Cisco S.E. Webex Event Series
Smart Business Communication System (SBCS)
SIP Trunking
Making the Most of VoIP in the IP-PBX

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Introduction

This is an application note intended to help Cisco Partners understand what a Session Initiation Protocol (SIP) Trunk is, why it is strategically important to the SMB, and how to implement a SIP Trunk on the Cisco Smart Business Communication System (SBCS) IP-Private Branch Exchange (IP-PBX). This is a topic requested by the Sales Channel and is a frequent support request case with the PDI helpdesk. This topic is being delivered as a Partner Webinar and this document covers a bit more detail than time allows in the Webinar, so it can be used as a reference.

Definition of a SIP Trunk

A SIP Trunk is a single, integrated, quality of service (QoS) enabled IP connection that a SMB will lease from a service provider network to carry all its voice, data, and video traffic.

A SIP trunk provides the SMB with WAN IP connectivity to a Service Provider (SP) who utilizes the IETF SIP signaling protocol and infrastructure elements for call control and routing of voice traffic within their network, to interconnecting VoIP networks and to the PSTN.

What a SIP trunk does is Register the SIP Endpoints assigned specific and unique E.164 telephone numbers at the customers IP-PBX with a Service Provider SIP Registrar and then allow subsequent handling of all SIP Methods utilizing the SP Proxy, Redirect, and Application servers elements. The most prevalent SIP Methods will be those associated with call (session) establishment and disestablishment.

Benefits of SIP Trunking

The motivation for a SMB to want to integrate their IP-PBX with an IP network capable of carrying VoIP calls in a reliable manner should be focused on the benefits of SIP Trunking as follows.

Reduce the total cost of ownership of the IP-PBX

- Legacy circuit switched TDM lines and trunks can be drastically reduced, eliminating separate voice (TELCO) charges to the SMB.

CAPEX + OPEX of your own IP-PBX < recurring tariff charges for hosted services.

- Including increased productivity through the implementation of IP capabilities and tools (presence based routing, unified voice mail/email, SNR …etc)

Consolidates both Voice and Data on a single IP network infrastructure

- Better use of bandwidth provided by a single SP

Extends the reach of unified VoIP communications beyond the Small Business Data LAN

It allow for integration with IP applications (rich media sharing, video …etc) to enhance customer interactions and relationships.

Direct Inward Dial (DID) telephone numbers as you require (1:1, 1:many)

- Local Number Portability included
- E911 included
- IP Bandwidth is the only limit to the number of calls it can carry on a single DID.

Lower Telecom Call Costs (negotiated rates through VoIP SP)

SMB Telecom Industry Evolution

The SIP Trunk is a new technology offered in SIP signaling enabled networks. SIP enabled IP networks came about as the result of the IP converged network, with the realization that so much content and applications are delivered on the IP stack along with voice for a unified communications experience, and Service Providers made their network infrastructure available as a commodity for interconnect to this content, application and VoIP carriage.

SIP

There are several VoIP supporting protocols besides SIP:

- ITU H.323 Family of protocols originally for Video but adapted for VoIP
- MGCP: Media Gateway Control Protocol (Smart Switch controlling dumb gateways)
- Megaco/H.248: Standard Version of MGCP

SIP was patterned after another successful Internet standard (HTTP), didn't attempt to reinvent the PSTN, did not dictate architecture or services, is non hierarchical, works on any IP network, is scalable and reliable and leverages the best of existing standards (URLs, MIME, RFC822). It is designed as a client server protocol and has peer to peer (P2P) capability since each SIP endpoint can be both client and server.

Legacy SMB Telecom

In the 1990's, if you were a small business, there were three technologies available to provide business voice telecommunications and feature functionality. All three technologies gave the SMB the ability to have capabilities needed in the SMB (music on hold, attendant console, hold and transfer, etc.) and allowed interconnection to the rest of the Public Switched Telephone Network (PSTN) when voice communications were required with parties external to the SMB.

- The Private Branch Exchange (PBX)
- The Key Telephone System
- Hosted PBX Services (as in Centrex)

The PBX used to be big Capital Expense (CAPEX) investment and also has Operational Expense (OPEX) of operating it and maintaining it, but offered the most comprehensive set of features otherwise only available per expensive monthly Tariff from the Telephone Company. For reasons of cost it was not deployed among smaller sized business as much as it was among medium to larger sized enterprises (60 – 8,000 users possible). It had a leased circuit switched trunk interface to the PSTN (usually ISDN PRI) which was shared buy all the lines in the SMB. They were usually proprietary in nature of protocol and interoperability within the PBX.

The Key telephone System was essentially a more economical and smaller premise based phone system which could be leased or owned, with individual shared line interface to the PSTN and a popular group of features that lent themselves to functions most needed at the Small business. This was usually found only in smaller businesses.

Hosted PBX was a leased *managed service* whereby your phones and wire pairs were controlled at a TELCO Central Office (CO) switch providing a bundle of business features and capabilities competitive with the PBX offerings. It offered the same robust feature set as a PBX without the ownership issues, but for a recurring cost as a Tarriffed service offering of the TELCO. This was also a line based PSTN interface and while lines could be shared, they were usually deployed 1:1 (one line to each subscriber in the SMB).

- CENTREX was the Bell Labs trademarked name for such an business line offering, first developed by N.Y. Telephone Company in the 1960s and later hosted on the 1AESS and 5ESS switches
- IBN was Nortel's version of the same type of offering on DMS100 Family C.O. Switch

The IP PBX in the Converged IP Network

The case for increased efficiency and productivity of the "SMART" IP converged network is easy to see when we acknowledge that Voice is composed of IP packets of a certain type and priority that ride a common infrastructure and can interoperate with other IP applications anywhere in the network.

The Legacy PBX experienced diminished growth even in the cases where IP adjuncts were added since this only added IP Data on an otherwise closed and proprietary system. These PBXs were still too large for the small business.

The pure IP-PBX emerged as the lower cost, smaller footprint, converged IP network solution handling unified communications capabilities in addition to legacy PBX features.

As in the case of the legacy telecom systems, an IP-PBX can be deployed a few different ways of which the most common are:

- Customer Premises based IP-PBX ← which this paper if focused on
- Pure Hosted IP-PBX (or Hosted Centrex) Service

Session Initiation Protocol (SIP) based IP-PBX systems offer the most interoperable solution in the industry, with the most advanced features being offered by IP-PBX vendors and application partners using the IETF Industry standard signaling protocol of choice for Voice over IP systems (VoIP).

Many small business have migrated to the IP-PBX and are operating VoIP with unified communications within the boundaries of their campus LAN and are now starting to consider the full SIP Trunk interconnect to the rest of the world, to make the most of their investment and the technology available, reduce costs and improve productivity.

