



Avaya Solution & Interoperability Test Lab

Application Notes for configuring MiContact Center Enterprise from Mitel Networks Corporation to interoperate with Avaya IP Office R10.1 - Issue 1.0

Abstract

These Application Notes describe the configuration steps required for Mitel MiContact Center Enterprise (MiCC) to interoperate with Avaya IP Office.

Readers should pay attention to **Section 2**, in particular the scope of testing as outlined in **Section 2.1** as well as the observations noted in **Section 2.2**, to ensure that their own use cases are adequately covered by this scope and results.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

1. Introduction

These Application Notes describe the configuration steps required for MiContact Center Enterprise (MiCC) from Mitel Networks Corporation to interoperate with Avaya IP Office R10.1. The Avaya IP Office consists of an Avaya IP Office Server Edition R10.1 running on a virtual platform as the Primary Server with an Avaya IP Office IP500V2 R10.1 as the secondary expansion sever.

MiCC is built for the MiVoice MX-ONE platform and also supports 3rd party communication servers such as the Avaya IP Office. MiCC is an all-in-one contact center with a single software stream for seamless growth, feature extension and deployment flexibility, supporting up to 1,500 agents on a single system and scales to 15,000 concurrent agents in a network environment.

Additional modules include the following which were also tested as part of the overall solution.

- MiContact Center Enterprise Campaign Manager
- MiContact Center Enterprise Outbound
- MiContact Center Enterprise Call Recording.

2. General Test Approach and Test Results

The general test approach was to configure the MiCC to communicate with the Avaya IP Office as implemented on a customer's premises using a SIP connection. Testing focused on verifying that MiCC connected with IP Office over the SIP trunk and all features behaved as expected. Various call scenarios outlined in **Section 2.1** were performed to simulate real call types as would be observed on a customer premises. See **Figure 1** for a network diagram. The interoperability compliance test included both feature functionality and serviceability tests.

Agents use the Mitel MiCC Agent application and can either log into a Mitel MiCC softphone or use the MiCC Agent application to log into an IP Office hardphone.

Testing using the Mitel MiCC softphone involved making calls to a number that was routed from the IP Office to MiCC. These calls were routed to a script that route calls to the MiCC agent. Calls may be made from a simulated PSTN which are then routed to the MiCC and are routed internally by the MiCC and answered by the MiCC Agent application.

Testing with MiCC agent logged into the IP Office hardphone involved making calls to a number that is routed from the IP Office to MiCC. These calls are routed to a script that routed calls to the MiCC agent. Calls may be made from a simulated PSTN which are then routed to the MiCC via the SIP trunk. The MiCC agent using the IP Office hardphone is then called by the MiCC and once that call is answered the customer call is then connected to the IP Office hardphone. Once the call is established the agent can then transfer/conference/hold/ the call using MiCC Agent application.

DevConnect Compliance Testing is conducted jointly by Avaya and DevConnect members. The jointly-defined test plan focuses on exercising APIs and/or standards-based interfaces pertinent to the interoperability of the tested products and their functionalities. DevConnect Compliance Testing is not intended to substitute full product performance or feature testing performed by DevConnect members, nor is it to be construed as an endorsement by Avaya of the suitability or completeness of a DevConnect member's solution.

Avaya recommends our customers implement Avaya solutions using appropriate security and encryption capabilities enabled by our products. The testing referenced in these DevConnect Application Notes included the enablement of supported encryption capabilities in the Avaya products. Readers should consult the appropriate Avaya product documentation for further information regarding security and encryption capabilities supported by those Avaya products.

Support for these security and encryption capabilities in any non-Avaya solution component is the responsibility of each individual vendor. Readers should consult the appropriate vendor-supplied product documentation for more information regarding those products.

For the testing associated with these Application Notes, the interface between Avaya systems and MiCC did not include use of any specific encryption features as requested by Mitel.

2.1. Interoperability Compliance Testing

The testing included:

- Logging an agent into the MiCC Agent application MiCC softphone
- Logging an agent into the MiCC Agent application using the IP Office physical phone
- Making inbound calls to the MiCC Agent using skill based routing
- Supervised and unsupervised transfer
- Conference calls
- Features such as call back, outbound campaigns and call recording
- Serviceability testing which included a simulated LAN failure from the MiCC server and Agent applications

2.2. Test Results

Tests were performed to insure full interoperability of the Mitel solution as a whole with the IP Office using a SIP Trunk connection. The tests were all functional in nature and performance testing was not included. All test cases that were executed passed successfully.

2.3. Support

Technical support from Mitel can be obtained through the following:

Web: www.Mitel.com/service-and-support

3. Reference Configuration

Figure 1 shows the network topology during compliance testing. MiCC is installed on a Windows Server 2012R2 operating system. Microsoft SQL was also installed on the same server. (SQL may also be installed on a separate server). The MiCC was configured to connect to the IP Office Server Edition and tested using both the MiCC softphone and using IP Office endpoints from both the IP Office Server Edition and IP Office IP500V2. Once testing was completed with the MiCC connected to the Server Edition it was then configured to connect to the IP Office IP500V2 over a SIP trunk and tested using both the MiCC softphone and using IP Office endpoints from both the IP Office Server Edition and IP Office IP500V2.

Note: Two separate SIP connections to the IP Office Server Edition and the IP Office IP500V2 were tested.

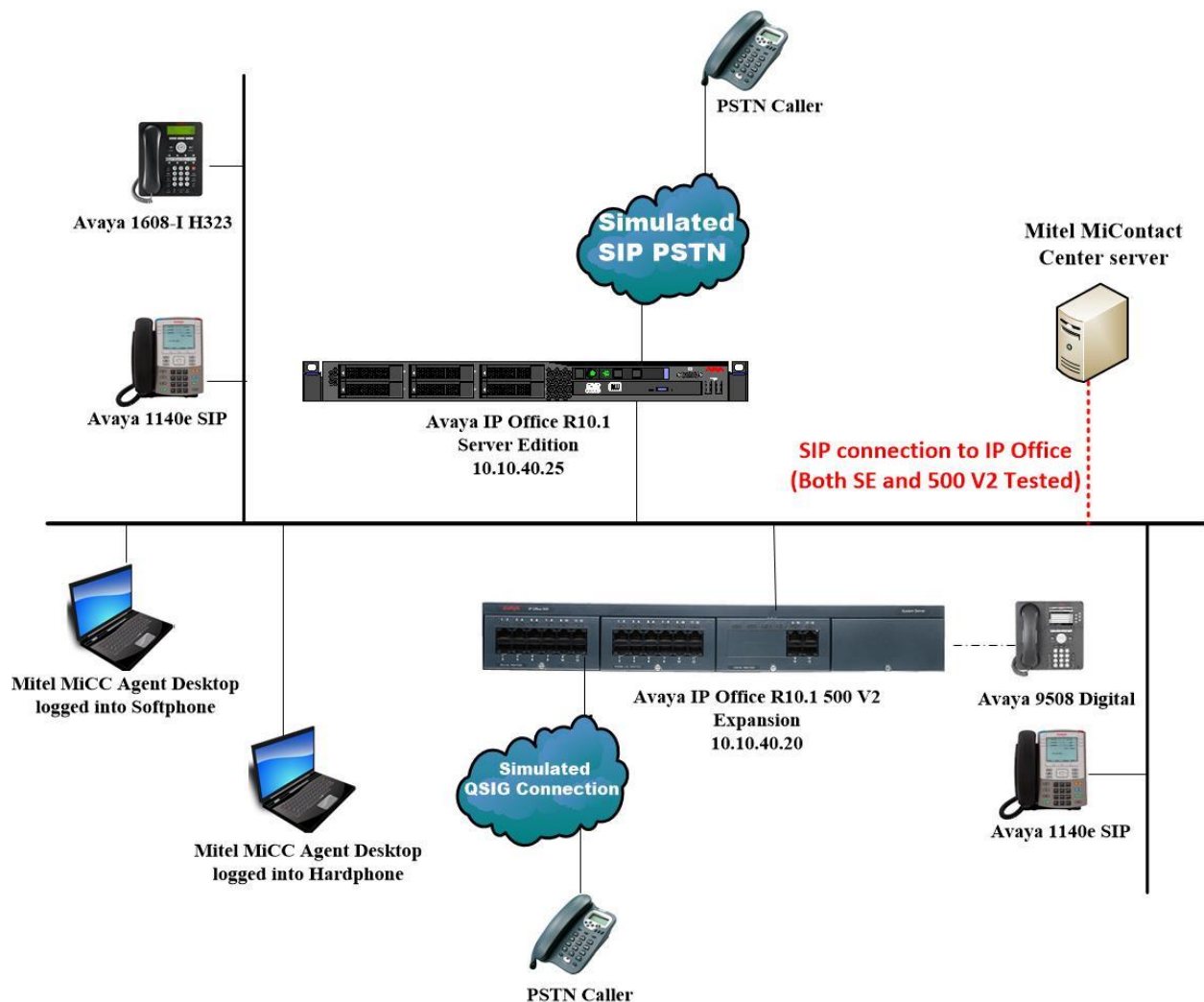


Figure 1: Network Solution of Mitel MiContact Center Enterprise with Avaya IP Office Server Edition and IP Office IP500 V2 R10.1

4. Equipment and Software Validated

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

Equipment/Software	Release/Version
Avaya IP Office Primary Server Server Edition running on a Virtual Platform	R10.1.0.0 Build 237
Avaya IP Office Secondary Expansion Avaya IP Office 500 V2	R10.1.0.0 Build 237
Avaya IP Office Manager running on a Windows 7 PC	R10.1.0.0 Build 237
Avaya 1608-I H323 Deskphone	1608UA1_350B.bin
Avaya 9608 H323 Deskphone	R6.6401
Avaya 1140e SIP Deskphone	R04.04.28.00
Avaya 9508 Digital Deskphone	R0.60
Mitel MiContact Center Enterprise running on a Windows 2012 Server	MiContact Center Enterprise 9.2 SP2
Mitel MiContact Center Enterprise Agent application running on a Windows 7 PC	MiContact Center Enterprise Agent 9.2 SP2

Note: Compliance Testing is applicable when the tested solution is deployed with a standalone IP Office 500 V2 and also when deployed with IP Office Server Edition in all configurations.

