676174 65776179 2E736662 6C616273 6D2E6C6F
        63616C30 82012230 0D06092A 864886F7 0D010101 05000382 010F0030 82010A02
        82010100 A49A3134 5E08009B EE26CDAF 7FF649EC C1D8AEC4 10AF2E82 E1E074CE
quit
certificate ca 67101D7CE9C812B140EFC231D1BD207A
3082036F 30820257 A0030201 02021067 101D7CE9 C812B140 EFC231D1 BD207A30
0D06092A 8648886F7 0D010105 0500304A 31153013 060A0992 268993F2 2C640119
16056C6F 63616C31 18301606 0A099226 8993F22C 64011916 08736662 6C616273
6D311730 15060355 0403130E 7366626C 6162736D 2D44432D 4341301E 170D3135
30363136 30373131 34305A17 0D323030 36313630 37323133 395A304A 31153013
060A0992 268993F2 2C640119 16056C6F 63616C31 18301606 0A099226 8993F22C
64011916 08736662 6C616273 6D311730 15060355 0403130E 7366626C 6162736D
2D44432D 43413082 0122300D 06092A86 4886F70D 01010105 00038201 0F003082
010A0282 01010097 8E8BF74A 05562140 6AEDDABC 13679FF0 50BABA47 F2753452
FDACFF61 ECF08926 F9B57BAE C647671F 3669FCD2 5E31AADA 8942E449 8DCA2156
11D47211 AD66C0F9 F3BD5627 DB5AB22 7BA530CD B367D88B 44680003 3A6A87DA
7F6A75C6 2CF708B0 52EE82C7 66DA01FD 56AD5881 631862F0 38622546 0F3C0934
89AFA9FF 3ECED21 3A9EC8DB 8499E216 C68851F9 3F5B6F42 75F05F52 A4C5D95B
487A72DB B27AE5F4 DB89193A B0F707EB E7C537F7 EDD7EC69 B4D8BA64 4D7F512F
C522CFAB AC813224 48B911CD A516E181 29C255A7 795F6151 3269B541 CC93E655
FC3A9698 4Bcae97F EB0E1035 09B0940A 53C94601 0EC47A8A B695E9DC 62A23C4E
2B2E0538 DE030702 03010001 A351304F 300B0603 551D0F04 04030201 86300F06
03551D13 0101FF04 05300301 01FF301D 0603551D 0E041604 14931CD 4B19D22
D6B70355 1FDFC2AA 6D94FE4 D3030106 092B0601 02000037 15010403 20100302
0D06092A 8648886F7 0D010105 0500304A 31153013 060A0992 268993F2 2C640119
16056C6F 63616C31 18301606 0A099226 8993F22C 64011916 08736662 6C616273
6D311730 15060355 0403130E 7366626C 6162736D 2D44432D 4341301E 170D3135
30363136 30373131 34305A17 0D323030 36313630 37323133 395A304A 31153013
060A0992 268993F2 2C640119 16056C6F 63616C31 18301606 0A099226 8993F22C
64011916 08736662 6C616273 6D311730 15060355 0403130E 7366626C 6162736D
quit
voice-card 0
codec complexity medium
voice service voip
  no ip address trusted authenticate
  address-hiding
  allow-connections sip to sip
  redirect ip2ip
  fax protocol pass-through g711ulaw
  no fax-relay sg3-to-g3
  sip
    rel1xx disable
    conn-reuse
    early-offer forced
    midcall-signaling passthru
!

voice class uri sfb sip
  host 10.80.22.7
voice class codec 1
  codec preference 1 g711ulaw
!

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

voice translation-profile E164dialing
  translate called 1
!

license udi pid CISCO2901/K9 sn FTX174081SJ
license boot module c2900 technology-package securityk9
hw-module pvdm 0/0
username <myuser> privilege 15 secret 5 $1$Afj1$f7FNxxxxxdttg76H4G1.
username cisco privilege 15 secret 4 xxxx6S2i4ntXrpb4RFmfqY

!  
interface Embedded-Service-Engine0/0
   no ip address
   shutdown
!  
interface GigabitEthernet0/0
   ip address 10.80.22.7 255.255.255.0
   duplex half
   speed auto
!
interface GigabitEthernet0/1
   no ip address
   shutdown
   duplex auto
   speed auto
!
ip forward-protocol nd
!

ip http server
no ip http secure-server
ip http timeout-policy idle 60 life 86400 requests 10000
!
ip route 0.0.0.0 0.0.0.0 10.80.22.1
ip ssh server algorithm encryption aes128-ctr aes192-ctr aes256-ctr
ip ssh client algorithm encryption aes128-ctr aes192-ctr aes256-ctr
!
control-plane
!
voice-port 0/0/0
   no vad
   cptone IN
voice-port 0/0/1
no vad
cптone IN
station-id name WindstreamFax
station-id number +1469341XXXX
caller-id enable
!
mgcp profile default
!
dial-peer voice 50 pots
  service session
destination-pattern +1469341XXXX
  no digit-strip
  port 0/0/1
  forward-digits all
!
dial-peer voice 5000 voip
  description ** Windstream Inbound FAX **
  session protocol sipv2
  session transport tcp tls
  incoming called-number +1469341XXXX
  incoming uri to sfb
  voice-class sip srtp-auth sha1-80
dtmf-relay rtp-nte
srtp
codec g711ulaw
fax-relay ecm disable
  no vad
!
dial-peer voice 5001 voip
  description ** Windstream Outbound FAX **
  translation-profile outgoing E164dialing
destination-pattern 9722657262
session protocol sipv2
session target dns:lync.sfblabsm.local:5067
session transport tcp tls
  voice-class sip srtp-auth shal-80
voice-class sip localhost dns:gateway.sfblabsm.local:5061
voice-class sip options-keepalive
dtmf-relay rtp-nte
srtp
codec g711ulaw
fax-relay ecm disable
no vad
!
!
gateway
timer receive-rtp 1200
!
sip-ua
crypto signaling default trustpoint skypelab
!
gatekeeper
  shutdown
!
credentials
!
line con 0
  login local
line aux 0
line 2
  no activation-character
  no exec
  transport preferred none
transport output pad telnet rlogin lapb-ta mop udptn v120 ssh
stopbits 1
line vty 0 4
  exec-timeout 0 0
  logging synchronous
  login local
  transport input telnet ssh
line vty 5 15
  login local
  transport input telnet ssh
!
scheduler allocate 20000 1000
!
end
### Acronyms

<table>
<thead>
<tr>
<th>Acronym</th>
<th>Definitions</th>
</tr>
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<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>SfB</td>
<td>Skype for Business</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>SIP</td>
<td>Session Initiation Protocol</td>
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