The small business can also now partner with the FCC regulated Service Provider offering the SIP Trunk who can offer some centralized (hosted) services that may interest an SMB in addition to other telephony services bundled into their lease cost:

- E911
- CALEA
- Local Number Portability (LNP)
- Direct Inward Dial (DID) Numbers
- Billing
- VoIP to TDM Gateways

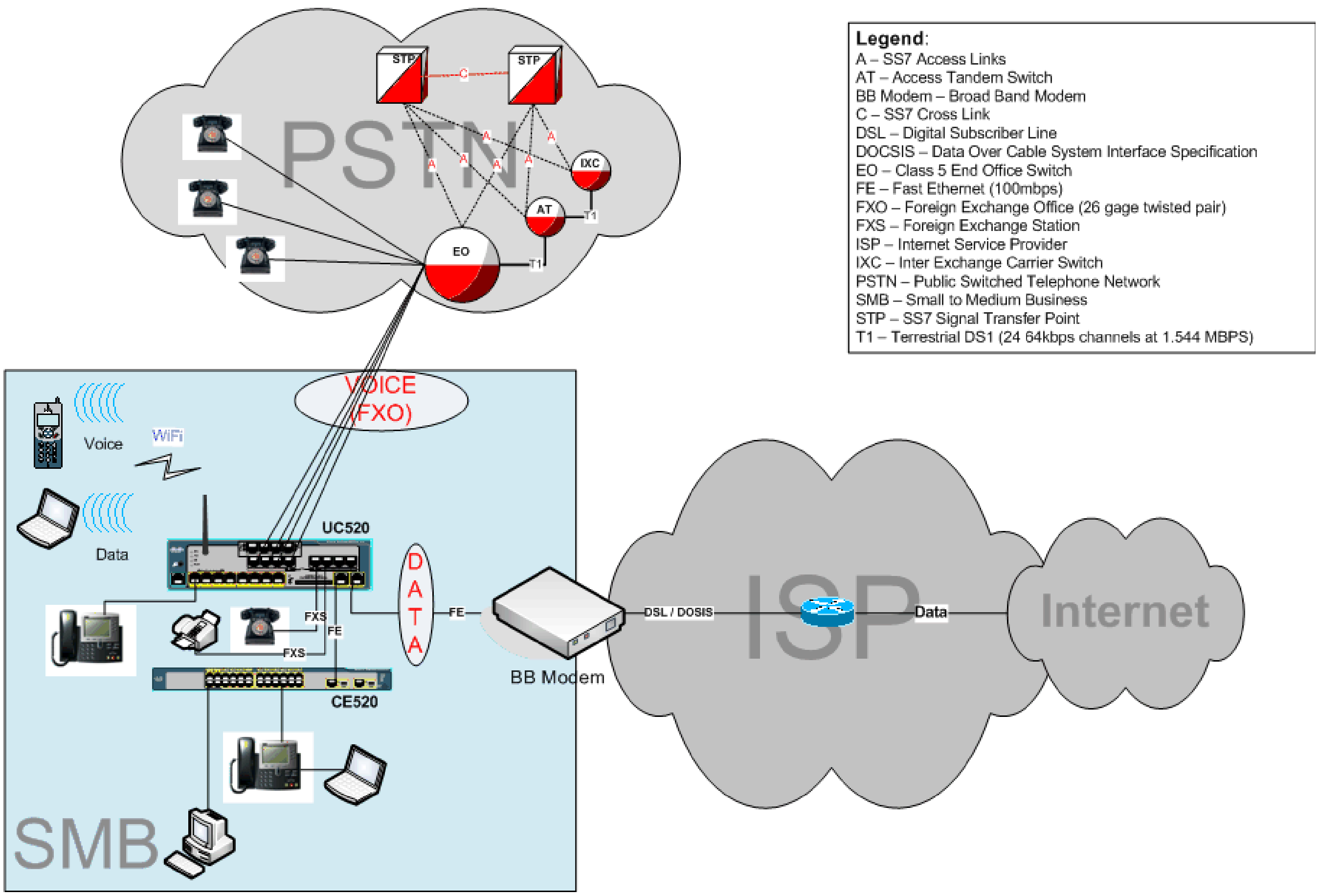
SBCS Interface

The SBCS product suite includes the UC520, which has a Fast Ethernet WAN interface that allows connection to the broad band router an ISP for data only or to the demarcation router of a SP network for SIP Trunk interconnect for voice and data.

SBCS with PSTN Analog Interface

The SBCS allows analog line interface to interconnect to Telephone Company FXO dial tone lines (up to 12 FXO interfaces on the UC520).

The initial implementation of the SBCS may have been done with separate PSTN Voice interconnect, while a data connection from a local ISP is also used required for internet access.



Examples of the BB Modem include a DOCSIS Cable Modem or a DSL Modem.

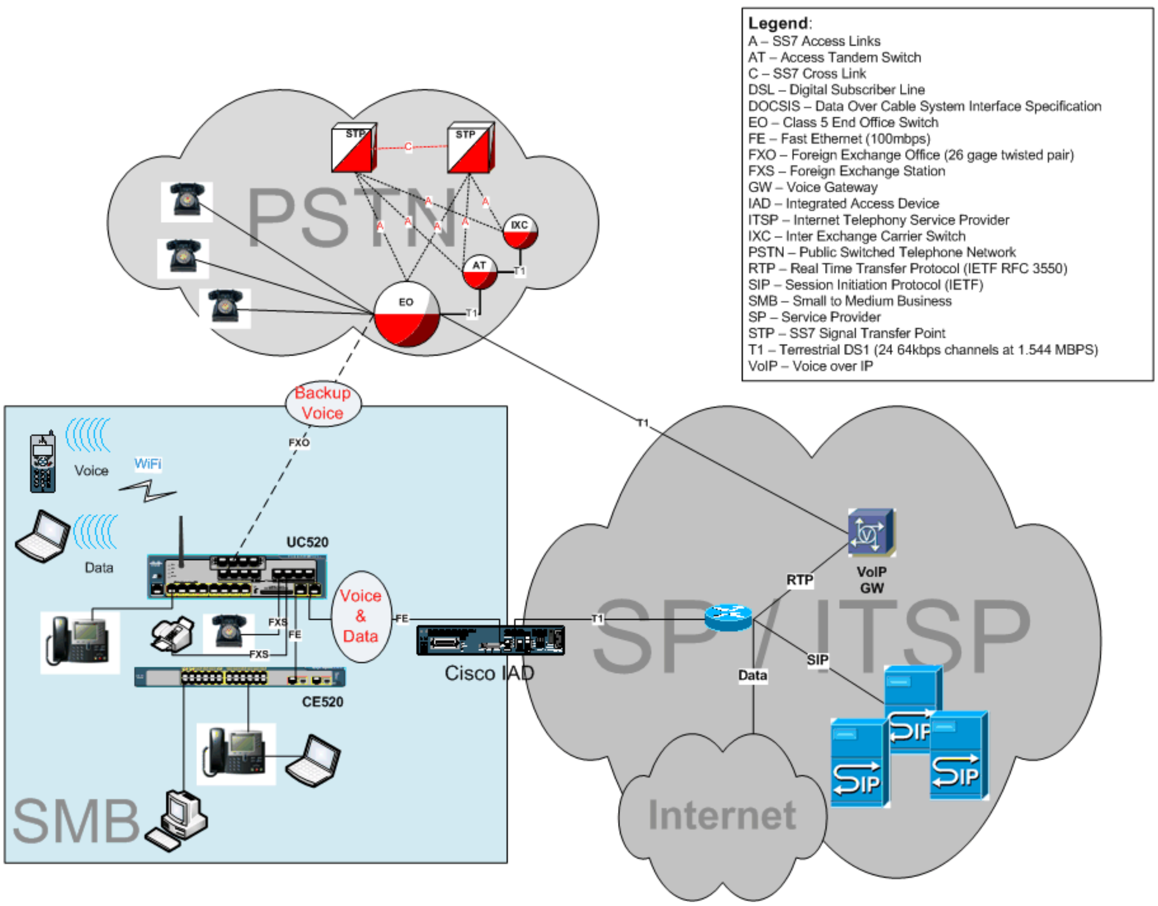
In this scenario, the SMB is operating a VoIP 'island' and enjoys the benefits of an IP converged network only within the confines of their LAN. Most unified communications applications they use cannot be extended beyond their office in this case.