5. Configure Avaya IP Office

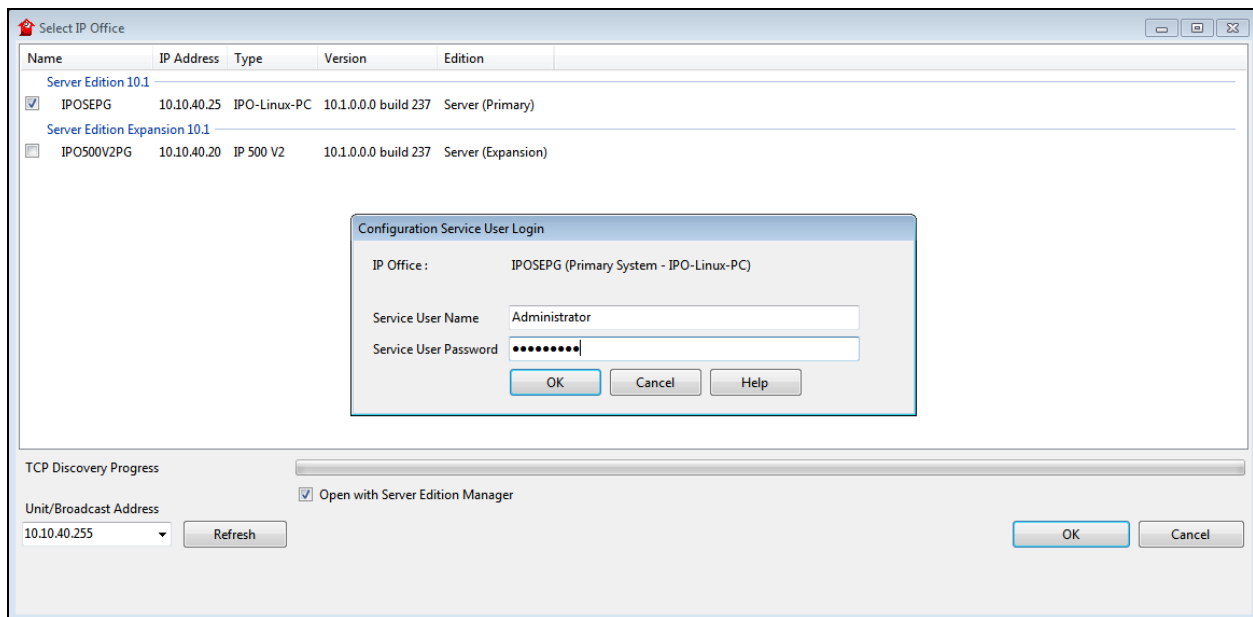
It is assumed that a fully functioning IP Office is in place with the necessary licensing. The configuration and verification operations illustrated in this section were all performed using Avaya IP Office Manager. The information provided in this section describes the configuration of IP Office for this solution. For all other provisioning information such as initial installation and configuration, please refer to the product documentation in **Section 9**. The configuration operations described in this section can be summarized as follows:

- Launch Avaya IP Office Manager
- Display LAN Configuration
- Configure SIP Trunks
- Configure Short Codes
- Save Configuration

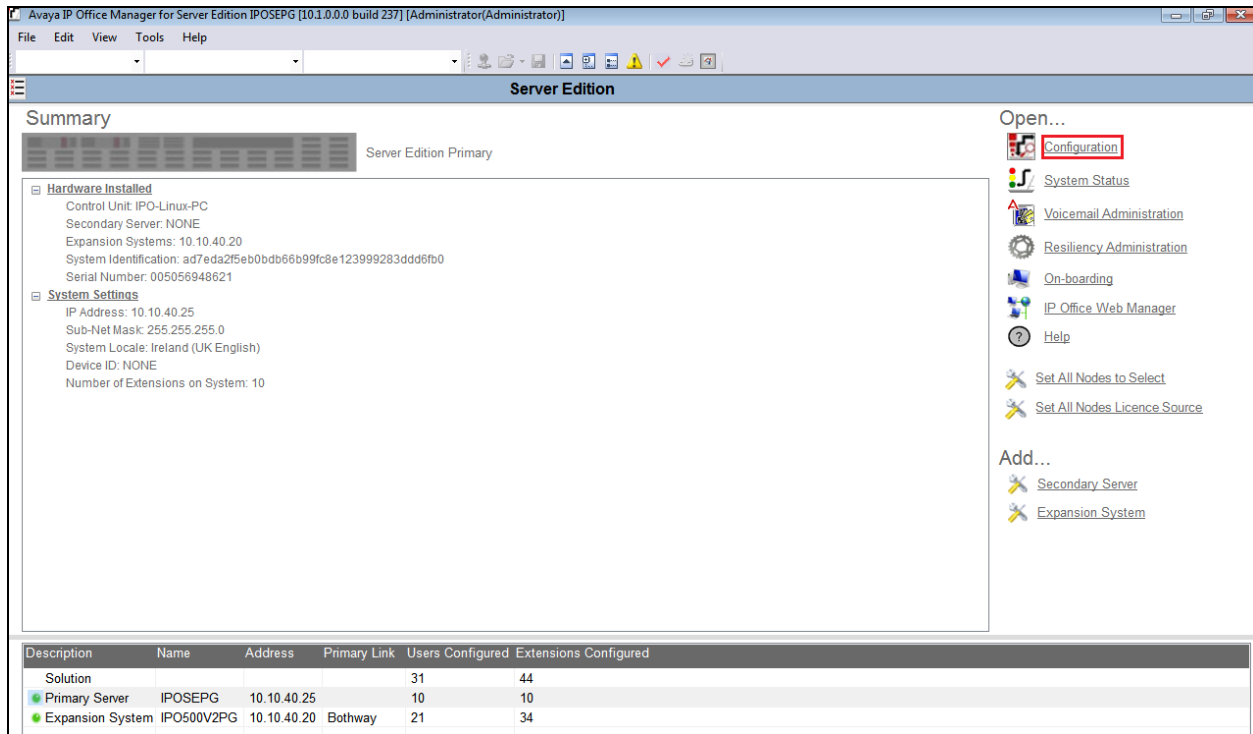
Note: The configuration of PSTN trunks and routes are outside the scope of these Application Notes.

5.1. Launch Avaya IP Office Manager (Administration)

From the IP Office Manager PC, click **Start** → **Programs** → **IP Office** → **Manager** to launch the Manager application (not shown). Select the appropriate IP Office (in the example below the IP Office Server Edition is chosen). Enter the appropriate credentials and click on the **OK** button to receive the IP Office configuration.

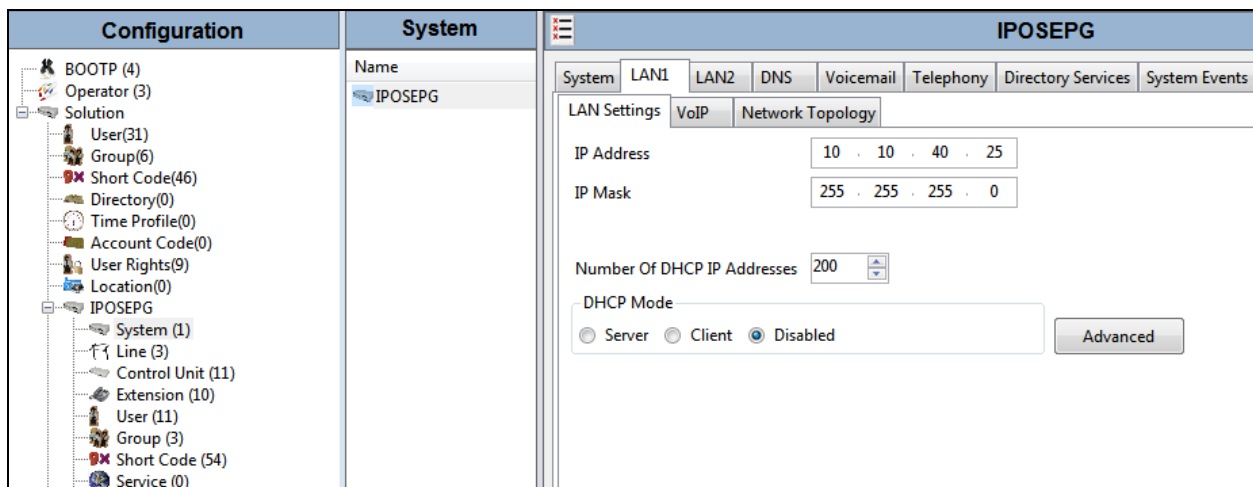


Select **Configuration** as shown below.



