



Avaya Solution & Interoperability Test Lab

Configuring Alcatel OmniPCX Enterprise with Avaya Meeting Exchange™ Enterprise Edition 5.2 – Issue 1.0

Abstract

These Application Notes present a sample configuration for a network consisting of an Alcatel OmniPCX Enterprise and Avaya Meeting Exchange™ Enterprise Edition. These two systems are connected via a SIP trunk.

Testing was conducted via the Internal Interoperability Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

The purpose of this interoperability application note is to validate Alcatel OmniPCX Enterprise (OXE) with Avaya Meeting Exchange™ Enterprise Edition (MX). The sample network is shown in **Figure 1**, where the Alcatel OmniPCX Enterprise supports the Alcatel ipTouch 4028 / 4038 / 4068 IP Telephones. A SIP trunk is used to connect Alcatel OmniPCX Enterprise and Avaya Meeting Exchange™ Enterprise. All inter-system calls are carried over this SIP trunk. Alcatel phones are registered to Alcatel OmniPCX Enterprise. Alcatel OmniPCX Enterprise registered stations use extensions 3600x.

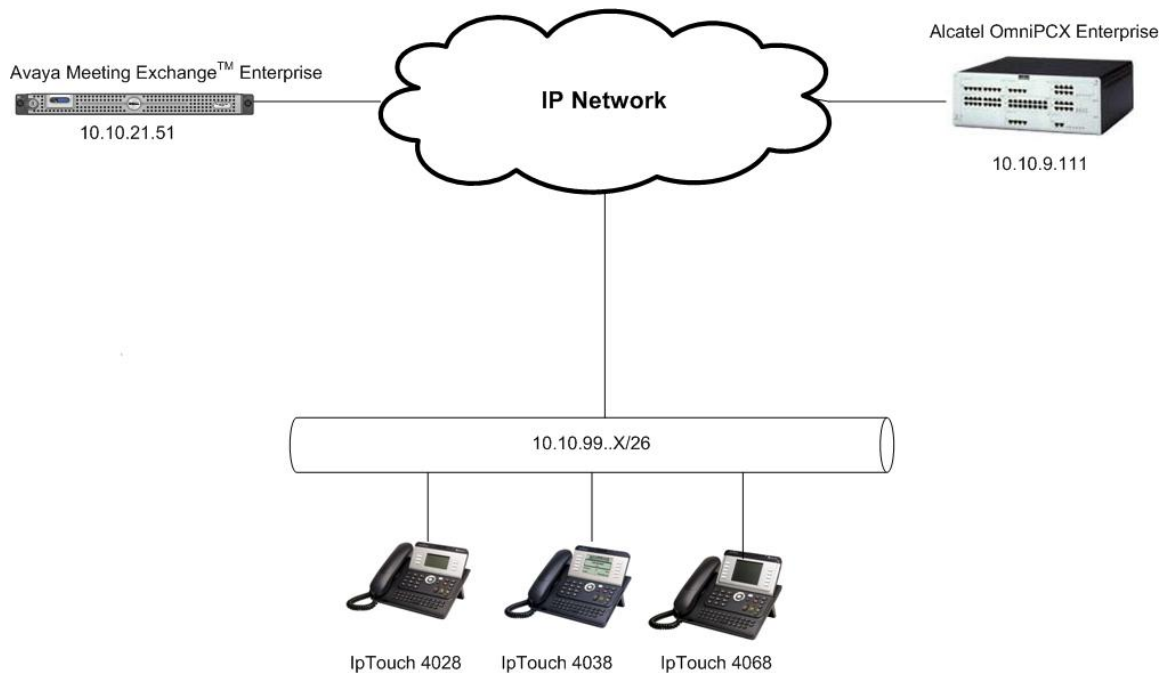


Figure 1: Connection of Alcatel OmniPCX Enterprise and Avaya Meeting Exchange™ Enterprise Edition via a SIP trunk

1.1. Equipment and Software Validated

The following equipment and software/firmware were used for the sample configuration:

Hardware Component	Software Version
Alcatel OmniPCX Enterprise	9.1 (I1.605-16-c)
Alcatel ipTouch NOE Telephone	4.20.71
Avaya S8510 server	Avaya Meeting Exchange TM Enterprise Edition R5.2 (Build 5.2.1.0.4) + mx-bridge patch (5.2.1.30.1-1)
Windows Computer	Avaya Bridge Talk (BT) 5.2.0.0.7

2. Configure Alcatel OmniPCX Enterprise

This section shows the configuration in Alcatel OmniPCX Enterprise. All configurations in this section are administered using the Command Line Interface. These Application Notes assumed that the basic configuration has already been administered. For further information on Alcatel OmniPCX Enterprise, please consult with references [2] and [3].

The procedures include the following areas:

- Verify Alcatel OXE Licences
- Access the Alcatel OXE Manager
- Administer IP Domain
- Administer SIP Trunk Group
- Administer Gateway
- Administer SIP Proxy
- Administer SIP External Gateway
- Administer Network Routing Table
- Administer Prefix Plan
- Administer Codec on SIP Trunk Group

Note: All configuration is completed using the Alcatel OXE manager menu. To enter the menu type **mgr** at the CLI prompt.