SBCS with SIP TRUNK

Most SIP Trunk based SPs will only guarantee the network connection if it is their own network. The reason for this is quality of service (QoS) cannot be guaranteed over an intermediate ISP network. So when migrating to a SIP Trunk, the ISP interconnection as well as the Telco line cost is replaced with the single SP network interconnection.

If you did find a SP that allowed the BYOB (bring your own broadband) they may insist on over provisioning the bandwidth and using some type of call admission control to insure voice packets always get through, but it is more likely they won't encourage this type of connection.

This next diagram shows the SIP Trunk topology.



I have dotted the FXO analog interface since it may be needed in very small (if any) quantities when implementing SIP trunking (emergency backup or dedicated legacy fax).

The ISP BB Modem was replaced with the SP demarcation router (The Cisco IAD is the router of choice for many SPs). And since this is a SP (and not an ISP) they are extending their network capabilities, including SIP infrastructure, PSTN VoIP hop off gateways, and Internet interconnect all the way out to the SMB. Their connection is usually a T1 and they will guarantee a service level agreement for bandwidth and VoIP prioritization via QoS.

Quality of Service (QoS) implementation ensures that Voice traffic is treated efficiently without affecting its function or performance. Bandwidth, delay, jitter, and packet loss can all be thought of in terms of tolerances to stay within before you can no longer guarantee so much of each to a given user/application. QoS is required in a network in order to have an application insurance policy which maximizes the ROI on the network infrastructure. While QoS is implemented at L2 within the LAN, it is the L3 'end to end' QoS implementation that the SP will offer for VoIP.

SIP signaling and RTP Media are protected. Layer 3 TOS/DSCP (Type of Service/Differentiated Services Code Point) markings will be added by the SBCS to every VoIP packet handed to the SP and it will honor the same on ingress packets for full duplex voice communications.

The managed access router of the SP usually provides the following:

- NAT ALG functionality
- QoS for SIP trunk calls and SLA guarantees
- WAN conversion to Ethernet on UC500
- Well-defined demarcation point of troubleshooting for SP and customer

The SP doesn't own or manage anything else at the SMB, and the SBCS remains the responsibility of the customer and supporting VAR.

The bandwidth required for a number of simultaneous calls over a SIP trunk depends on the Codec you allow to be negotiated within the SIP signaling sessions. It can be G711 or G729 (the current default Codecs in UC500).

- G711: assume 87kpbs bi directional bandwidth required for each call
- G729: assume 31kbps/call.

So if you bandwidth is asymmetrical, we don't want to exceed the lower number. Also keep in mind that while QoS will assure and expedite forwarding of VoIP SIP Signaling and RTP Media packets, there will also be best effort data traffic required to get through to sustain business, so factor in a % of the bandwidth you would not assume available for VoIP.

If for instance you had 1.5mbps (a T1 WAN connection SIP Trunk) you could run about 17 simultaneous calls with G711, but that would leave nothing for data traffic. Leaving 30% of bandwidth for data calls would result in support of 12 simultaneous calls. You can use call admission controls to assist this.

There is a VoIP bandwidth Calculator on line you can use as well.

http://tools.cisco.com/Support/VBC/do/CodecCalc1.do

Cisco Validated SIP Trunk Providers

Cisco has designated a select few service providers for Managed SIP Trunking that Partners can recommend to the SMB customer. The program allows for

- Joint SP to Cisco testing to ensure quality of service offerings.
- SP receiving SIP Trunking Designation will be eligible for potential Joint Cisco/SP GTM activities.
- SP may be integrated into the Cisco Configuration Assistant (CCA) tool drop down menu

If a SP does not qualify, or does not wish to be tested/integrated, Cisco will support a self certification process to assist their GTM activities.

Implementation

This section includes Configuration and Operation of the SIP Trunk on the SBCS (UC500). It can be used as a reference and not a step by step guide for every SIP Trunk provider.

Prerequisites and Starting Conditions

A public routable IP address is assigned to the UC520 FE0/0 port (The official documentation recommends a STATIC IP)

The DNS IP addresses of the SIP Trunk Provider are to be already configured on the UC520.

NTP servers are configured on the UC520 using CCA. The IP addresses are provided by the SIP Trunk SP.

You will need the SIP Trunk account and credentials from a SP that you select. This will include authentication credentials and FQDNs of their SIP Servers. It will also include one or more DIDs that you will use to originate and terminate calls. In this document I am using a test account from Cbeyond.

This was done using CCA 1.8. Please use the latest available SBCS bundle.

Usually a SP will provide a demarcation router at the customer premises since they manage the network end to end (including QoS marking on RTP media and signaling). In my case, I did not use a SP router and connected over the public internet (which you would never do for QoS reasons in the real world). In the case of the SP provided router, you wouldn't have access to it anyway.

Configuration

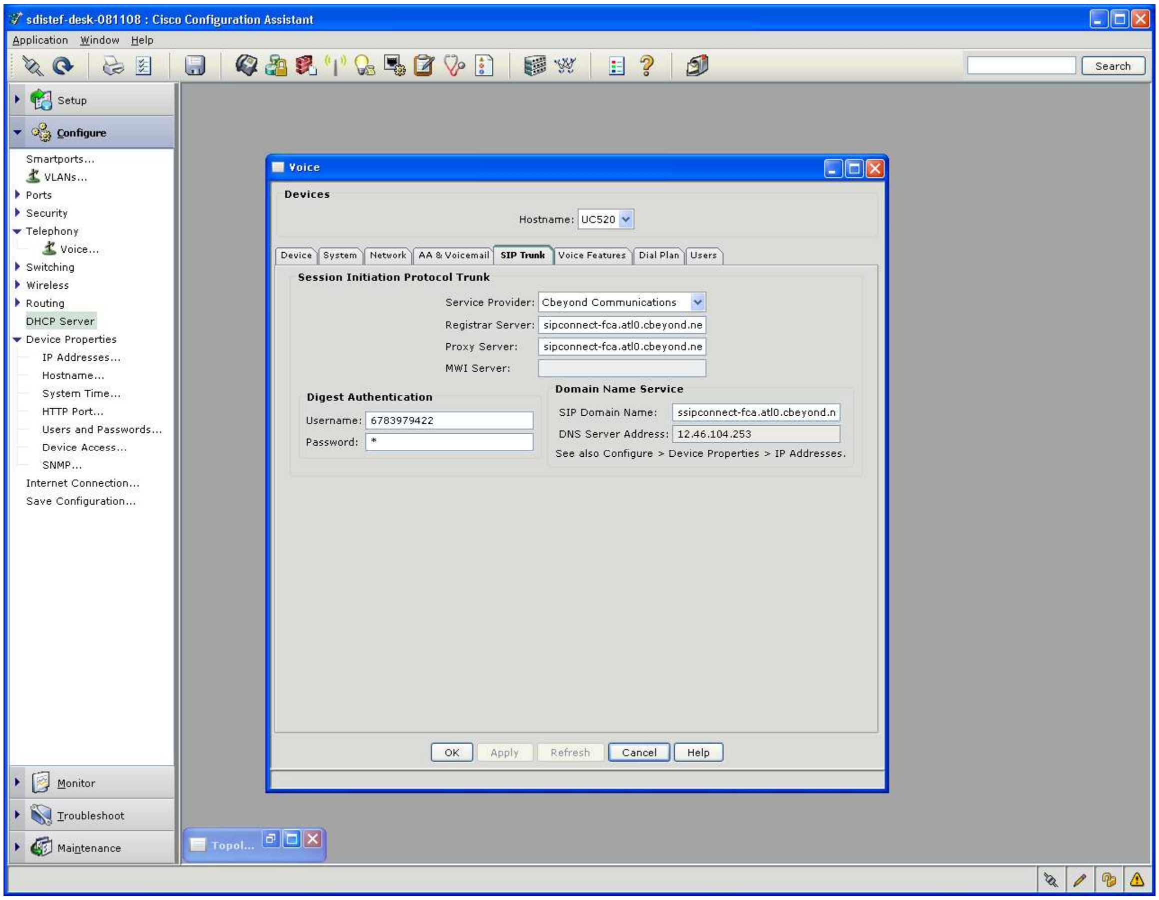
This entire SIP Trunk configuration was performed using Cisco Configuration Assistant and a UC520W-16U SKU with the normal default configuration. This is not to say that there aren't some customizations you can make later with CLI (we will discuss some), but out of the box, you can use CCA and get it working.