5.2. Display LAN Configuration

From the left window navigate to **System** and in the main window click on the **LAN1** tab and within that tab select the **LAN Settings** tab. The **IP Address** of the IP Office is shown and this will be required for the setup in **Section 6**.



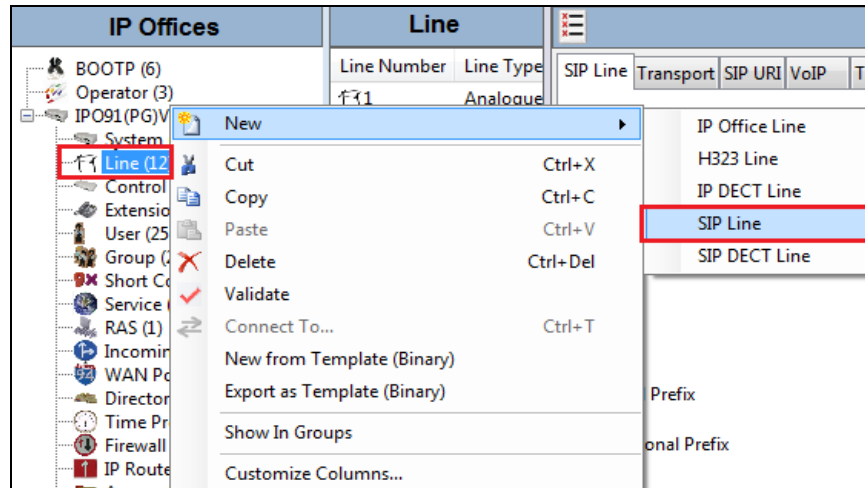
Select the **VoIP** tab and note the following below. **SIP Trunks Enable** is ticked and the **Layer 4 Protocol** settings are set.

The screenshot displays the IPOSEPG configuration window with the **VoIP** tab selected. The interface is divided into several sections:

- System Tab:** Includes LAN1, LAN2, DNS, Voicemail, Telephony, Directory Services, System Events, SMTP, SMDR, VoIP, VoIP Security, and Contact Center.
- VoIP Tab:**
 - LAN Settings:**
 - ☒ H323 Gatekeeper Enable
 - ☐ Auto-create Extn
 - ☐ Auto-create User
 - ☐ H323 Remote Extn Enable
 - H.323 Signalling over TLS: Disabled
 - Remote Call Signalling Port: 1720
 - SIP Settings:**
 - ☒ SIP Trunks Enable
 - ☒ SIP Registrar Enable
 - ☐ Auto-create Extn/User
 - ☐ SIP Remote Extn Enable
 - SIP Domain Name: devconnect.local
 - SIP Registrar FQDN: (empty)
 - Layer 4 Protocol:**
 - ☒ UDP: UDP Port 5060, Remote UDP Port 5060
 - ☒ TCP: TCP Port 5060, Remote TCP Port 5060
 - ☒ TLS: TLS Port 5061, Remote TLS Port 5061
 - Challenge Expiry Time (secs): 10
 - RTP Settings:**
 - Port Number Range:** Minimum 40750, Maximum 50750
 - Port Number Range (NAT):** Minimum 40750, Maximum 50750
 - ☒ Enable RTCP Monitoring on Port 5005
 - RTCP collector IP address for phones: 0 . 0 . 0 . 0

5.3. Create SIP Line

To create the SIP trunk from the IP Office to the MiCC server, navigate to **System** and right click on **Line** followed by **New** → **SIP Line**.



In the subsequent **SIP Line** window, enter the following in the **SIP Line** tab. **ITSP Domain Name**, this will be the telephony domain name, in this example **devconnect.local** was used.

Note: Defaults were used for the remaining fields.

A screenshot of the 'SIP Line - Line 21' configuration window. The 'SIP Line' tab is selected. The 'Line Number' is 21. The 'ITSP Domain Name' is 'devconnect.local'. The 'Local Domain Name' is empty. The 'URI Type' is 'SIP'. The 'Location' is 'Cloud'. The 'Prefix' is empty. The 'National Prefix' is '0'. The 'International Prefix' is '00'. The 'Country Code' is empty. The 'Name Priority' is 'System Default'. The 'Description' is empty. The 'In Service' checkbox is checked. The 'Check OOS' checkbox is checked. The 'Session Timers' section has 'Refresh Method' set to 'Auto' and 'Timer (seconds)' set to 'On Demand'. The 'Redirect and Transfer' section has 'Incoming Supervised REFER' set to 'Auto', 'Outgoing Supervised REFER' set to 'Auto', 'Send 302 Moved Temporarily' unchecked, and 'Outgoing Blind REFER' unchecked.

Click on the **Transport** tab enter the IP address of the MiCC Server in the **ITSP Proxy Address** field. **Layer 4 Protocol** was set to **TCP** and **Port 5060** was used.

The screenshot shows the 'SIP Line - Line 21' configuration window with the 'Transport' tab selected. The 'ITSP Proxy Address' field contains '10.10.40.128'. Under 'Network Configuration', 'Layer 4 Protocol' is set to 'TCP' and 'Send Port' is '5060'. 'Use Network Topology Info' is set to 'None' and 'Listen Port' is '5060'. 'Explicit DNS Server(s)' are set to '0 . 0 . 0 . 0' for both addresses. 'Calls Route via Registrar' is unchecked. 'Separate Registrar' is empty.

In the **SIP URI** tab click on the **Add** button.

The screenshot shows the 'SIP Line - Line 21' configuration window with the 'SIP URI' tab selected. A table with columns: URI, Groups, Local URI, Contact, Display Name, Identity, Header, Originator Number, Send Caller ID, Diversion Header, Credential, and Max Calls is visible. The 'Add...' button in the top right corner of the table area is highlighted with a red rectangle. Below the table are 'Remove' and 'Edit...' buttons.

In the subsequent window, enter the following:

- **Local URI** Enter **Auto**
- **Contact** Enter **Auto**
- **Display Name** Enter **Auto**
- **Identity** Select **None** from the dropdown menu
- **Header** Select **P Asserted ID** from the dropdown menu
- **Send Caller Id** Select **None** from the dropdown menu
- **Diversion Header** Select **None** from the dropdown menu
- **Incoming Group** Enter the SIP trunk number
- **Outgoing Group** Enter the SIP trunk number
- **Max Sessions** Enter the amount of trunks to be created (this will depend on license numbers)

Click the **OK** button.

The screenshot shows the 'SIP Line' configuration window with the 'Engineering' tab selected. The main table lists SIP lines with columns: URI, Groups, Local URI, Contact, Display Name, Identity, Header, Originator Number, Send Caller ID, Diversion Header, Credential, and Max Calls. A single line is configured with values: 1, 21, 21, Auto, Auto, Auto, None, PAI, None, None, 0: <Non..., 10. To the right of the table are buttons for 'Add...', 'Remove', and 'Edit...'. Below the table is an 'Edit URI' dialog box. This dialog contains several sections: 'Local URI' (Auto), 'Contact' (Auto), 'Display Name' (Auto), 'Identity' (None), 'Header' (P Asserted ID), 'Forwarding And Twinning' (Originator Number: empty, Send Caller Id: None), 'Diversion Header' (None), 'Registration' (0: <None>), 'Incoming Group' (21), 'Outgoing Group' (21), and 'Max Sessions' (10). At the bottom right of the dialog are 'OK' and 'Cancel' buttons.

URI	Groups	Local URI	Contact	Display Name	Identity	Header	Originator Number	Send Caller ID	Diversion Header	Credential	Max Calls
1	21	21	Auto	Auto	Auto	None	PAI	None	None	0: <Non...	10

Edit URI

Local URI: Auto

Contact: Auto

Display Name: Auto

Identity: None

Header: P Asserted ID

Forwarding And Twinning

Originator Number:

Send Caller Id: None

Diversion Header: None

Registration: 0: <None>

Incoming Group: 21

Outgoing Group: 21

Max Sessions: 10

OK Cancel

Click on the **VoIP** tab System Default was left as the Codec Selection with **G.711 ALAW 64K** being used as the default Codec. **DTMF Support** was left as **RFC2833/RFC4733** for compliance testing but this may differ on a customer site. **Re-invite Supported** and **PRACK/100rel Supported** were also ticked. Click the **OK** button once everything is set correctly (not shown).

SIP Line - Line 21

Tabs: SIP Line | Transport | SIP URI | **VoIP** | SIP Credentials | SIP Advanced | Engineering

Codec Selection: System Default

Unused:

- G.711 ULAW 64K
- G.722 64K
- G.729(a) 8K CS-ACELP

Selected:

- G.711 ALAW 64K

Local Hold Music: ☐

Re-invite Supported: ☒

Codec Lockdown: ☐

Allow Direct Media Path: ☐

Force direct media with phones: ☐

PRACK/100rel Supported: ☒

Fax Transport Support: None

DTMF Support: RFC2833/RFC4733

Media Security: Disabled

For compliance testing the values under the **SIP Advanced** tab were left as default as shown below except for **Allow Empty INVITE** this must be ticked in order for the solution to work.