2.1. Verify Alcatel OXE Licenses

From the CLI prompt, use the **spadmin** command and from the menu shown, select option 2 **Display active file**. This will show the license files installed on the system.

```
Display current counters ..... 1
Display active file ..... 2
Check active file coherency ..... 3
Install a new file ..... 4
Read the system CPUID ..... 5
CPU-Ids management ..... 6
Display active and new file ..... 7
Display OPS limits ..... 8
Display ACK code ..... 9
Exit ..... 0
```

2.2. Access the Alcatel OXE Manager

Establish a Telnet connection to the CS board of the Alcatel OXE. At the CLI prompt type **mgr** and a menu is then presented.

```
+--Select an object-----+
|
| -> Shelf
|   Media Gateway
|   PWT/DECT System
|   System
|   Translator
|   Classes of Service
|   Attendant
|   Users
|   Users by profile
|   Set Profile
|   Groups
|   Speed Dialing
|   Phone Book
|   Entities
|   Trunk Groups
|   External Services
|   Inter-Node Links
|   X25
|   DATA
|   Applications
|   Specific Telephone Services
|   ATM
|   Events Routing Discriminator
|   Security and Access Control
|   IP
|   SIP
|   DHCP Configuration
|   Alcatel-Lucent 8&9 Series
|   SIP Extension
|   Encryption
|   Passive Com. Server
|   SNMP Configuration
|
+-----+
```

2.3. Administer IP Domain

To create an IP domain select **IP → IP domain**. Complete the following option:

- **IP Domain Name** **node1.mmsil.local**

Click **ctrl+v** to complete.

```
+--Create: IP domain-----+
|
|      Node Number (reserved) : 1
|      Instance (reserved)   : 1
|      IP Domain Number      : 0
|
|      IP Domain Name : node1.mmsil.local
|      Country        + Default
|      Intra-domain Coding Algorithm + Default
|      Extra-domain Coding Algorithm + Default
|      FAX/MODEM Intra domain call transp + NO
|      FAX/MODEM Extra domain call transp + NO
|      G722 allowed in Intra-domain + NO
|      G722 allowed in Extra-domain + NO
|      Tandem Primary Domain : -1
|      Domain Max Voice Connection : -1
|      IP Quality of service : 0
|      Contact Number : -----
|      Backup IP address : -----
|      Trunk Group ID : 10
|      IP recording quality of service : 0
|      Time Zone Name + System Default
|      Calling Identifier : -----
|      Supplement. Calling Identifier : -----
|      SIP Survivability Mode + NO
|
+-----+
```

2.4. Administer SIP Trunk Group

To add a SIP Trunk Group select **Trunk Groups** → **Create**. Complete the following options:

- **Trunk Group ID** A desired ID number
- **Trunk Group Type** **T2**
- **Trunk Group Name** A desired name

Click **ctrl+v** to continue.

```
+--Create: Trunk Groups-----+
|
|      Node Number (reserved) : 1
|      Trunk Group ID : 10
|
|      Trunk Group Type + T2
|      Trunk Group Name : To MX
|      UTF-8 Trunk Group Name : -----
|      Number Compatible With : -1
|      Remote Network : 255
|      Shared Trunk Group + False
|      Special Services + Nothing
|
+-----+
```

On the next screen complete the following options:

- **Q931 Signal Variant** **ABC-F**
- **T2 Specification** **SIP**

Click **ctrl+v** to complete configuration.

```
+--Create: Trunk Groups-----+
|
|      Node number : 1
|      Transcom Trunk Group + False
|      Auto.reserv.by Attendant + False
|      Overflow trunk group No. : -1
|      Tone on seizure + False
|      Private Trunk Group + False
|      Q931 Signal variant + ABC-F
|      SS7 Signal variant + No variant
|      Number Of Digits To Send : 0
|      Channel selection type + Quantified
|      Auto.DTMF dialing on outgoing call + NO
|      T2 Specification + SIP
|      Homogenous network for direct RTP + NO
|      Public Network COS : 0
|      DID transcoding + False
|      Can support UUS in SETUP + True
|
|      Implicit Priority
|
|      Activation mode : 0
|      Priority Level : 0
|
|      Preempter + NO
|      Incoming calls Restriction COS : 10
|      Outgoing calls Restriction COS : 10
|      Callee number mpt1343 + NO
|      Overlap dialing + YES
|      Call diversion in ISDN + NO
|
+-----+
```

2.5. Administer SIP Gateway

To configure a SIP Gateway select **SIP → SIP Gateway**. Complete the following options:

- **SIP Trunk Group** SIP trunk group number defined in **Section 24**
- **DNS Local Domain Name** Enter domain name for the Alcatel OXE
- **SIP Proxy Port Number** **5060**

Click **ctrl+v** to complete.

```
+--Review/Modify: SIP Gateway-----+
|
|      Node Number (reserved) : 1
|      Instance (reserved) : 1
|      Instance (reserved) : 1
|
|      SIP Subnetwork : 9
|      SIP Trunk Group : 10
|      IP Address : 10.10.9.111
|      Machine name - Host : node1
|      SIP Proxy Port Number : 5060
|      SIP Subscribe Min Duration : 1800
|      SIP Subscribe Max Duration : 86400
|      Session Timer : 1800
|      Min Session Timer : 1800
|      Session Timer Method : RE_INVITE
|      DNS local domain name : mmsil.local
|      DNS type : DNS A
|      SIP DNS1 IP Address : -----
|      SIP DNS2 IP Address : -----
|      SDP in 18x : False
|      Cac SIP-SIP : False
|      INFO method for remote extension : True
|      Dynamic Payload type for DTMF : 97
|
+-----+
```

2.6. Administer SIP Proxy

To configure a SIP Proxy select **SIP → SIP Proxy**. Complete the following options:

