Avaya Solution & Interoperability Test Lab

Application Notes for Configuring SIP Trunking between the Comdasys Mobile Convergence Solution and an Avaya IP Office Telephony Solution – Issue 1.0

Abstract

These Application Notes describe the steps to configure trunking using the Session Initiation Protocol (SIP) between the Comdasys Mobile Convergence Solution and Avaya IP Office. Comdasys Mobile Convergence Solution allows GSM telephones with a wireless LAN interface to be assigned a telephone extension on Avaya IP Office.

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

The Comdasys Mobile Convergence Solution together with Avaya IP Office allows “dual mode” mobile endpoints to act as local IP Office extensions. In addition to a GSM interface, such endpoints have a wireless LAN interface and a SIP client. When used within the coverage range of the local wireless LAN, incoming and outgoing calls for these endpoints are made via the mobile endpoint wireless LAN interface. When outside this coverage area, incoming and outgoing calls are made via the GSM network. When mobile endpoints enter or exit the wireless LAN coverage area, calls are “handed over” between the GSM and wireless LAN networks. The Mobile Convergence Client needs to be installed on the mobile phone. Placing phone calls and feature invocation are executed transparently for the end-user either in the WIFI or GSM mode. Avaya Voicemail Pro was also included in the test configuration.

SIP is a standards-based communications approach designed to provide a common framework to support multimedia communication. RFC 3261 **[5]** is the primary specification governing this protocol. In the configuration described in these Application Notes, SIP is used as the signaling protocol between the Avaya components and the Comdasys Mobile Convergence Solution. SIP manages the establishment and termination of connections and the transfer of related information such as the desired codec, calling party identity, etc.

1.1. Interoperability Compliance Testing

The following tests were performed as part of the compliance testing. When appropriate, the tests were covered for calls established via the wireless LAN (WLAN) interface and the GSM interface of the client endpoints involved.

- Outgoing/incoming local/PSTN call
- Outgoing/incoming local/PSTN call rejection
- Outgoing/incoming local/PSTN call cancellation
- Call forwarding
- Supervised/blind transfer
- Consultation
- Hold/retrieve
- Manual handover from WLAN
- Automatic handover from WLAN/GSM
- Interruption to Comdasys server LAN interface
- Interruption to Comdasys server power

1.2. Support

Support is available via the Comdasys distributor network.

2. Reference Configuration

The following diagram illustrates the configuration which was used for testing.

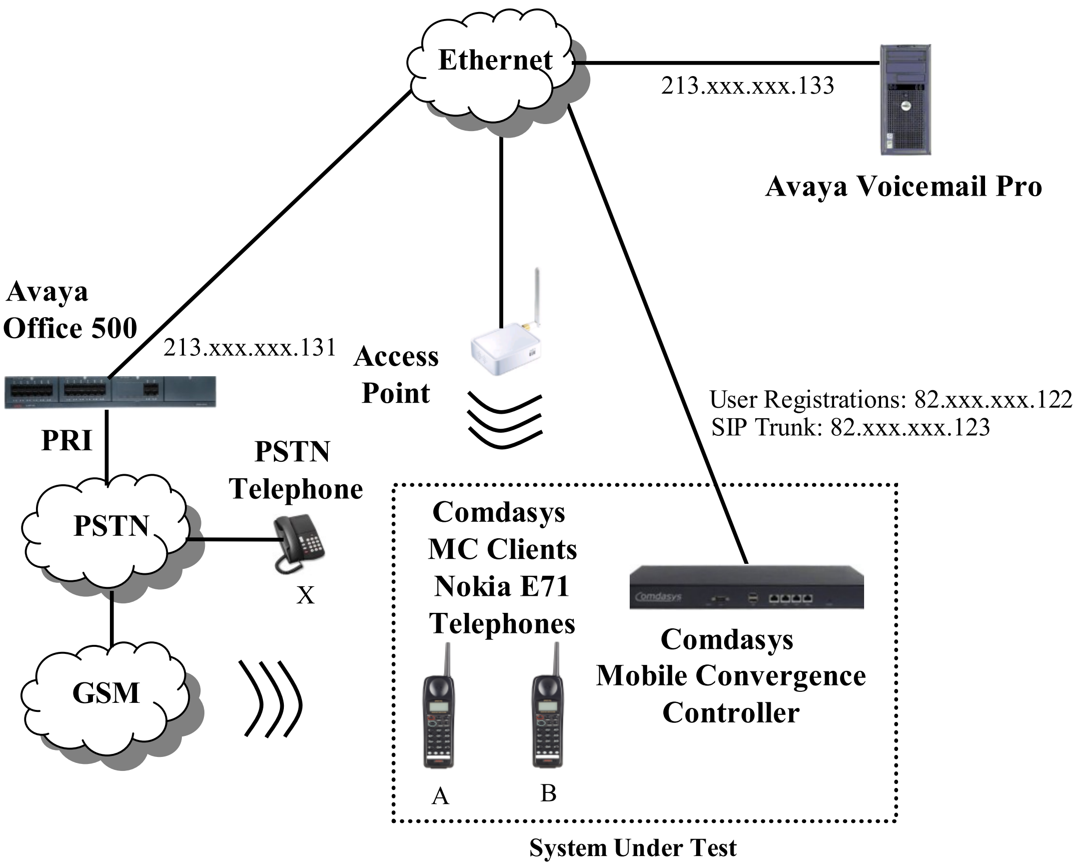

Figure 1: Reference Configuration

The Comdasys Mobile Convergence Controller is connected to Avaya IP Office and acts as a series of individually registered SIP subscribers. An additional SIP trunk between Avaya IP Office and the MC Controller is used for routing GSM calls. Various call flow scenarios are illustrated in the following diagrams.