SIP Line - Line 21

Tabs: SIP Line | Transport | SIP URI | VoIP | **SIP Advanced** | Engineering

Addressing

Association Method: By Source IP address

Call Routing Method: Request URI

Suppress DNS SRV Lookups: ☐

Identity

Use "phone-context": ☐

Add user=phone: ☐

Use + for International: ☐

Use PAI for Privacy: ☐

Use Domain for PAI: ☐

Swap From and PAI/Diversion: ☐

Caller ID from From header: ☐

Send From In Clear: ☐

Cache Auth Credentials: ☒

User-Agent and Server Headers:

Send Location Info: Never

Add UUI header: ☐

Add UUI header to redirected calls: ☐

Media

Allow Empty INVITE: ☒

Send Empty re-INVITE: ☐

Allow To Tag Change: ☐

P-Early-Media Support: None

Send SilenceSupp=Off: ☐

Force Early Direct Media: ☐

Media Connection Preservation: Disabled

Indicate HOLD: ☐

Call Control

Call Initiation Timeout (s): 4

Call Queuing Timeout (m): 5

Service Busy Response: 486 - Busy Here

on No User Responding Send: 408-Request Timeout

Action on CAC Location Limit: Allow Voicemail

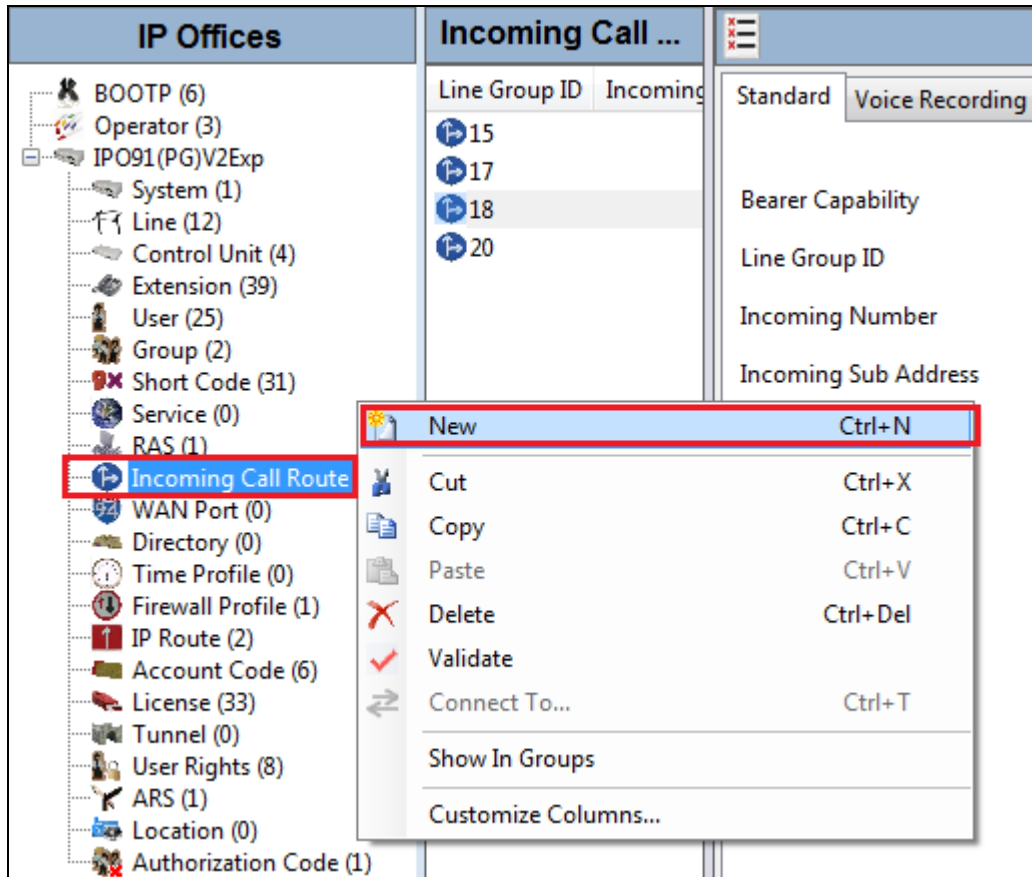
Suppress Q.850 Reason Header: ☐

Emulate NOTIFY for REFER: ☐

No REFER if using Diversion: ☐

5.4. Configure Incoming Call Route

To configure the Incoming Call Route, navigate to **System** and right click on **Incoming Call Route** followed by **New**.



In the subsequent window, enter the following in the **Standard** tab. Enter the SIP LINE that was created in **Section 5.3** for the **Line Group ID**. Defaults were used for the remaining fields.

The screenshot shows a configuration window titled '21' with three tabs: 'Standard', 'Voice Recording', and 'Destinations'. The 'Standard' tab is active. It contains the following fields and values:

Field	Value
Bearer Capability	Any Voice
Line Group ID	21
Incoming Number	
Incoming Sub Address	
Incoming CLI	
Locale	
Priority	1 - Low
Tag	
Hold Music Source	System Source
Ring Tone Override	None

In the **Destinations** tab, enter a . (full stop/period) in the **Destination** field. Click on the **OK** button.

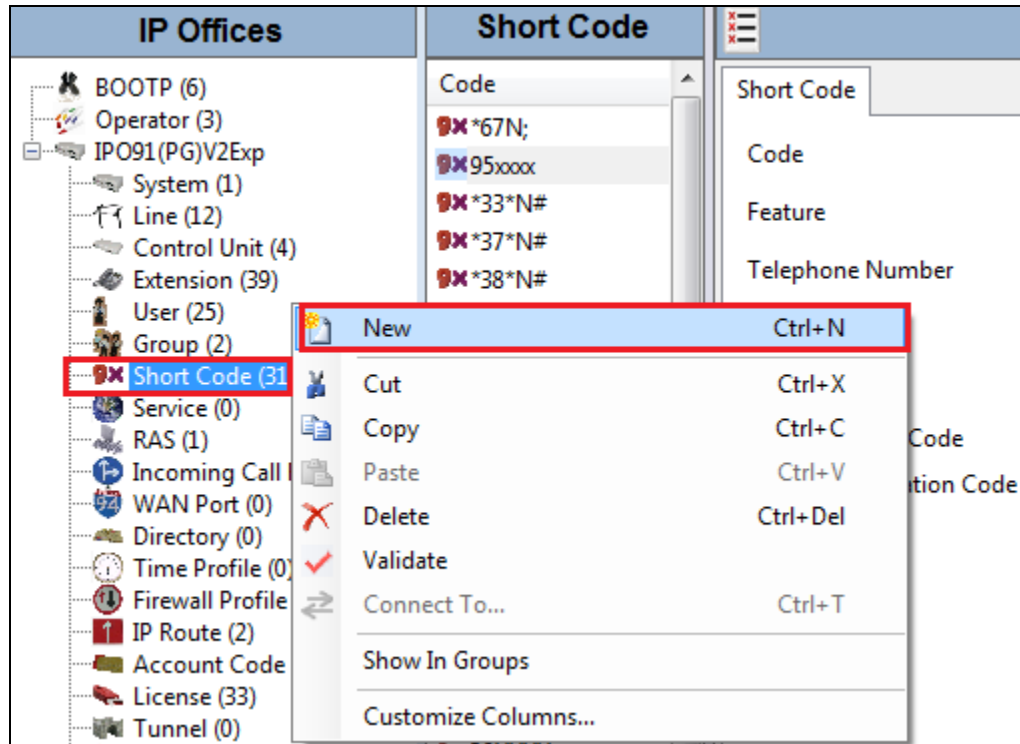
The screenshot shows the same configuration window with the 'Destinations' tab active. It contains a table with the following data:

TimeProfile	Destination	Fallback Extension
Default Value	.	

At the bottom right, there are three buttons: 'OK', 'Cancel', and 'Help'. The 'OK' button is highlighted with a red box.

5.5. Create Short Code (Route Calls)

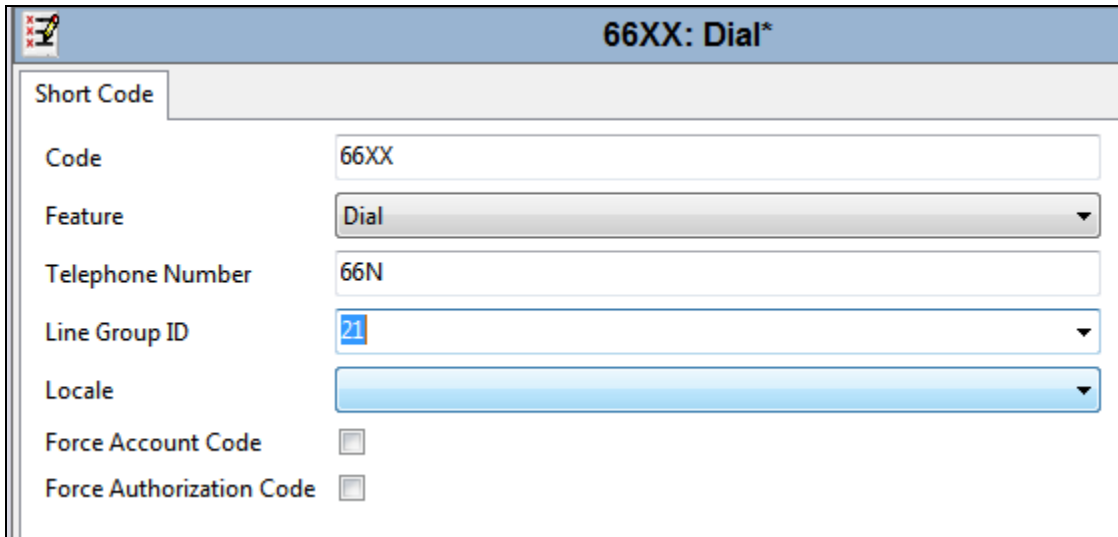
A Short Code needs to be configured on the IP Office to route calls to Flexi server. Right click on **Short Code**, and select **New**.