- **Minimal authentication method** **SIP None**

Click **ctrl+v** to complete.

```
+--Review/Modify: SIP Proxy-----+
|
|      Node Number (reserved) : 1
|      Instance (reserved) : 1
|      Instance (reserved) : 1
|      SIP initial time-out : 500
|      SIP timer T2 : 4000
|      Dns Timer overflow : 5000
|      Recursive search : False
|      Minimal authentication method : SIP None
|
|      Authentication realm : -----
|      Only authenticated incoming calls : False
|      Framework Period : 3
|      Framework Nb Message By Period : 25
|      Framework Quarantine Period : 1800
|      TCP when long messages : True
|
+-----+
```

2.7. Administer SIP External Gateway

Configure a SIP connection to the Meeting Exchange by creating a SIP External Gateway. Select **SIP → SIP Ext Gateway → Create**. Complete the following options:

- **SIP External Gateway ID** A desired ID number
- **Gateway Name** A desired name
- **SIP Remote domain** Enter the MX ip address
- **SIP Port Number** **5060**
- **SIP Transport Type** **TCP**
- **Trunk Group Number** The trunk group number defined in **Section 2.4**
- **Minimal authentication method** **SIP None**

Click **ctrl+v** to complete.

```
+--Create: SIP Ext Gateway-----+
|
|   Node Number (reserved) : 1
|   Instance (reserved) : 1
|   SIP External Gateway ID : 0
|
|       Gateway Name : MX
|       SIP Remote domain : 10.10.21.51
|       PCS IP Address : -----
|       SIP Port Number : 5060
|       SIP Transport Type + TCP
|       RFC3262 Forced use + True
|       Belonging Domain : -----
|       Registration ID : -----
|   Registration ID P_Asserted + False
|       Registration timer : 0
|       SIP Outbound Proxy : -----
|       Supervision timer : 0
|       Trunk group number : 10
|       Pool Number : -1
|       Outgoing realm : -----
|       Outgoing username : -----
|
|       Outgoing Password : -----
|       Confirm : -----
|
|       Incoming username : -----
|       Incoming Password : -----
|       Confirm : -----
|
|   RFC 3325 supported by the distant + True
|       DNS type + DNS A
|       SIP DNS1 IP Address : -----
|       SIP DNS2 IP Address : -----
|       SDP in 18x + False
|   Minimal authentication method + SIP None
|   INFO method for remote extension + False
|       Send only trunk group algo + False
|           To EMS + False
|       Routing Application + False
|       Dynamic Payload type for DTMF : 97
+-----+
```

2.8. Administer Network Routing Table

In the sample configuration, network number 15 was used. To administer the routing table for network number 15, select **Translator → Network Routing Table** and then select **15**. Complete the following options:

- **Associated Ext SIP gateway** Use the SIP External Gateway ID defined in **Section 2.7**

Click **ctrl+v** to complete.

```
+--Review/Modify: Network Routing Table-----+
|
|      Node Number (reserved) : 1
|      Instance (reserved) : 1
|      Network Number : 15
|
|      Rank of First Digit to be Sent : 1
|      Incoming identification prefix : -----
|      Protocol Type + ABC_F
|      Numbering Plan Descriptor ID : 11
|      ARS Route list : 0
|      Schedule number : -1
|      ATM Address ID : -1
|      Network call prefix : -----
|      City/Town Name : -----
|      Send City/Town Name + False
|      Associated Ext SIP gateway : 0
|      Enable UTF8 name sending + True
|
+-----+
```

2.9. Administer Prefix Plan

In the sample configuration, MX conference numbers are 5 digits in length and begin with 3888. To administer the prefix plan for dialing into conferences from Alcatel OXE, select **Translator → Prefix Plan → Create**. Complete the following options:

- **Number** **3888**
- **Prefix Meaning** **Routing No**

Click **ctrl+v** to continue.

```
+--Create: Prefix Plan-----+
|
|      Node Number (reserved) : 1
|      Instance (reserved) : 1
|      Number : 3888
|
|      Prefix Meaning + Routing No.
|
+-----+
```

On the next screen complete the following options:

- **Network Number** Use network number administered in **Section 2.8**
- **Node Number/ABC-F Trunk Group**
 Use the trunk group number administered in **Section 2.4**
- **Number of Digits** 5

Click **ctrl+v** to complete.

```
+--Create: Prefix Plan-----+
|
|      Network Number : 15
|      Node Number/ABC-F Trunk Group : 10
|      Number of Digits : 5
|      Number With Subaddress (ISDN) + NO
|      Default X25 ID.pref. + NO
|
+--+
```

2.10. Administer Codec on SIP Trunk Group

To create a codec on the SIP Trunk Group select **Trunk Groups → Trunk Group**. The parameter **IP Compression Type** has two possible values, G711 and Default. If the parameter **Default** is chosen then this value is determined by the parameter **Compression Type** administered in **System → Other System Param. → Compression Parameters**. Compression type is either G.729 or G.723.

```
+--Review/Modify: Compression Parameters-----+
|
|      Node Number (reserved) : 1
|      Instance (reserved) : 1
|      Instance (reserved) : 1
|      System Option + Compression Type
|
|      Compression Type + G 729
|
+--+
```

For the above values to hold true, all other options for compression in the Alcatel OXE must be set to non-compressed options. Ensure the following parameters are set accordingly:

Navigate to **IP → IP Domain**

- Intra-Domain Coding Algorithm = default
- Extra-Domain Coding Algorithm = default

Navigate to **IP → TSC/IP**

- Default Voice Coding Algorithm = without compression

Navigate to **IP → INT/IP**

- Default Voice Coding Algorithm = without compression

3. Configure Avaya Meeting Exchange™ Enterprise

This section describes the steps for configuring the Meeting Exchange to interoperate with Alcatel OmniPCX Enterprise via SIP trunking. It is assumed that the Meeting Exchange is installed and licensed as described in the product documentation (see reference [1]). The following steps describe the administrative procedures for configuring Meeting Exchange:

- Configure SIP Connectivity
- Configure Dialout
- Map DNIS Entries
- Configure Audio Preferences
- Restarting the Meeting Exchange server
- Configure Bridge Talk

The following instructions require logging in to the Meeting Exchange console using an ssh connection to access the Command Line Interface (CLI) with the appropriate credentials.