We will now jump right into the CCA GUIs and walk through the configuration. The sub headers are sometimes named after the CCA GUI TABS to help you follow along.

SIP Trunk

The drop down box allows you to select the S.P. from a list of SPs that have completed interoperability, or you can always select the "General" and try it. In the case of the named SPs, you still need to provide the domain name, Registration Server, and Proxy Server along with the authentication credentials you receive with your SIP Trunk Account.

This is specific to my test account and you would substitute your own test account credentials here (please don't use these).



AA & Voicemail

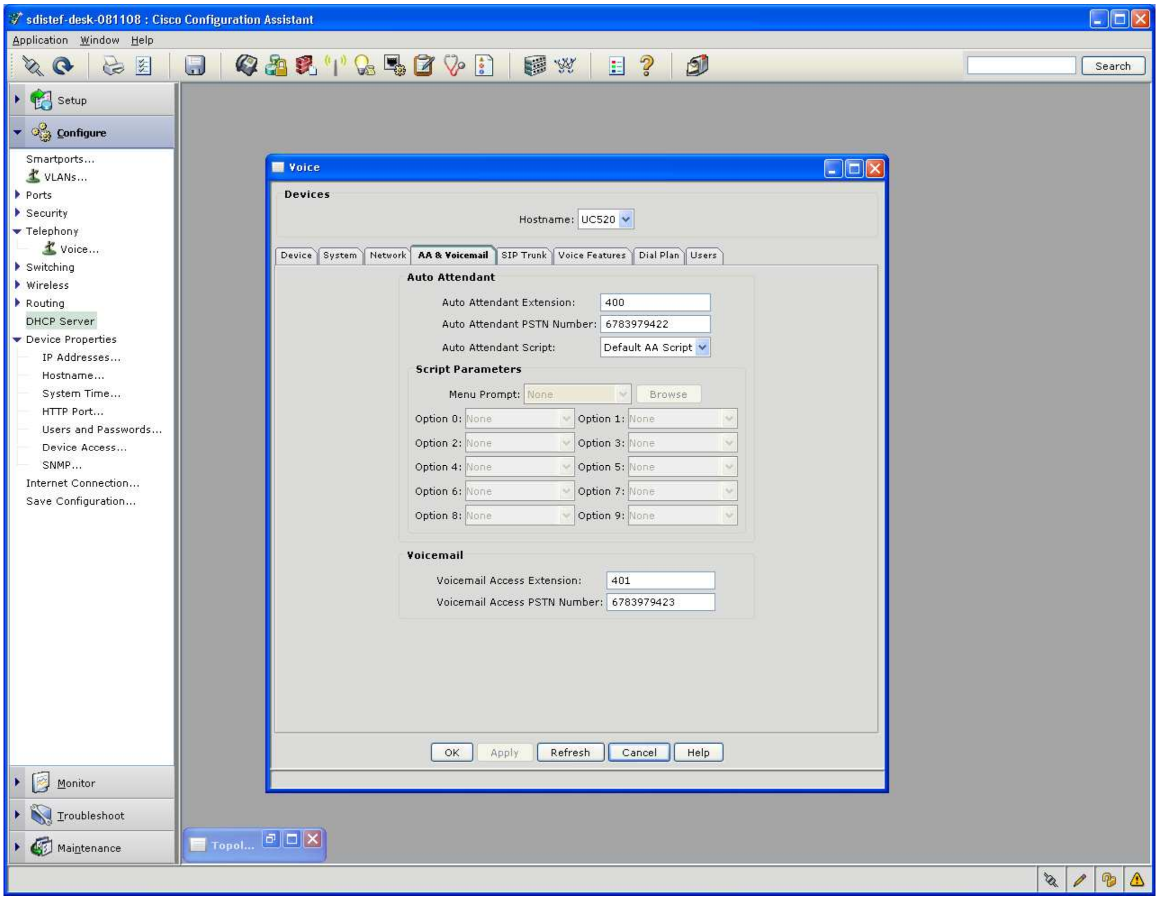
The DID(s) that the SIP Trunk SP assigned to your SIP Trunk account are yours to use as you see fit.

We have seen a lot of cases where if one DID is purchased, you can assign it to the Automated Attendant. This is the DID that may appear in Directory Assistance or the Yellow Book.

If you have multiple DIDs, we have seen where one can be used to access voice mail boxes of any phone from the outside world. You can also assign the DID to phones.

Lets look at AA and VMS access.

Note: the first time you access your voice mail box, you must be on the LAN to set it up. Subsequently, you can use the PSTN access DID to use it.


sdistef-desk-081108 : Cisco Configuration Assistant
Application Window Help
Search
Setup
Configure
Smartports...
VLANs...
Ports
Security
Telephony
Voice...
Switching
Wireless
Routing
DHCP Server
Device Properties
IP Addresses...
Hostname...
System Time...
HTTP Port...
Users and Passwords...
Device Access...
SNMP...
Internet Connection...
Save Configuration...
Monitor
Troubleshoot
Maintenance
Voice
Devices
Hostname: UC520
Device
System
Network
AA & Voicemail
SIP Trunk
Voice Features
Dial Plan
Users
Auto Attendant
Auto Attendant Extension: 400
Auto Attendant PSTN Number: 6783979422
Auto Attendant Script: Default AA Script
Script Parameters
Menu Prompt: None
Browse
Option 0: None
Option 1: None
Option 2: None
Option 3: None
Option 4: None
Option 5: None
Option 6: None
Option 7: None
Option 8: None
Option 9: None
Voicemail
Voicemail Access Extension: 401
Voicemail Access PSTN Number: 6783979423
OK
Apply
Refresh
Cancel
Help
Topol...

DID Mapping to Phones

The DID configuration sub menu off the Dial Plan Tab on the Voice configuration window allows you to MAP DIDs to particular extensions, so they may be accessed directly from the outside world (no need to transfer from the AA).