In the subsequent window, enter the following:

- **Code** Enter the number range that will be routed to MiCC server (during compliance testing, all numbers beginning with 66 that were 4 digits in length were sent to MiCC server, therefore **66XX** was entered).
- **Feature** Select **Dial** from the dropdown menu.
- **Telephone Number** **66N** which is 6 plus the numbers entered after 66.
- **Group Line ID** Enter the SIP Line created in **Section 5.3**.

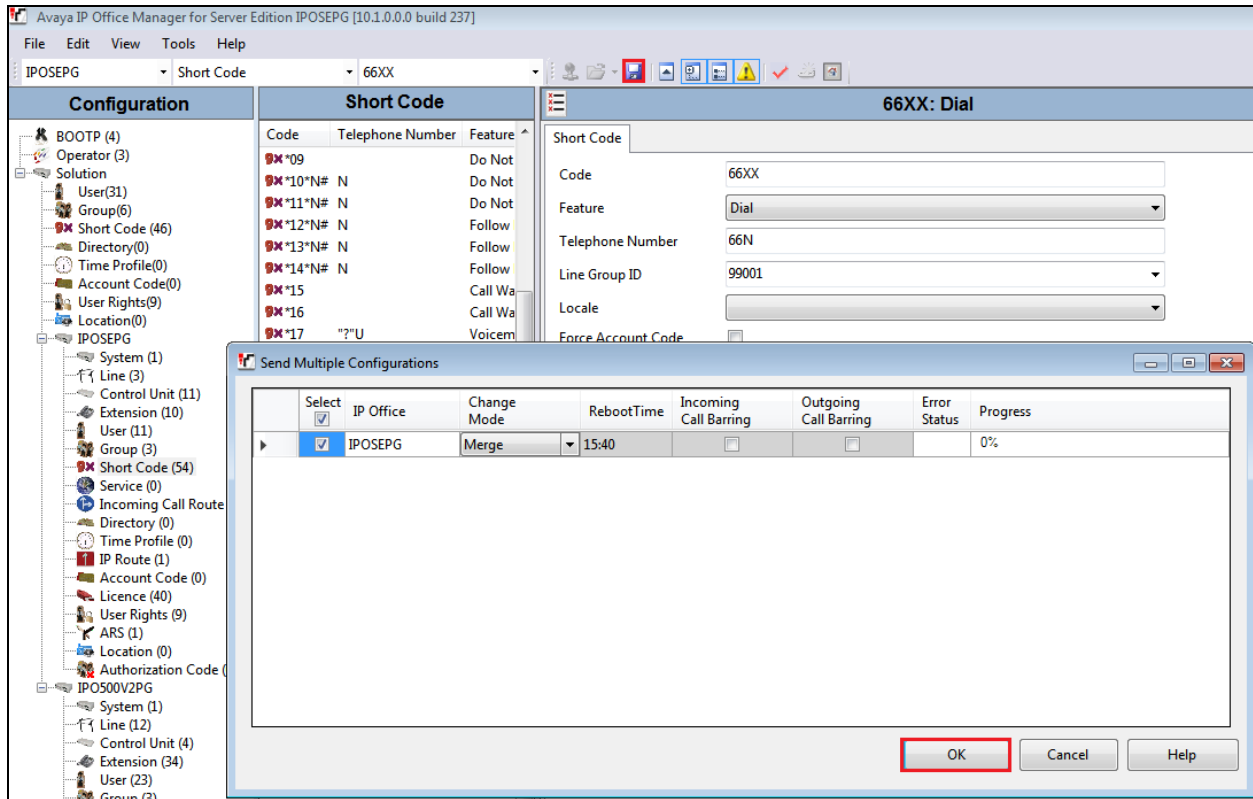
Click the **OK** button (not shown).



66XX: Dial*	
Short Code	
Code	66XX
Feature	Dial
Telephone Number	66N
Line Group ID	21
Locale	
Force Account Code	<input type="checkbox"/>
Force Authorization Code	<input type="checkbox"/>

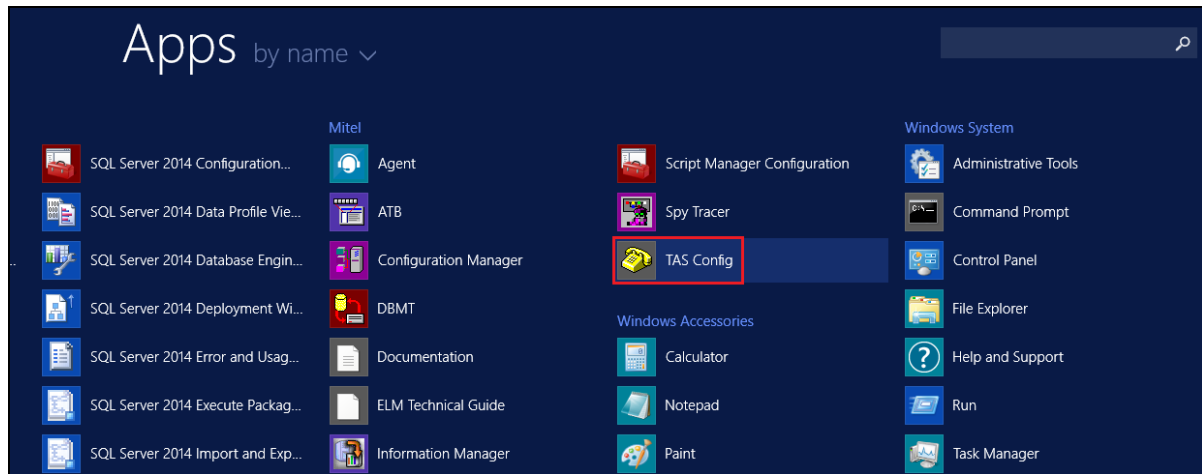
5.6. Save Configuration

Once all the configurations have been made it must be saved to IP Office. Click on the **Save** icon at the top of the screen and the following window appears, click on **OK** to commit the changes to memory.

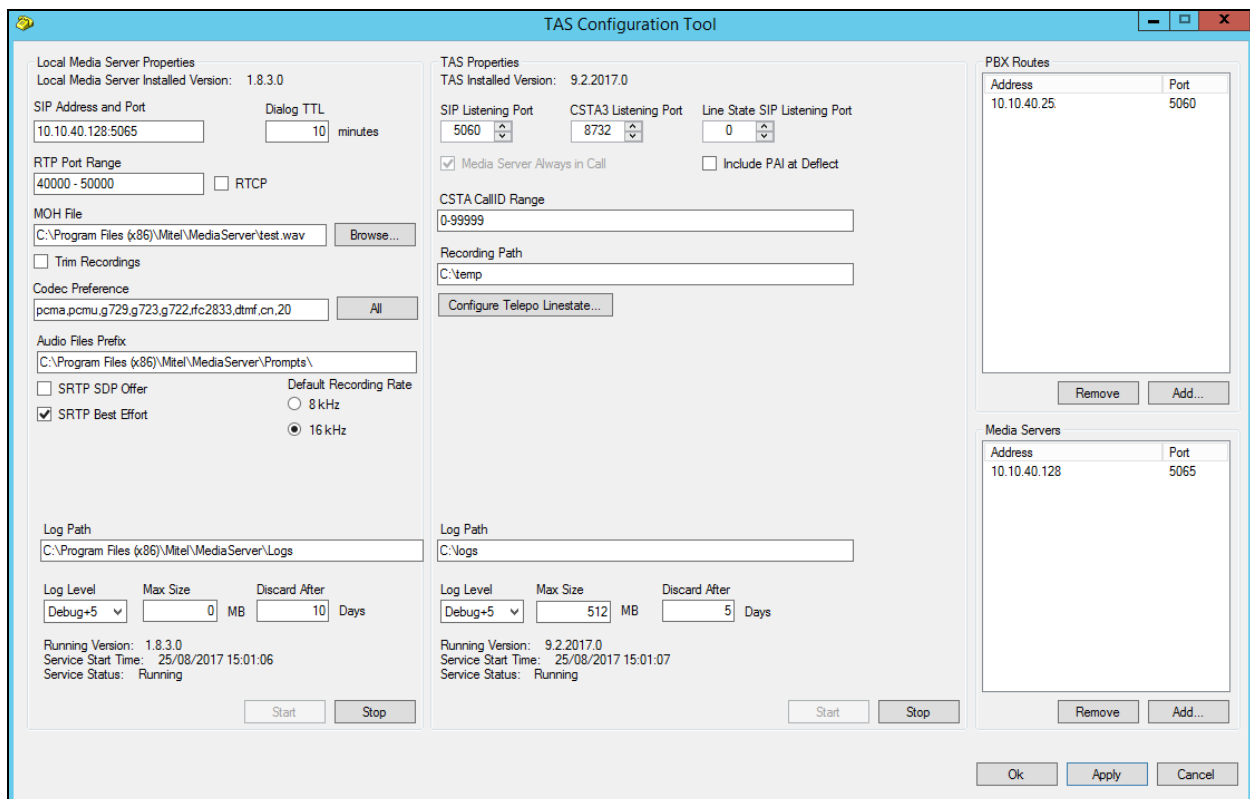


6. Configure Mitel MiContact Center Enterprise Server

Although a Mitel engineer will setup the solution the following section show information on the connection to Session Manager that was used for compliance testing, it may prove useful. The connection to the Avaya solution is configured in **TAS Config** as shown below.