3.1. Configuring SIP Connectivity

Log in to the Meeting Exchange server console using an ssh Client to access the Command Line Interface (CLI) with the appropriate credentials. Configure settings that enable SIP connectivity between the Meeting Exchange server and other devices by editing the **system.cfg** file as follows:

- Edit **/usr/ipcb/config/system.cfg**
- Add Meeting Exchange server IP address
 - o **IPAddress=(10.10.21.51)**
- Depending on the SIP signalling protocol, TCP or UDP, add one of the following lines to populate the From Header Field in SIP INVITE messages:
 - o **MyListener=<sip:6000@10.10.21.51:5060;transport=tcp>**
 - o **MyListener=<sip:6000@10.10.21.51:5060;transport=udp>**

Note: The user field 6000, defined for this SIP URI must conform to RFC 3261. For consistency, it is selected to match the user field provisioned for the **respContact** entry (see below).
- Depending on the SIP signalling protocol, TCP or UDP , add one of the following lines to provide SIP Device Contact address to use for acknowledging SIP messages from the Meeting Exchange server:
 - o **respContact=<sip:6000@10.10.21.51:5060;transport=tcp>**
 - o **respContact=<sip:6000@10.10.21.51:5060;transport=udp>**
- Add the following lines to set the Min-SE timer to **900** seconds in SIP INVITE messages from the Meeting Exchange server:
 - o **sessionRefreshTimerValue= 900**
 - o **minSETimerValue= 900**

3.2. Configure Dialout

To enable Dial-Out from the Meeting Exchange to Alcatel OXE, edit the **telnumToUri.tab** file as follows:

- Edit **/usr/ipcb/config/telnumToUri.tab** file with a text editor
- Add the following line to the file to route outbound calls from the Meeting Exchange to the Alcatel OXE.
* **sip:\$0@10.10.9.111:5060;transport=tcp**

3.3. Map DNIS Entries

The DNIS entry is the number dialed by Alcatel subscribers to access a conference on Meeting Exchange. The DNIS entry needs to be mapped on Meeting Exchange to enable access to a conference. To map DNIS entries, run the **cbutil** utility on Meeting Exchange. Log in to the Meeting Exchange with a ssh connection with the appropriate credentials. Enable Dial-In access (via passcode) to conferences provisioned on the Meeting Exchange as follows:

- Add a DNIS entry for a **scan call function** corresponding to DID **38888** by entering the following command at the command prompt:
cbutil add <dnis> <rg> <msg> <ps> <ucps> <func> [-o <of> -l <ln> -c <cn> -crs <n> -cre <n> -cc <code>]

where the variables for add command is defined as follows:

- o **<dnis>** DNIS
- o **<rg>** Reservation Group
- o **<msg>** Annunciator message number
- o **<ps>** Prompt Set number (0-20)
- o **<ucps>** Use Conference Prompt Set (y/n)
- o **<func>** One of: DIRECT/SCAN/ENTER/HANGUP/AUTOVL/FLEX
- o **-o <of>** Optional On-failure function – one of: ENTER/HANGUP
- o **-l <"ln">** Optional line name to associate with caller
- o **-c <"cn">** Optional company name to associate with caller
- o **-crs <n>** Optional conference room start number
- o **-cre <n>** Optional conference room end number

In this sample configuration, the DNIS entry for a **scan call function** was added corresponding to DNIS 38888 by entering the following command at the command prompt:

```
[MXSIL]# cbutil add 38888 0 247 1 N SCAN
cbutil
Copyright 2004 Avaya, Inc. All rights reserved.
```

At the command prompt, enter **cbutil list** to verify the DNIS entries provisioned.

```
[MXSIL]# cbutil list
cbutil
Copyright 2004 Avaya, Inc. All rights reserved.

DNIS   Grp Msg PS  CP Function On Failure Line Name Company Name Room Start
-----
38888  0   247  1   N  SCAN      DEFAULT
```

3.4. Configure Audio Preferences file

The **audioPreferences.cfg** file located at **/usr/ipcb/config/** specifies the order in which codecs are offered in the Session Description Protocol. Set the **telephone-event** value to **payloadType** of **97**.

```
# audioPreferences.cfg
# This table is an ordered list of MIME subtypes specifying the codecs
# supported
# by this media server. The list is specified in the order in which an SDP
# offer
# will list the various MIME subtypes on the m=audio line.
# For static payload type numbers (i.e. numbers between 0 - 96) please use the
# iana registered numbering scheme.
# See: http://www.iana.org/assignments/rtp-parameters
mimeSubtype      payloadType
PCMU              0
# PCMA            8
# G722            9
G729             18
# iLBC20         98
# wbPCMU        102
# wbPCMA        103
telephone-event  97
# iSAC          104
# G726_16       105
# G726_24       106
# G726_32       107
# G726_40       108
```