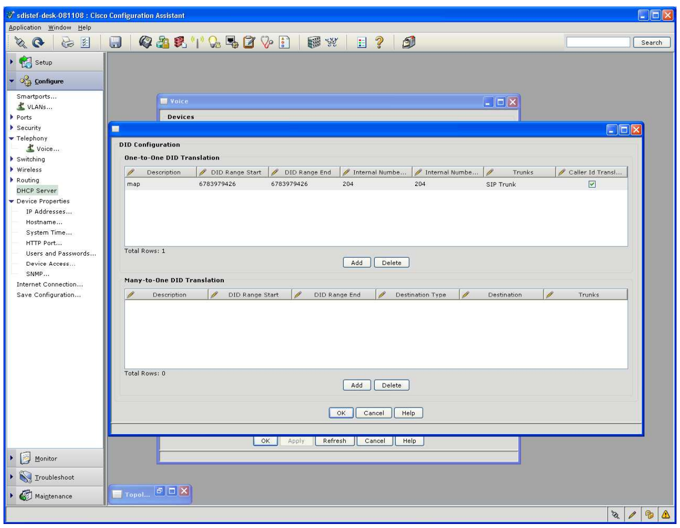
Some notes regarding Caller ID:

1) If the caller-id check box is selected and the last digits of the extension match that of the DID, then the outbound number would be the DID

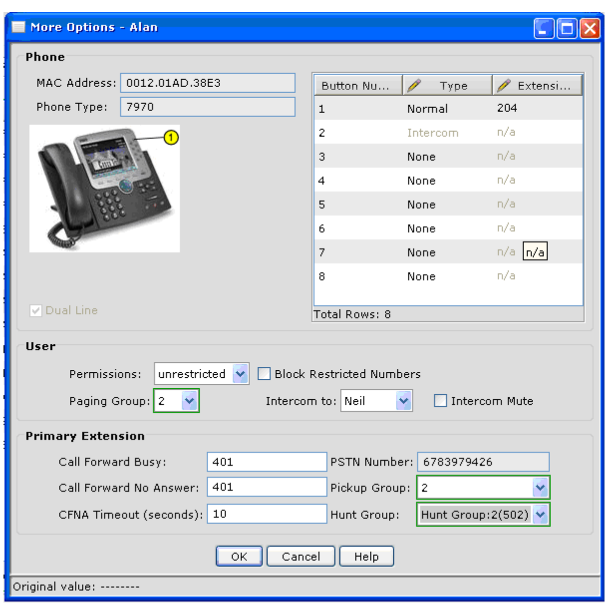
2) If the caller-id check box is selected and the last digits of the extension DONOT match that of the DID, then an error would be shown.

3) If caller-id check box is not selected outbound number would be the AA DID.

4) For any internal extension without DID, outbound number would be AA DID.



When you do this, you will notice the phone user gets populated with that DID.



SAVE

Save the running configuration from CCA (this performs a 'write mem' on the UC520)

SIP TLS

I did not configure SIP over TLS. While that is supported on IOS Gateways, I didn't have a need for it nor have we had any SIP Trunk provider that we have interfaced with thus far asked to support TLS. Probably because the provider offering SIP trunking also owns the IP access interconnect into the UC520 itself (managed service) so security is not a concern.

If this does need to be tested there needs to be some business justification especially if this involves any development work on IOS.

http://www.cisco.com/en/US/docs/ios/12_4t/12_4t11/FeatTLS.html

Resulting CLI

In this section, we will browse on the UC520 (SSH or Telnet to the CLI prompt) to see what the resulting configuration pushed from CCA looks like.

DID Mapping

Look to see how CCA built translation rules. My DIDs are 6783979422 (AA ext 400), 23 (VMS ext 401), 26 (Phone extension 204)

```
voice translation-rule 4
 rule 1 /6783979426/ /204/
!
voice translation-rule 410
 rule 1 /^9\(.......)\)$/ /678\1/
 rule 2 /401/ /6783979423/
 rule 3 /400/ /6783979422/
 rule 4 /^9\(.*\)/ /\1/
 rule 5 /^...$/ /6783979422/
!
voice translation-rule 1111
 rule 1 /204/ /6783979426/
 rule 15 /.*/ /6783979422/
!
voice translation-rule 1112
 rule 1 /^9/ //
!
voice translation-rule 2000
 rule 1 /6783979423/ /401/
!
voice translation-rule 2001
 rule 1 /6783979422/ /400/
```

Access List

An ACL is added to allow SIP protocol to pass through the external interface of the UC500.

```
access-list 105 permit udp any any eq 5060
access-list 105 permit udp any eq 5060 any
```

Dial Peers for SIP Calls and QOS

One of these for each call type actually…

```
dial-peer voice 1001 voip
 corlist outgoing call-local
 description ** Outgoing call to SIP trunk (Cbeyond Communications) **
 translation-profile outgoing PSTN_Outgoing
 destination-pattern 9[2-9]......
 voice-class codec 1
 voice-class sip dtmf-relay force rtp-nte
 session protocol sipv2
 session target sip-server
 dtmf-relay rtp-nte
 ip qos dscp cs5 media ← Precedence 5 for RTP Media (highest)
 ip qos dscp cs4 signaling ← Precedence 4 for SIP Signaling (next highest)
 no vad
!
```

Reference http://www.cisco.com/en/US/docs/ios/12_2t/12_2t2/feature/guide/ft_dscp.html for explanation of QOS.

SIP User Agent

CBeyond uses SIPCONNECT[1] so one DID can register the whole IP-PBX account, or they can each register individually.

```
sip-ua
 authentication username 6783979422 password 7 040C5354587919
 no remote-party-id
 retry invite 2
 retry register 10
 timers connect 100
 registrar dns:sipconnect-fca.atl0.cbeyond.net expires 3600
 sip-server dns:sipconnect-fca.atl0.cbeyond.net
  host-registrar
```

EPhone DNs

You will see ephone DNs (most) with the no-reg primary.

The one you assigned a DID to will look as follows as well as the ones you assigned to `AA and VMS`.

```
ephone-dn  12  dual-line
 number 203 no-reg primary
 pickup-group 2
 label 203
 description Pete Conrad
 name Pete Conrad
 call-forward busy 401
 call-forward noan 401 timeout 10
!
```

---

1 http://www.sipforum.org/sipconnect

```
!
ephone-dn  13  dual-line
 number 204 secondary 6783979426 no-reg primary
 pickup-group 2
 label 204
 description Alan Bean
 name Alan Bean
 call-forward busy 401
 call-forward noan 401 timeout 10

ephone-dn  77
 number 6783979422
 description SIP AA trunk registration
 preference 10
!
!
ephone-dn  78
 number 6783979423
 description SIP VM trunk registration
 preference 10
!
```

Other CLI

CODEC

The UC500 currently supports the following Codecs for voice in the U.S., preferred in the given order in the default configuration. You will see voice class codec referenced from dial peers.