The connection to IP Office is defined under **PBX Routes** along with the port number. The Codecs used can be seen under **Codec Preference** and the other settings are used for the connection to the Mitel solution.



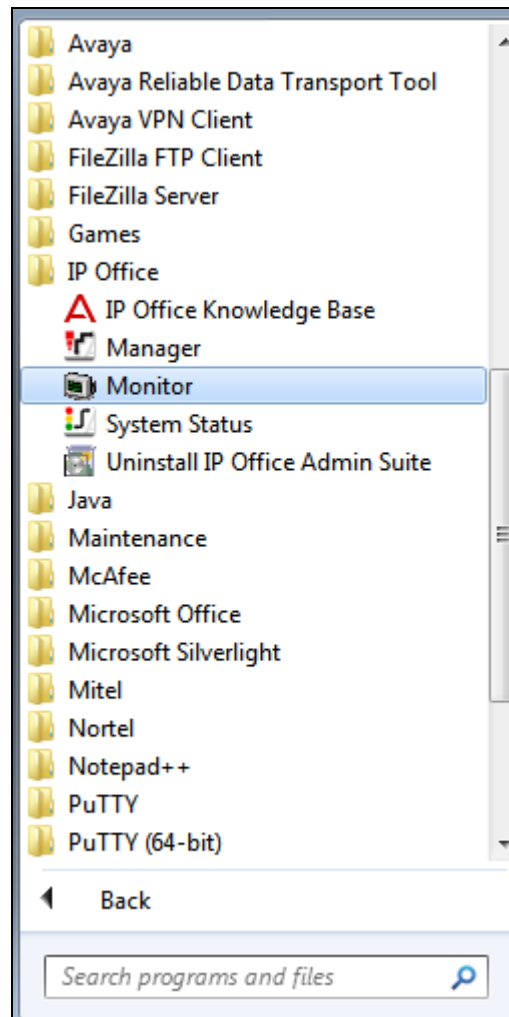
7. Verification Steps

This section provides the tests that can be performed to verify correct configuration of Avaya and Mitel solution.

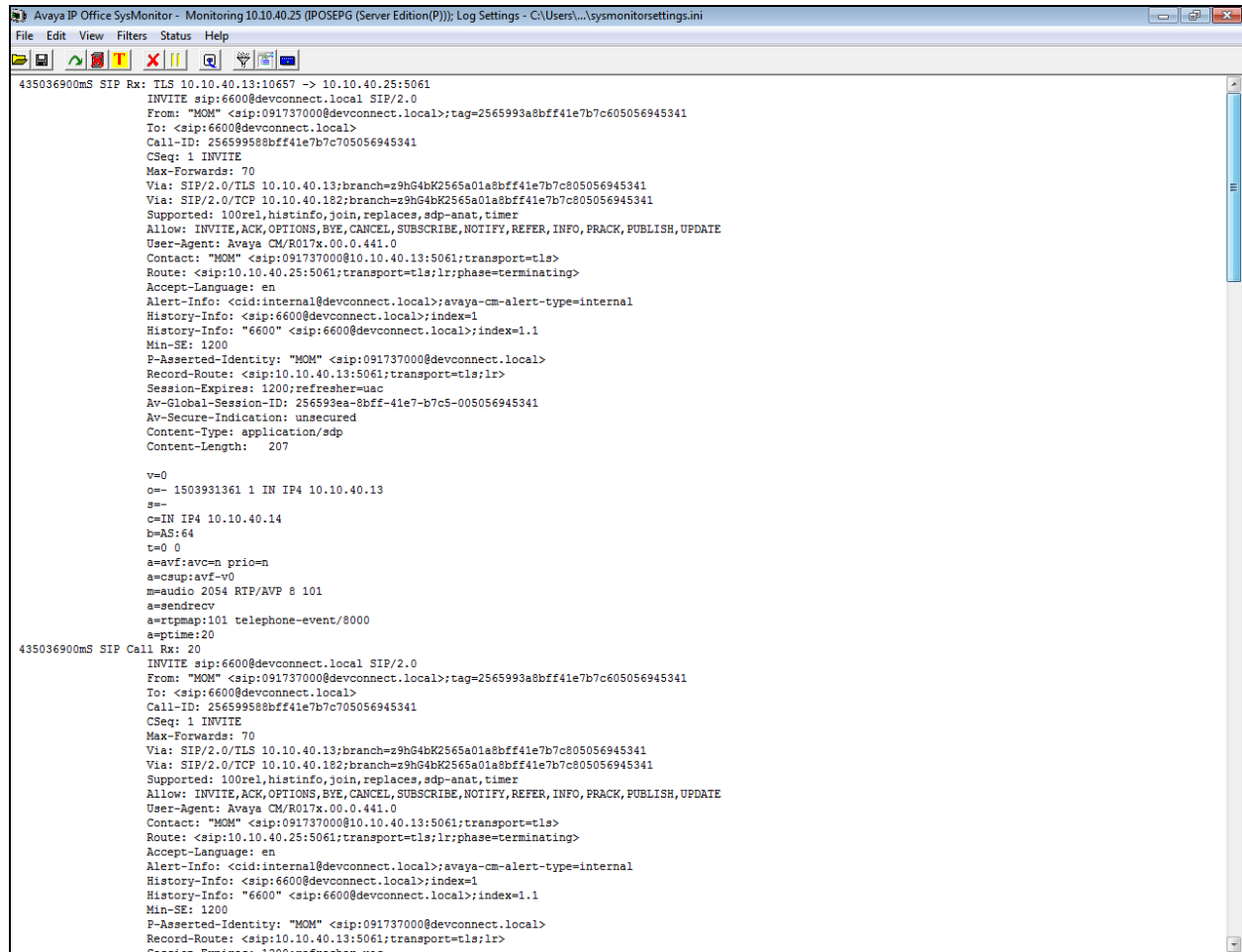
- Verify the SIP connection on IP Office.
- Verify that a call can be successfully made to the MiCC agent and answered on the MiCC Agent application.

7.1. Verify the SIP connection on IP Office

The IP Office Monitor can be used in order to troubleshoot any SIP connection errors. The **Monitor** is selected as shown below.



The Monitor is setup to shown SIP TX/RX messages and can be used to diagnose any SIP errors and find issues with the SIP connection.



The screenshot displays the Avaya IP Office SysMonitor application window. The title bar reads "Avaya IP Office SysMonitor - Monitoring 10.10.40.25 (IPOSEG (Server Edition(P))) Log Settings - C:\Users\...\sysmonitorsettings.ini". The menu bar includes File, Edit, View, Filters, Status, and Help. The main window shows a list of SIP messages. The first message is a SIP Rx (receive) message from 10.10.40.13 to 10.10.40.25 at 435036900ms. The second message is a SIP Call Rx (receive) message from 10.10.40.13 to 10.10.40.14 at 435036900ms. Both messages are INVITE requests from "MOM" to "6600" with various SIP headers and body parameters.