3.5. Restarting the Meeting Exchange Server

After the configuration changes are made, restart the services issuing the command **service mx-bridge restart**

```
# service mx-bridge restart
/etc/init.d/mx-bridge: Restarting bridge
/etc/init.d/mx-bridge: Server type is DCB
/etc/init.d/mx-bridge: Stopping DCB conferencing server bridge via uninitdcb.sh
Stopping notificationCtrlServer service:
killproc notificationCtrlServer
[ OK ]
Sending CMD_SHUTDOWN level 3 message to the INIT_KEY queue.
Waiting for 6 processes to stop
Waiting for 2 processes to stop
Waiting for 1 processes to stop
Waiting for 1 processes to stop
destroy.
/etc/init.d/mx-bridge: mx-bridge startup
/etc/init.d/mx-bridge: Server type is DCB
.....
.....
.....
Add Process Key 145 IP address 10.10.6.20
Add Process Key 146 IP address 10.10.6.20
key ID 101
key ID 102
key ID 110
===== INITDCB
=====
FirstMusic = 3199.
FirstLink = 3199.
FirstRP = 3198.
FirstOper = 3195.
numUserLCNs = 3195.
```

3.6. Bridge Talk

The following steps utilize the Avaya Bridge Talk application to provision a sample conference on the Meeting Exchange. This sample conference enables both Dial-In and Dial-Out access to audio conferencing for endpoints on the Public Switched Telephone Network.

Notes: If any of the features displayed in the Avaya Bridge Talk screen captures are not present, contact an authorized Avaya Sales representative to make the appropriate changes.

3.6.1. Initializing Bridge Talk

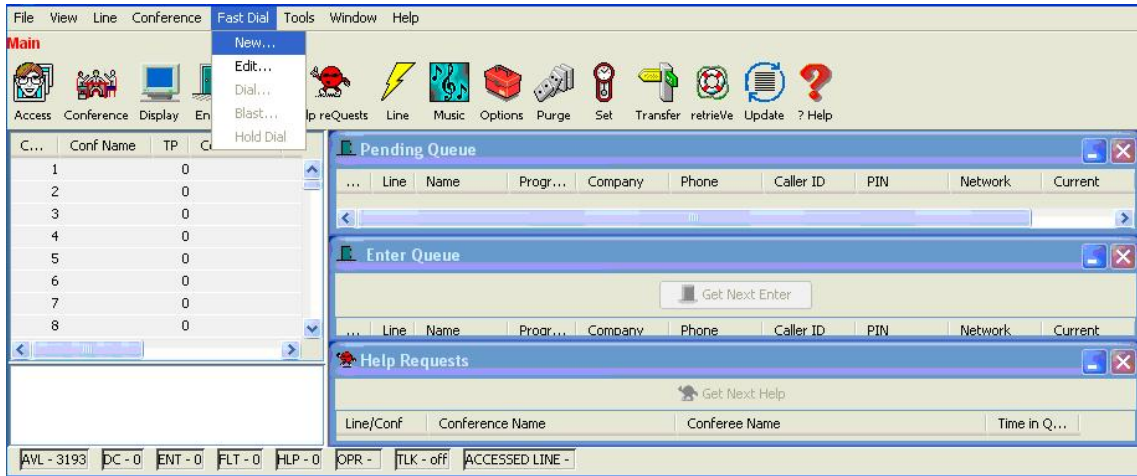
Invoke the Avaya Bridge Talk application as follows:

- Double-click on the desktop icon from a Personal Computer loaded with the Avaya Bridge Talk application and with network connectivity to the Meeting Exchange (Not shown).
- Enter the appropriate credentials in the **Sign-In** and **Password** fields.
- Enter the IP address of the Meeting Exchange server (**10.10.21.51** for this sample configuration) in the **Bridge** field as shown below.

A screenshot of the 'Avaya Bridge Talk login' dialog box. The dialog has a blue title bar with the text 'Avaya Bridge Talk login' and a red close button. The main area is light yellow and contains four labeled input fields: 'Sign-In:' with the text 'tp', 'Password:' which is empty, 'Bridge:' with the IP address '10.10.21.51' and a dropdown arrow, and 'Operator:' with the text 'Next available' and a dropdown arrow. At the bottom are two buttons: 'OK' and 'Exit'.

3.6.2. Creating a Dial Out list

Provision a dial list that is utilized for Dial-Out (e.g., Blast dial and Fast dial) from the Meeting Exchange. From the Avaya Bridge Talk Menu Bar, click **Fast Dial** → **New**.