This basically means that inside the SDP package of the SIP INVITE, we will negotiate to use these Codecs in this order.

```
voice class codec 1
 codec preference 1 g711ulaw
 codec preference 2 g729r8
```

Operation

Making test calls is the best way to confirm its working…

To see these messages, Telnet or SSH to the CLI and enter:

UC520#debug ccsip message

UC520#term mon

Confirm Registration

UC520#show debug

CCSIP SPI: SIP Call Message tracing is enabled (filter is OFF)

UC520#show sip register status

```
Line         peer         expires(sec)  registered
============ ============= ============ ===========
```

6783979422	20053	64	yes
6783979423	20052	64	yes
6783979426	20056	1931	yes

And you will see these exchanges periodically, SIP Registration messages being challenged (401), and then responded to with credentials and accepted (200 OK).

REGISTER sip:sipconnect-fca.atl0.cbeyond.net:5060 SIP/2.0
Via: SIP/2.0/UDP 12.19.92.195:5060;branch=z9hG4bK61636
From: <sip:6783979422@sipconnect-fca.atl0.cbeyond.net>;tag=5E730D78-11FC
To: <sip:6783979422@sipconnect-fca.atl0.cbeyond.net>
Date: Mon, 15 Sep 2008 22:51:20 GMT
Call-ID: 5D05CB23-828711DD-80CFD9F9-94599E0A
User-Agent: Cisco-SIPGateway/IOS-12.x
Max-Forwards: 70
Timestamp: 1221519080
CSeq: 14 REGISTER
Contact: <sip:6783979422@12.19.92.195:5060>
Expires: 3600
Content-Length: 0

Sep 15 22:51:20.378: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Received:
SIP/2.0 **401 Unauthorized**
Via:SIP/2.0/UDP 12.19.92.195:5060;branch=z9hG4bK61636
From:<sip:6783979422@sipconnect-fca.atl0.cbeyond.net>;tag=5E730D78-11FC
To:<sip:6783979422@sipconnect-fca.atl0.cbeyond.net>
Call-ID:5D05CB23-828711DD-80CFD9F9-94599E0A
CSeq:14 REGISTER
WWW-Authenticate:DIGEST
realm="BroadWorks",qop="auth",algorithm=MD5,nonce="BroadWorksXfl5oygthTgexx4cBW"
Content-Length:0

Sep 15 22:51:20.378: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Sent:
REGISTER sip:sipconnect-fca.atl0.cbeyond.net:5060 SIP/2.0
Via: SIP/2.0/UDP 12.19.92.195:5060;branch=z9hG4bK6316F0
From: <sip:6783979422@sipconnect-fca.atl0.cbeyond.net>;tag=5E730D78-11FC
To: <sip:6783979422@sipconnect-fca.atl0.cbeyond.net>
Date: Mon, 15 Sep 2008 22:51:20 GMT
Call-ID: 5D05CB23-828711DD-80CFD9F9-94599E0A
User-Agent: Cisco-SIPGateway/IOS-12.x
Max-Forwards: 70
Timestamp: 1221519080
CSeq: 15 REGISTER
Contact: <sip:6783979422@12.19.92.195:5060>
Expires: 3600
Authorization: Digest username="6783979422",realm="BroadWorks",uri="sip:sipconnect-fca.atl0.cbeyond.net:5060",response="46d8db90e1cf7d4729ec6936ab6cc2a3",nonce="BroadWorksXfl5oygthTgexx4cBW",cnonce="68552D6F",qop="auth",algorithm=MD5,nc=00000001
Content-Length: 0

Sep 15 22:51:20.450: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Received:
SIP/2.0 200 OK
Via:SIP/2.0/UDP 12.19.92.195:5060;branch=z9hG4bK6316F0
From:<sip:6783979422@sipconnect-fca.atl0.cbeyond.net>;tag=5E730D78-11FC
To:<sip:6783979422@sipconnect-fca.atl0.cbeyond.net>
Call-ID:5D05CB23-828711DD-80CFD9F9-94599E0A
CSeq:15 REGISTER
Contact:<sip:6783979422@12.19.92.195:5060>;q=0.5;expires=3599
Content-Length:0

Call Set Up

Sep 15 23:03:18.282: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Sent:
INVITE sip:19193926219@sipconnect-fca.atl0.cbeyond.net:5060 SIP/2.0
Via: SIP/2.0/UDP 12.19.92.195:5060;branch=z9hG4bK651E8C
From: "Alan Shepard" <sip:6783979422@sipconnect-fca.atl0.cbeyond.net>;tag=5E7E0208-68
To: <sip:19193926219@sipconnect-fca.atl0.cbeyond.net>
Date: Mon, 15 Sep 2008 23:03:18 GMT
Call-ID: 51B6EA42-82B111DD-818DD9F9-94599E0A@ssipconnect-fca.atl0.cbeyond.net
Supported: 100rel,timer,resource-priority,replaces
Min-SE: 1800
Cisco-Guid: 1328462136-2192642525-2173229561-2488901130
User-Agent: Cisco-SIPGateway/IOS-12.x
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER
CSeq: 101 INVITE
Max-Forwards: 70
Timestamp: 1221519798
Contact: <sip:6783979422@12.19.92.195:5060>
Expires: 180
Allow-Events: telephone-event
Content-Type: application/sdp
Content-Disposition: session;handling=required
Content-Length: 282

v=0
o=CiscoSystemsSIP-GW-UserAgent 9487 5443 IN IP4 12.19.92.195
s=SIP Call
c=IN IP4 12.19.92.195
t=0 0
m=audio 17144 RTP/AVP 0 18 101
c=IN IP4 12.19.92.195
a=rtpmap:0 PCMU/8000
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16

Sep 15 23:03:18.350: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Received:
SIP/2.0 401 Unauthorized
Via:SIP/2.0/UDP 12.19.92.195:5060;branch=z9hG4bK651E8C
From:"Alan Shepard"<sip:6783979422@sipconnect-fca.atl0.cbeyond.net>;tag=5E7E0208-68
To:<sip:19193926219@sipconnect-fca.atl0.cbeyond.net>;tag=1612032942-1221519858264
Call-ID:51B6EA42-82B111DD-818DD9F9-94599E0A@ssipconnect-fca.atl0.cbeyond.net
CSeq:101 INVITE
WWW-Authenticate:DIGEST
realm="BroadWorks",qop="auth",algorithm=MD5,nonce="BroadWorksXfl5pduu0Th9ywpoBW"
Content-Length:0

Sep 15 23:03:18.354: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Sent:
ACK sip:19193926219@sipconnect-fca.atl0.cbeyond.net:5060 SIP/2.0
Via: SIP/2.0/UDP 12.19.92.195:5060;branch=z9hG4bK651E8C
From: "Alan Shepard" <sip:6783979422@sipconnect-fca.atl0.cbeyond.net>;tag=5E7E0208-68
To: <sip:19193926219@sipconnect-fca.atl0.cbeyond.net>;tag=1612032942-1221519858264
Date: Mon, 15 Sep 2008 23:03:18 GMT
Call-ID: 51B6EA42-82B111DD-818DD9F9-94599E0A@ssipconnect-fca.atl0.cbeyond.net
Max-Forwards: 70
CSeq: 101 ACK
Allow-Events: telephone-event
Content-Length: 0