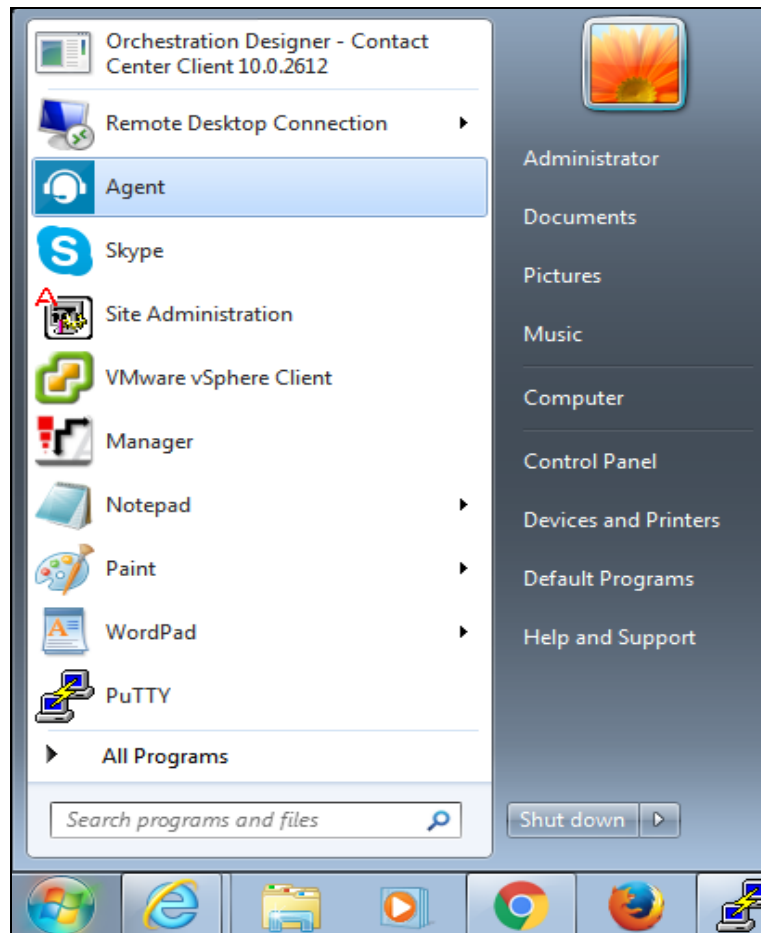
```
435036900ms SIP Rx: TLS 10.10.40.13:10657 -> 10.10.40.25:5061
INVITE sip:6600@devconnect.local SIP/2.0
From: "MOM" <sip:091737000@devconnect.local>;tag=2565993a8bffa1e7b7c805056945341
To: <sip:6600@devconnect.local>
Call-ID: 256599588bffa1e7b7c705056945341
CSeq: 1 INVITE
Max-Forwards: 70
Via: SIP/2.0/TLS 10.10.40.13:branch=z9hG4bK2565a01a8bffa1e7b7c805056945341
Via: SIP/2.0/TCP 10.10.40.182:branch=z9hG4bK2565a01a8bffa1e7b7c805056945341
Supported: 100rel,histinfo,join,replaces,sdp-anat,timer
Allow: INVITE,ACK,OPTIONS,BYE,CANCEL,SUBSCRIBE,NOTIFY,REFER,INFO,PRACK,PUBLISH,UPDATE
User-Agent: Avaya CM/R017x.00.0.441.0
Contact: "MOM" <sip:091737000@10.10.40.13:5061;transport=tls>
Route: <sip:10.10.40.25:5061;transport=tls;lr;phase=terminating>
Accept-Language: en
Alert-Info: <cid:internal@devconnect.local>;avaya-cm-alert-type=internal
History-Info: <sip:6600@devconnect.local>;index=1
History-Info: "6600" <sip:6600@devconnect.local>;index=1.1
Min-SE: 1200
P-Asserted-Identity: "MOM" <sip:091737000@devconnect.local>
Record-Route: <sip:10.10.40.13:5061;transport=tls;lr>
Session-Expires: 1200;refresher=uac
Av-Global-Session-ID: 2565993a8bffa1e7b7c5-005056945341
Av-Secure-Indication: unsecured
Content-Type: application/sdp
Content-Length: 207

v=0
o=- 15039931361 1 IN IP4 10.10.40.13
s=-
c=IN IP4 10.10.40.14
b=AS:64
t=0 0
a=avf:avc=n prio=n
a=cup:avf-v0
m=audio 2054 RTP/AVP 8 101
a=sendrecv
a=rtpmap:101 telephone-event/8000
a=ptime:20

435036900ms SIP Call Rx: 20
INVITE sip:6600@devconnect.local SIP/2.0
From: "MOM" <sip:091737000@devconnect.local>;tag=2565993a8bffa1e7b7c805056945341
To: <sip:6600@devconnect.local>
Call-ID: 256599588bffa1e7b7c705056945341
CSeq: 1 INVITE
Max-Forwards: 70
Via: SIP/2.0/TLS 10.10.40.13:branch=z9hG4bK2565a01a8bffa1e7b7c805056945341
Via: SIP/2.0/TCP 10.10.40.182:branch=z9hG4bK2565a01a8bffa1e7b7c805056945341
Supported: 100rel,histinfo,join,replaces,sdp-anat,timer
Allow: INVITE,ACK,OPTIONS,BYE,CANCEL,SUBSCRIBE,NOTIFY,REFER,INFO,PRACK,PUBLISH,UPDATE
User-Agent: Avaya CM/R017x.00.0.441.0
Contact: "MOM" <sip:091737000@10.10.40.13:5061;transport=tls>
Route: <sip:10.10.40.25:5061;transport=tls;lr;phase=terminating>
Accept-Language: en
Alert-Info: <cid:internal@devconnect.local>;avaya-cm-alert-type=internal
History-Info: <sip:6600@devconnect.local>;index=1
History-Info: "6600" <sip:6600@devconnect.local>;index=1.1
Min-SE: 1200
P-Asserted-Identity: "MOM" <sip:091737000@devconnect.local>
Record-Route: <sip:10.10.40.13:5061;transport=tls;lr>
Session-Expires: 1200;refresher=uac
```

7.2. Verify the MiCC Agent Application

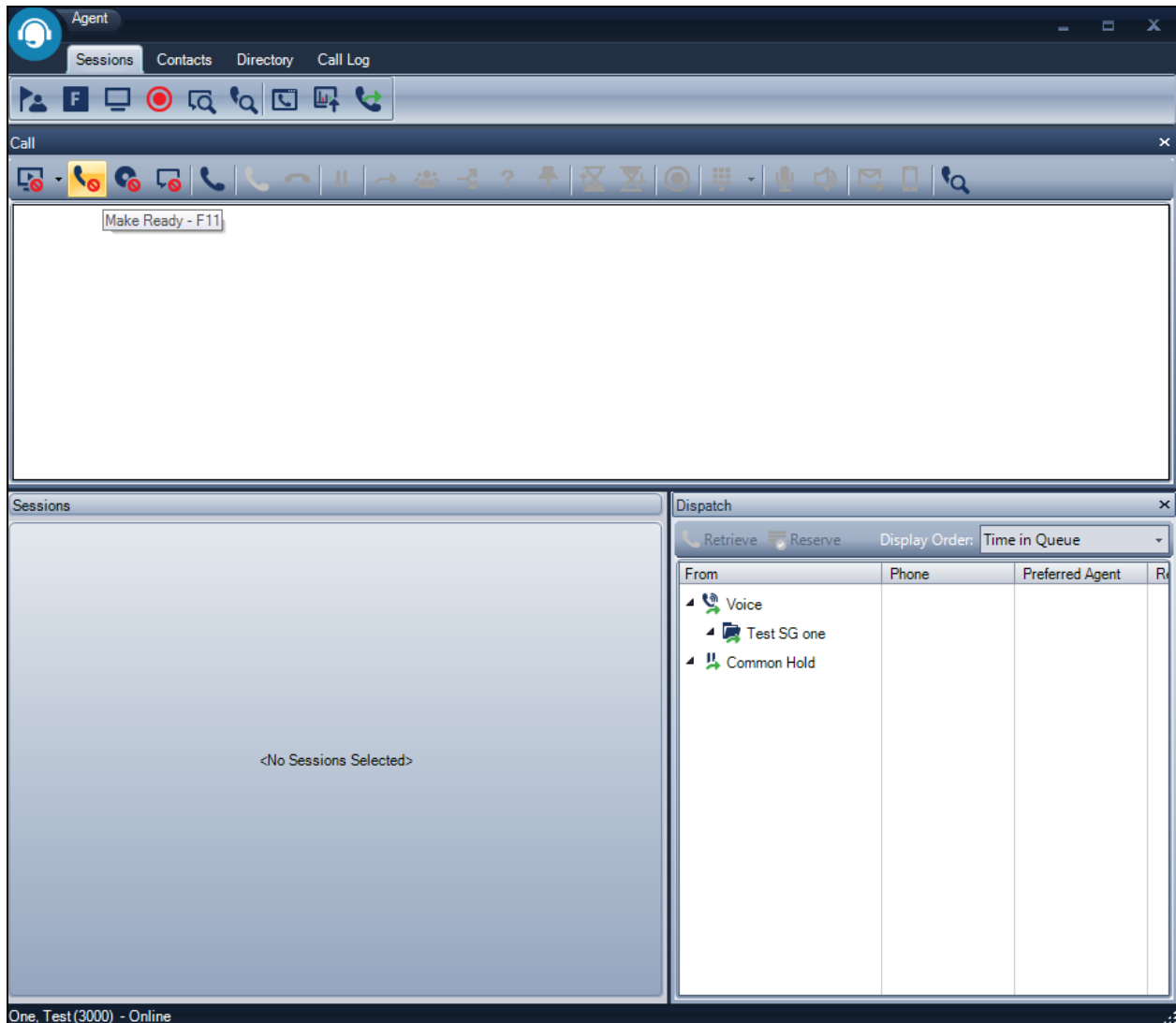
Open the MiCC Agent application as shown below by selecting **Agent** from the programs.