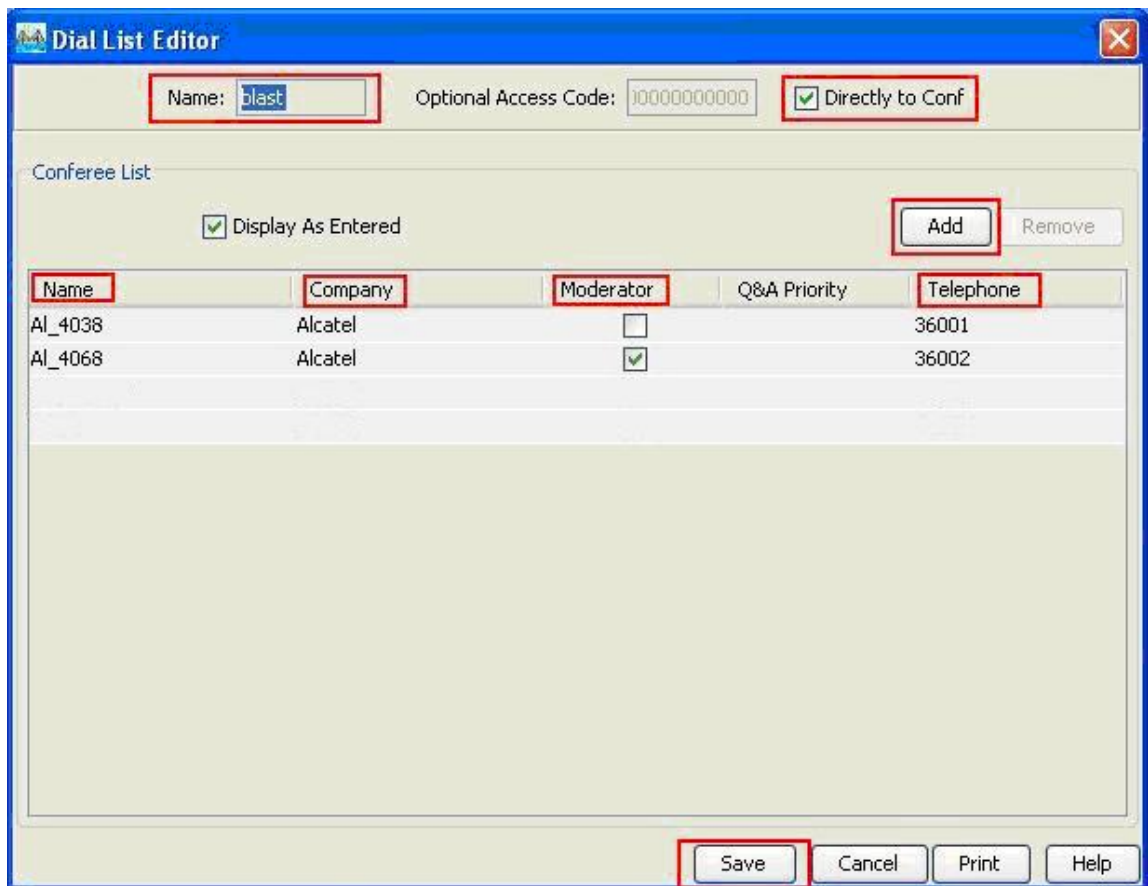


3.6.3. Creating a Dial List

From the **Dial List Editor** window that is displayed below:

- Enter a descriptive label in the **Name** field.
- Enable conference participants on the dial list to enter the conference without a passcode by selecting the **Directly to Conf** box as displayed.
- Add entries to the dial list by clicking on the **Add** button and enter **Name**, **Company** and **Telephone** number for dial out for each participant. [Optional] Moderator privileges may be granted to a conference participant by checking the **Moderator** box.

When finished, click on the **Save** button on the bottom of the screen.

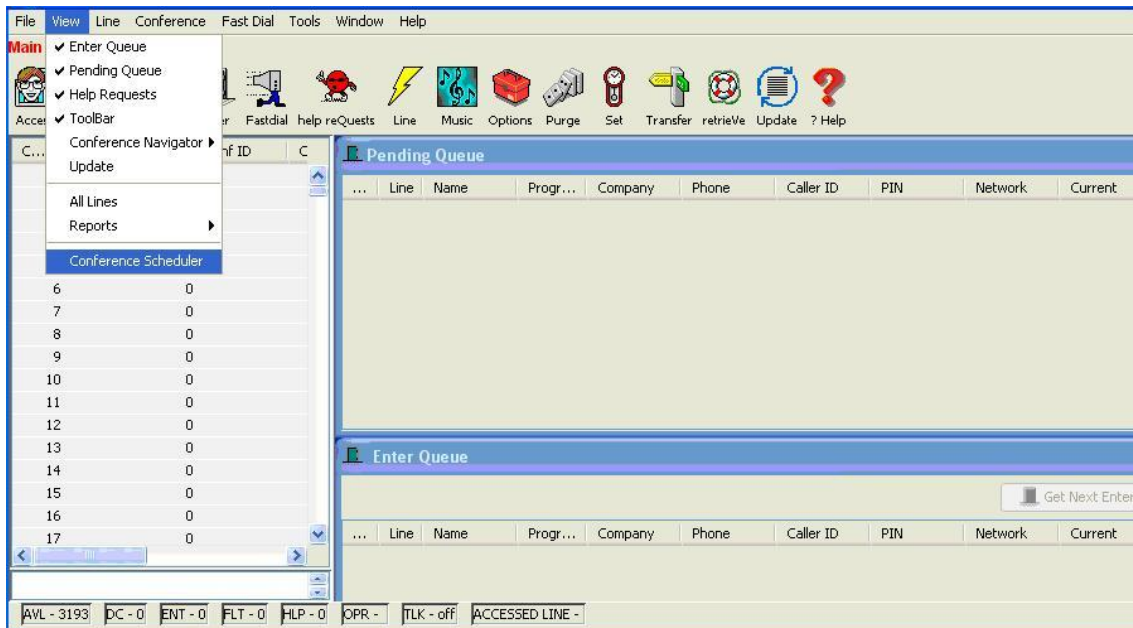


The screenshot shows the 'Dial List Editor' window. At the top, there is a 'Name' field containing 'blast', an 'Optional Access Code' field with '0000000000', and a checked 'Directly to Conf' checkbox. Below this is a 'Conferee List' section with a checked 'Display As Entered' checkbox and 'Add' and 'Remove' buttons. A table with five columns is displayed: 'Name', 'Company', 'Moderator', 'Q&A Priority', and 'Telephone'. The table contains two rows of data. The 'Save' button is highlighted at the bottom.