Sep 15 23:03:18.354: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Sent:
INVITE sip:19193926219@sipconnect-fca.atl0.cbeyond.net:5060 SIP/2.0
Via: SIP/2.0/UDP 12.19.92.195:5060;branch=z9hG4bK661873
From: "Alan Shepard" <sip:6783979422@sipconnect-fca.atl0.cbeyond.net>;tag=5E7E0208-68
To: <sip:19193926219@sipconnect-fca.atl0.cbeyond.net>
Date: Mon, 15 Sep 2008 23:03:18 GMT
Call-ID: 51B6EA42-82B111DD-818DD9F9-94599E0A@ssipconnect-fca.atl0.cbeyond.net
Supported: 100rel,timer,resource-priority,replaces
Min-SE: 1800
Cisco-Guid: 1328462136-2192642525-2173229561-2488901130
User-Agent: Cisco-SIPGateway/IOS-12.x
Allow: INVITE, OPTIONS, BYE, CANCEL, ACK, PRACK, UPDATE, REFER, SUBSCRIBE, NOTIFY, INFO, REGISTER
CSeq: 102 INVITE
Max-Forwards: 70
Timestamp: 1221519798
Contact: <sip:6783979422@12.19.92.195:5060>
Expires: 180
Allow-Events: telephone-event
Authorization: Digest
username="6783979422",realm="BroadWorks",uri="sip:19193926219@sipconnect-

fca.atl0.cbeyond.net:5060",response="c3b8b732d3e53ffceed914500ce9d9bb",nonce="BroadWorksXfl5pduu0Th9ywpoBW",cnonce="B5ADD34D",qop="auth",algorithm=MD5,nc=00000001
Content-Type: application/sdp
Content-Disposition: session;handling=required
Content-Length: 282

v=0
o=CiscoSystemsSIP-GW-UserAgent 9487 5443 IN IP4 12.19.92.195
s=SIP Call
c=IN IP4 12.19.92.195
t=0 0
m=audio 17144 RTP/AVP 0 18 101
c=IN IP4 12.19.92.195
a=rtpmap:0 PCMU/8000
a=rtpmap:18 G729/8000
a=fmtp:18 annexb=no
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-16

Sep 15 23:03:18.418: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Received:
SIP/2.0 100 Trying
Via:SIP/2.0/UDP 12.19.92.195:5060;branch=z9hG4bK661873
From:"Alan Shepard"<sip:6783979422@sipconnect-fca.atl0.cbeyond.net>;tag=5E7E0208-68
To:<sip:19193926219@sipconnect-fca.atl0.cbeyond.net>
Call-ID:51B6EA42-82B111DD-818DD9F9-94599E0A@ssipconnect-fca.atl0.cbeyond.net
CSeq:102 INVITE
Content-Length:0

Sep 15 23:03:20.130: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Received:
SIP/2.0 180 Ringing
Via:SIP/2.0/UDP 12.19.92.195:5060;branch=z9hG4bK661873
From:"Alan Shepard"<sip:6783979422@sipconnect-fca.atl0.cbeyond.net>;tag=5E7E0208-68
To:<sip:19193926219@sipconnect-fca.atl0.cbeyond.net>;tag=807056555-1221519860040
Call-ID:51B6EA42-82B111DD-818DD9F9-94599E0A@ssipconnect-fca.atl0.cbeyond.net
CSeq:102 INVITE
Allow:ACK,BYE,CANCEL,INFO,INVITE,OPTIONS,PRACK,REFER,NOTIFY
Supported:
Contact:<sip:sipconnect-fca1.atl0.cbeyond.net>
Content-Type:application/sdp
Content-Length:354

v=0
o=BroadWorks 1549149 1 IN IP4 72.16.223.241
s=-
c=IN IP4 72.16.223.241
t=0 0
m=audio 16856 RTP/AVP 0 101 100
a=rtpmap:101 telephone-event/8000

a=fmtp:101 0-15
a=rtpmap:100 X-NSE/8000
a=fmtp:100 192-194,200-202
a=sqn:0
a=cdsc:1 audio RTP/AVP 100
a=cpar:a=rtpmap:100 X-NSE/8000
a=cpar:a=fmtp:100 192-194,200-202
a=cdsc:2 image udptl t38

Sep 15 23:03:22.390: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Received:
SIP/2.0 200 OK
Via:SIP/2.0/UDP 12.19.92.195:5060;branch=z9hG4bK661873
From:"Alan Shepard"<sip:6783979422@sipconnect-fca.atl0.cbeyond.net>;tag=5E7E0208-68
To:<sip:19193926219@sipconnect-fca.atl0.cbeyond.net>;tag=807056555-1221519860040
Call-ID:51B6EA42-82B111DD-818DD9F9-94599E0A@ssipconnect-fca.atl0.cbeyond.net
CSeq:102 INVITE
Supported:
Contact:<sip:sipconnect-fca1.atl0.cbeyond.net>
Allow:ACK,BYE,CANCEL,INFO,INVITE,OPTIONS,PRACK,REFER,NOTIFY,UPDATE
Accept:multipart/mixed,application/media_control+xml,application/sdp
Content-Type:application/sdp
Content-Length:354

v=0
o=BroadWorks 1549149 1 IN IP4 72.16.223.241
s=-
c=IN IP4 72.16.223.241
t=0 0
m=audio 16856 RTP/AVP 0 101 100
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15
a=rtpmap:100 X-NSE/8000
a=fmtp:100 192-194,200-202
a=sqn:0
a=cdsc:1 audio RTP/AVP 100
a=cpar:a=rtpmap:100 X-NSE/8000
a=cpar:a=fmtp:100 192-194,200-202
a=cdsc:2 image udptl t38

Sep 15 23:03:22.398: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Sent:
ACK sip:sipconnect-fca1.atl0.cbeyond.net:5060 SIP/2.0
Via: SIP/2.0/UDP 12.19.92.195:5060;branch=z9hG4bK671E77
From: "Alan Shepard" <sip:6783979422@sipconnect-fca.atl0.cbeyond.net>;tag=5E7E0208-68
To: <sip:19193926219@sipconnect-fca.atl0.cbeyond.net>;tag=807056555-1221519860040
Date: Mon, 15 Sep 2008 23:03:18 GMT
Call-ID: 51B6EA42-82B111DD-818DD9F9-94599E0A@ssipconnect-fca.atl0.cbeyond.net
Max-Forwards: 70
CSeq: 102 ACK

Authorization: Digest
username="6783979422",realm="BroadWorks",uri="sip:19193926219@sipconnect-fca.atl0.cbeyond.net:5060",response="c3b8b732d3e53ffceed914500ce9d9bb",nonce="BroadWorksXfl5pduu0Th9ywpoBW",cnonce="B5ADD34D",qop="auth",algorithm=MD5,nc=00000001
Allow-Events: telephone-event
Content-Length: 0

Call Tear Down

UC520#
Sep 15 23:06:46.958: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Sent:
BYE sip:sipconnect-fca1.atl0.cbeyond.net:5060 SIP/2.0
Via: SIP/2.0/UDP 12.19.92.195:5060;branch=z9hG4bK68190F
From: "Alan Shepard" <sip:6783979422@sipconnect-fca.atl0.cbeyond.net>;tag=5E7E0208-68
To: <sip:19193926219@sipconnect-fca.atl0.cbeyond.net>;tag=807056555-1221519860040
Date: Mon, 15 Sep 2008 23:03:18 GMT
Call-ID: 51B6EA42-82B111DD-818DD9F9-94599E0A@ssipconnect-fca.atl0.cbeyond.net
User-Agent: Cisco-SIPGateway/IOS-12.x
Max-Forwards: 70
Timestamp: 1221520006
CSeq: 103 BYE
Authorization: Digest username="6783979422",realm="BroadWorks",uri="sip:sipconnect-fca1.atl0.cbeyond.net:5060",response="9e349f65a3e1bfec25b46774907cbf46",nonce="BroadWorksXfl5pduu0Th9ywpoBW",cnonce="894DB382",qop="auth",algorithm=MD5,nc=00000002
Reason: Q.850;cause=16
Content-Length: 0