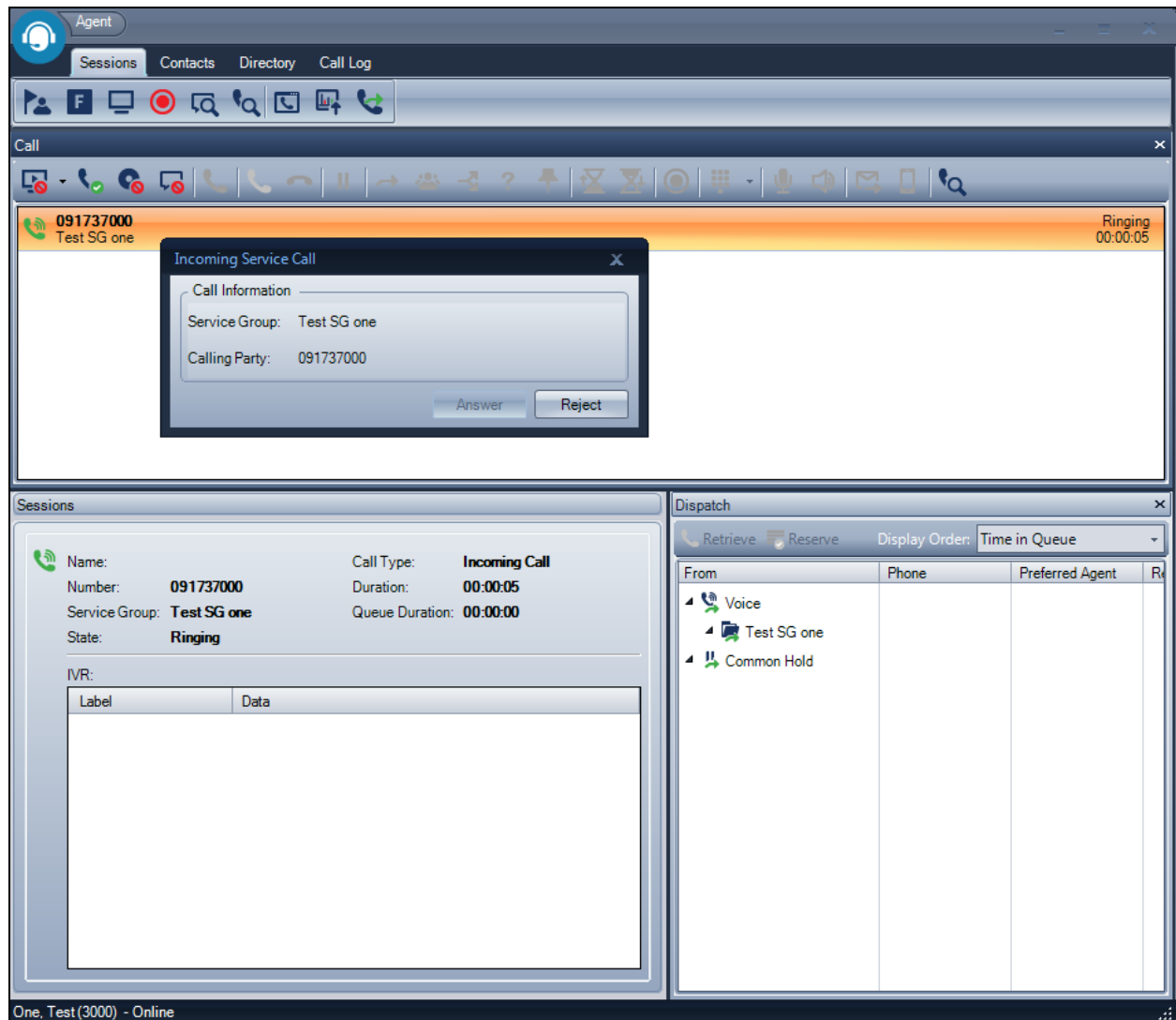
Enter the agent's details and note that the **Extension Type** is set to **Desktop Phone** which is selected in order to use the IP Office phone sets. The IP Office extension is entered; this can be any extension on the CIP Office and does not need to be configured in any specific way on either the IP Office or the MiCC. Press **OK** to log in.

A 'User Logon' dialog box with a dark blue title bar. It contains four input fields: 'Logon ID' with the value '1', 'Password' with a masked character '•', 'Extension Number' with the value '3000', and 'Extension Type' with a dropdown menu showing 'Desktop Phone'. At the bottom are 'OK' and 'Cancel' buttons.

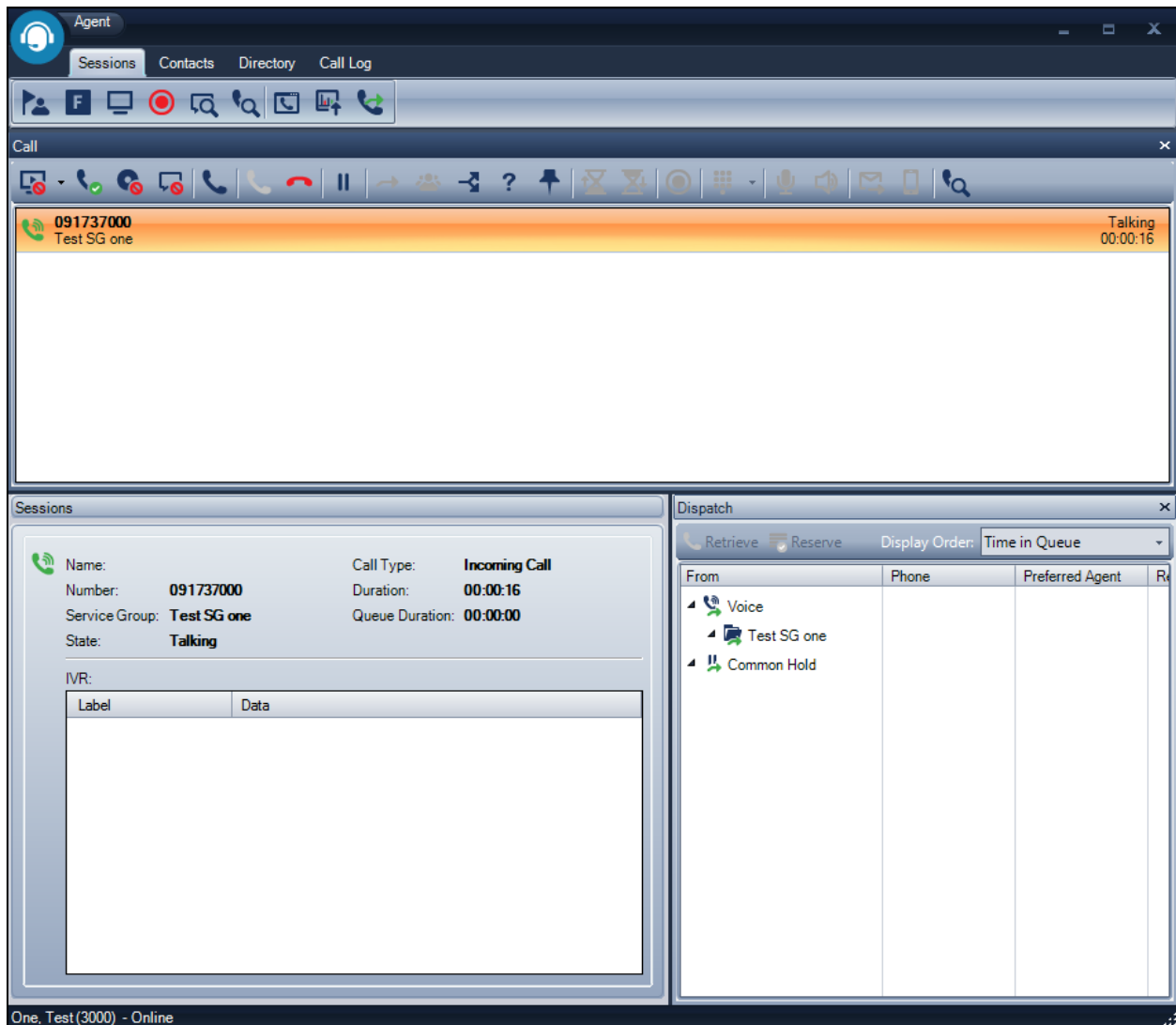
The following agents screen is displayed showing the controls to make the agent available.



A call is made to the number as defined in **Section 5.5**. The call must be answered using the IP Office handset.



Once the call is answered the agent can place the call on hold and transfer and conference the caller using the buttons on the agent desktop.



8. Conclusion

The interoperability of MiContact Center Enterprise from Mitel Networks Corporation with Avaya IP Office R10.1 was successful for this specific setup in order to place calls to and from Mitel MiCC Agents. All test cases passed successfully.

9. Additional References

These documents form part of the Avaya official technical reference documentation suite. Further information can be obtained from <http://support.avaya.com> or from your Avaya representative.

- [1] Avaya IP Office R10.1 Manager 10.1, Document Number 15-601011
- [2] Avaya IP Office R10.1 Doc library

Product Documentation for MiCC can be obtained from Mitel at: www.Mitel.com/support

©2017 Avaya Inc. All Rights Reserved.

Avaya and the Avaya Logo are trademarks of Avaya Inc. All trademarks identified by ® and ™ are registered trademarks or trademarks, respectively, of Avaya Inc. All other trademarks are the property of their respective owners. The information provided in these Application Notes is subject to change without notice. The configurations, technical data, and recommendations provided in these Application Notes are believed to be accurate and dependable, but are presented without express or implied warranty. Users are responsible for their application of any products specified in these Application Notes.

Please e-mail any questions or comments pertaining to these Application Notes along with the full title name and filename, located in the lower right corner, directly to the Avaya DevConnect Program at devconnect@avaya.com.