Name	Company	Moderator	Q&A Priority	Telephone
Al_4038	Alcatel	☐		36001
Al_4068	Alcatel	☒		36002

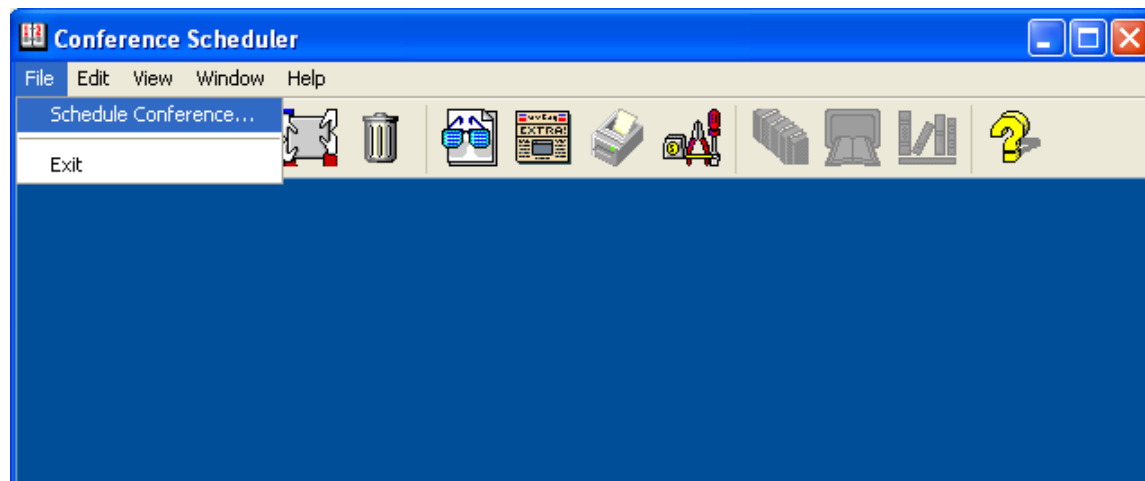
3.6.4. Conference Scheduler

From the **Avaya Bridge Talk** menu bar, click **View → Conference Scheduler** to provision a conference.



3.6.5. Scheduling a Conference

From the **Conference Scheduler** window, click **File → Schedule Conference**.



3.6.6. Provision a Conference

From the **Schedule Conference** window that is displayed, provision a conference as follows:

- Enter a unique **Conferee Code** to allow participants access to this conference.
- Enter a unique **Moderator Code** to allow participants access to this conference with moderator privileges.
- Enter a descriptive label in the **Conference Name** field.
- Administer settings to enable an **Auto Blast** dial by setting Auto/Manual as desired.

Select a dial list by clicking on the **Dial List** button, select a dial list from the **Create, Select or Edit Dial List** window that is displayed (not shown), and click on the **Select** button (to verify Dial out and Blast Dial out).

- When finished, click on the **OK** button on the bottom of the screen.

Schedule Conference [Administrator Access]

Conference Information

Status:	ENABLED	Mode:	UNATTENDED	Conference Type:	DAILY
Confirmation No.:		Conference ID:		Weekend:	YES
Name:		Billing Code Prompt:	DISABLED		
Telephone:		Accounting Code:	OFF	Start Date (dd/mm/yyyy):	09/12/2009
Sign-in Name:	md	Security Passcode:	OFF	End Date (dd/mm/yyyy):	09/12/2009
Res Group:	0	Change Conf Opt:	ON		
Conferee Code:	11111	Op Help Available:	ON	Name Record/Play:	OFF
Moderator Code:	22222	Block Dialout:	OFF	NRP Annunciator:	Browse
Conference Name:	Test1	Auto Blast:	Auto	PIN Mode:	OFF
Dial List:	blast	Blast Annunciator:	Browse	PIN List:	

Conference Features

Start Time:	00:00	End Time:	00:00	Code Duration:	0
Entry Tone:	Tone & Message	Exit Tone:	Tone & Message	Maximum Lines:	6
Hang up:	OFF	Music:	OFF	Security:	OFF
Auto Extend Duration:	ON	Auto Extend Ports:	ON		
Prompt Set:	English	Conference Viewer:	NO		

OK Cancel Help

4. Verification

This section provides the verification tests that can be performed on Alcatel OmniPCX Enterprise and Meeting Exchange to verify their proper configuration.

4.1. Verify Alcatel OmniPCX Enterprise

Verify the status of the SIP trunk group by using the **trkstat n** command, where **n** is the trunk group number being investigated. Verify that all trunks are in the **Free** state as shown below.

```
trkstat 10
```

=====+																					
	S	I	P	T	R	U	N	K	S	T	A	T	E								
	Trunk group number : 10																				
	Trunk group name : To ASM60																				
	Number of Trunks : 62																				
-----+																					
	Index :	1	2	3	4	5	6	7	8	9	10	11	12	13							
	State :	F	F	F	F	F	F	F	F	F	F	F	F	F							
-----+																					
	Index :	14	15	16	17	18	19	20	21	22	23	24	25	26							
	State :	F	F	F	F	F	F	F	F	F	F	F	F	F							
-----+																					
	Index :	27	28	29	30	31	32	33	34	35	36	37	38	39							
	State :	F	F	F	F	F	F	F	F	F	F	F	F	F							
-----+																					
	Index :	40	41	42	43	44	45	46	47	48	49	50	51	52							
	State :	F	F	F	F	F	F	F	F	F	F	F	F	F							
-----+																					
	Index :	53	54	55	56	57	58	59	60	61	62										
	State :	F	F	F	F	F	F	F	F	F	F										
-----+																					
	F: Free		B: Busy		Ct: busy Comp trunk						Cl: busy Comp link										
	WB: Busy Without B Channel					Cr: busy Comp trunk for RLIO inter-ACT link															
	WBD: Data Transparency without chan. WBM: Modem transparency without chan.																				
	D: Data Transparency					M: Modem transparency															
-----+																					