Sep 15 23:06:47.018: //-1/xxxxxxxxxxxx/SIP/Msg/ccsipDisplayMsg:
Received:
SIP/2.0 200 OK
Via:SIP/2.0/UDP 12.19.92.195:5060;branch=z9hG4bK68190F
From:"Alan Shepard"<sip:6783979422@sipconnect-fca.atl0.cbeyond.net>;tag=5E7E0208-68
To:<sip:19193926219@sipconnect-fca.atl0.cbeyond.net>;tag=807056555-1221519860040
Call-ID:51B6EA42-82B111DD-818DD9F9-94599E0A@ssipconnect-fca.atl0.cbeyond.net
CSeq:103 BYE
Content-Length:0

Statistics

Right from the SBCS UC520, the administrator can query for SIP Statistics if necessary.

SIP Statistics

UC520#sh sip statistics
SIP Response Statistics (Inbound/Outbound)
  Informational:
   Trying 16/19, Ringing 15/21,
   Forwarded 0/0, Queued 0/0,
   SessionProgress 0/0
  Success:

```
    OkInvite 15/5, OkBye 9/18,
    OkCancel 5/14, OkOptions 0/0,
    OkPrack 0/9, OkRegister 20/0
    OkSubscribe 0/0, OkNotify 22/0, OkPublish 0/0
    OkInfo 0/0, OkUpdate 0/1,
    202Accepted 0/0, OkOptions 0/0
  Redirection (Inbound only except for MovedTemp(Inbound/Outbound)) :
    MultipleChoice 0, MovedPermanently 0,
    MovedTemporarily 0/0, UseProxy 0,
    AlternateService 0
  Client Error:
    BadRequest 0/0, Unauthorized 27/0,
    PaymentRequired 0/0, Forbidden 0/0,
    NotFound 0/0, MethodNotAllowed 0/0,
    NotAcceptable 0/0, ProxyAuthReqd 0/0,
    ReqTimeout 0/0, Conflict 0/0, Gone 0/0,
    ConditionalRequestFailed 0/0,
    ReqEntityTooLarge 0/0, ReqURITooLarge 0/0,
    UnsupportedMediaType 0/0, UnsupportedURIScheme 0/0,
    BadExtension 0/0, IntervalTooBrief 0/0,
    TempNotAvailable 0/0, CallLegNonExistent 0/0,
    LoopDetected 0/0, TooManyHops 0/0,
    AddrIncomplete 0/0, Ambiguous 0/0,
    BusyHere 0/0, RequestCancel 5/86,
    NotAcceptableMedia 0/0, BadEvent 0/0,
    SETooSmall 0/0, , RequestPending 0/0
    UnsupportedResourcePriority 0/0
  Server Error:
    InternalError 0/0, NotImplemented 0/0,
    BadGateway 0/0, ServiceUnavail 0/0,
    GatewayTimeout 0/0, BadSipVer 0/0,
    PreCondFailure 0/0
  Global Failure:
    BusyEverywhere 0/0, Decline 0/0,
    NotExistAnywhere 0/0, NotAcceptable 0/0
  Miscellaneous counters:
    RedirectRspMappedToClientErr 0

SIP Total Traffic Statistics (Inbound/Outbound)
  Invite 19/23, Ack 7/23, Bye 18/9,
  Cancel 14/5, Options 0/0,
  Prack 9/0, Update 1/0,
  Subscribe 0/0, Notify 0/22, Publish 0/0
  Refer 0/0, Info 0/0,
  Register 0/40

Retry Statistics
  Invite 0, Bye 0, Cancel 0, Response 72,
  Prack 0, Reliable1xx 0, Notify 0, Info 0
  Register 0 Subscribe 0 Update 0 Options 0
```

SDP application statistics:
Parses: 37, Builds 21
Invalid token order: 0, Invalid param: 0
Not SDP desc: 0, No resource: 0

Last time SIP Statistics were cleared: <never>

SIP Status

Retrieving the status of the SIP stack is also possible from UC520.

UC520#sh sip status
SIP User Agent Status
SIP User Agent for UDP : ENABLED
SIP User Agent for TCP : ENABLED

SIP User Agent for TLS over TCP : ENABLED
SIP User Agent bind status(signaling): DISABLED
SIP User Agent bind status(media): DISABLED
SIP early-media for 180 responses with SDP: ENABLED
SIP max-forwards : 70
SIP DNS SRV version: 2 (rfc 2782)
NAT Settings for the SIP-UA
Role in SDP: NONE
Check media source packets: DISABLED
Maximum duration for a telephone-event in NOTIFYs: 2000 ms
SIP support for ISDN SUSPEND/RESUME: ENABLED
Redirection (3xx) message handling: ENABLED
Reason Header will override Response/Request Codes: DISABLED
Out-of-dialog Refer: DISABLED
Presence support is DISABLED

SDP application configuration:
Version line (v=) required
Owner line (o=) required
Timespec line (t=) required
Media supported: audio image
Network types supported: IN
Address types supported: IP4
Transport types supported: RTP/AVP udptl

SIP Calls

The administrator can query for SIP calls on the UC520 as well.

UC520#sh sip calls
SIP UAC CALL INFO
Number of SIP User Agent Client(UAC) calls: 0

SIP UAS CALL INFO

```
Call 1
SIP Call ID             : BW191228398150908-474190179@72.16.223.36
   State of the call      : STATE_ACTIVE (7)
   Substate of the call    : SUBSTATE_NONE (0)
   Calling Number         : 9193926219
   Called Number          : 6783979426
   Bit Flags              : 0x8C4401C 0x100 0x4
   CC Call ID             : 170
   Source IP Address (Sig ): 12.19.92.195
   Destn SIP Req Addr:Port : 72.16.223.36:5060
   Destn SIP Resp Addr:Port: 72.16.223.36:5060
   Destination Name        : 72.16.223.36
   Number of Media Streams : 1
   Number of Active Streams: 1
   RTP Fork Object         : 0x0
   Media Mode              : flow-through
   Media Stream 1
     State of the stream      : STREAM_ACTIVE
     Stream Call ID           : 170
     Stream Type              : voice+dtmf (1)
     Negotiated Codec         : g711ulaw (160 bytes)
     Codec Payload Type       : 0
     Negotiated Dtmf-relay    : rtp-nte
     Dtmf-relay Payload Type  : 101
     Media Source IP Addr:Port: 12.19.92.195:17480
     Media Dest IP Addr:Port  : 72.16.223.241:32588
     Orig Media Dest IP Addr:Port : 0.0.0.0:0

Options-Ping    ENABLED:NO    ACTIVE:NO
   Number of SIP User Agent Server(UAS) calls: 1
```

References

Please visit the support wiki for much more information and details on SIP Trunking.

http://supportwiki.cisco.com/ViewWiki/index.php/SBCS:SIP/H.323_trunks_-_Cisco_Unified_Communications_500_Series_-_Cisco_Smart_Business_Communication_Systems