4.2. Verify Avaya Meeting Exchange™ Enterprise

Verify all conferencing related processes are running on the Meeting Exchange as follows:

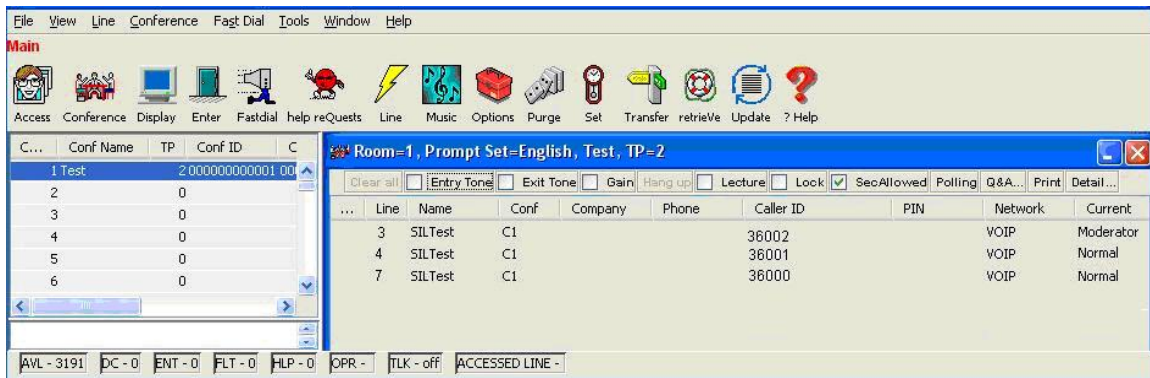
- Log in to the Meeting Exchange server console to access the CLI with the appropriate credentials.
- cd to **/usr/dcb/bin**
- At the command prompt, run the script **service mx-bridge status** and confirm all processes are running by verifying an associated 4-digit Process ID (PID) for each process.

```
# service mx-bridge status
5042 ?      00:00:01 initdcb
5604 ?      00:00:00 log
5607 ?      00:00:00 bridgeTranslato
5608 ?      00:00:00 netservices
5626 ?      00:00:00 timer
5627 ?      00:00:00 traffic
5628 ?      00:00:00 chdbased
5629 ?      00:00:00 startd
5630 ?      00:00:00 cdr
5631 ?      00:00:00 modapid
5632 ?      00:00:00 schapid
5633 ?      00:00:01 callhand
5634 ?      00:00:00 initipcb
5644 ?      00:00:00 sipagent
5645 ?      00:00:00 msdispatcher
5646 ?      00:00:00 serverComms
5648 ?      00:00:00 softms
5649 ?      00:00:00 softms
5650 ?      00:00:00 softms
5651 ?      00:00:00 softms
5652 ?      00:00:00 softms
5653 ?      00:00:00 softms
4022 ?      00:00:00 postmaster with 27 children
```

4.2.1. Verify Call Routing

Verify end to end signaling/media connectivity between the Meeting Exchange and Alcatel OXE. This is accomplished by placing calls from Alcatel end points to the Meeting Exchange. This step utilizes the Avaya Bridge Talk application to verify calls to and from the Meeting Exchange are managed correctly, e.g., callers are added/removed from conferences. This step will also verify the conferencing applications provisioned.

- Configure a conference with Auto Blast enabled and provision a dial list. From an Alcatel endpoint, dial a number that corresponds to DNIS **38888** to enter a conference as **Moderator** (with passcode) and blast dial is invoked automatically. When answered these callers enter the conference.
- If not already logged on, log in to the Avaya Bridge Talk application with the appropriate credentials
- **Double-Click on the highlighted Conf #** to open a **Conference Room** window
- Verify conference participants are added/removed from conferences by observing the Conference Navigator and/or Conference Room windows.



4.3. Verified Scenarios

The following scenarios have been verified for the configuration described in these Application Notes.

- Conference calls including various telephone types (see **Figure 1**) on the Alcatel OmniPCX Enterprise can be made using G.711mu/A-law and G.729.
- Scan, Flex, and Direct Conference modes.
- Name Record/Play (NRP).
- RFC 2833 DTMF support for all moderator and conferee commands.
- Manual and automatic blast dial-out to conference participants.
- Network outage failure and recovery.
- Bridge process ("softms") failure and recovery.
- Session timers on Meeting Exchange.
- Line and Conference transfer

5. Conclusion

As illustrated in these Application Notes, Alcatel OmniPCX Enterprise can interoperate with Avaya Meeting Exchange™ Enterprise Edition using a SIP trunk.

6. Additional References

Product documentation for Avaya products may be found at <http://support.avaya.com>

- [1] *Administering Meeting Exchange™ 5.2 Service Pack 1 Servers*, Doc # 04-603548, Issue 1 Release 5.2.1

Product documentation for Alcatel products may be found at:

- [2] <http://enterprise.alcatellucent.com/?product=OmniPCXEnterprise&page=overview>
- [3] <http://enterprise.alcatel-lucent.com/?dept=ResourceLibrary&page=Landing>

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