Call from Client to PSTN via LAN:

1. The wireless endpoint sends an INVITE to the Mobile Convergence Controller.
2. The Mobile Convergence Controller uses its IP Office SIP registration account to send an INVITE to IP Office.
3. IP Office makes a call to the called endpoint using its PSTN interface

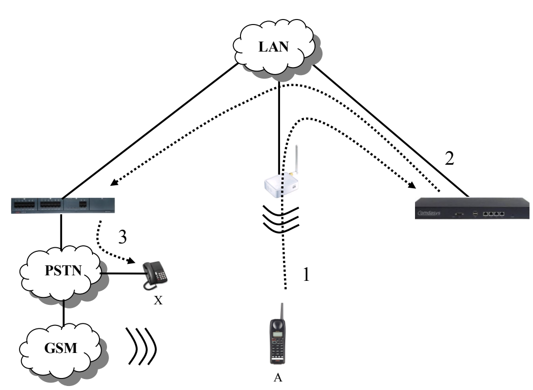

Figure 2: Call from Client to PSTN via LAN

Call from Client to PSTN via GSM:

1. The GSM endpoint calls the "Call Through" number via the PSTN.
2. IP Office sends an INVITE to the Mobile Convergence Controller trunk interface ("Call Through").
3. The Mobile Convergence Controller sends an INVITE to IP Office via its SIP registration account.
4. IP Office makes a call to the called endpoint via its PSTN interface.

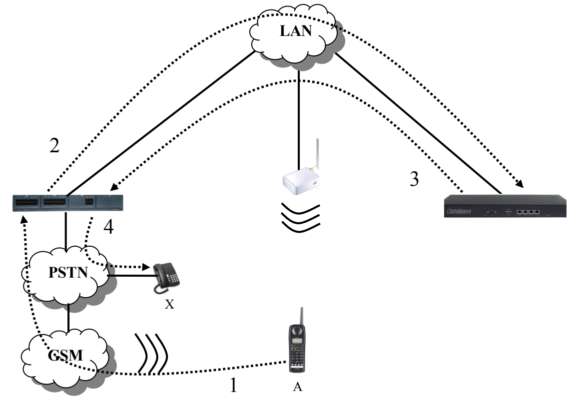

Figure 3: Call from Client to PSTN via GSM

Call from PSTN to Client via LAN:

1. The calling endpoint calls the PSTN number of the SIP client associated with the called endpoint.
2. IP Office sends an INVITE to the Mobile Convergence Controller via its SIP account.
3. The Mobile Convergence Controller sends an INVITE to the wireless endpoint.

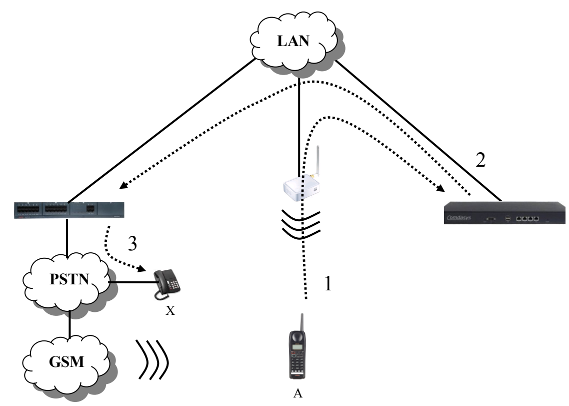

Figure 4: Call from PSTN to Client via LAN

Call from PSTN to Client via GSM:

1. The calling endpoint calls the PSTN number of the SIP client associated with the called endpoint.
2. IP Office sends an INVITE to the Mobile Convergence Controller SIP registration account.
3. The MC Controller sends an INVITE to IP Office via its SIP trunk. The GSM number configured in the MC Controller is used to call the mobile endpoint ("static roaming").
4. IP Office makes a call to the called endpoint using its PSTN interface

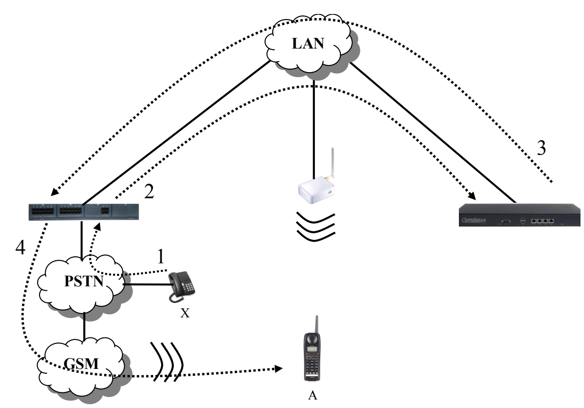

Figure 5: Call from PSTN to Client via GSM

The telephone numbers used for testing are shown in the following table.

Endpoint	Ext	PSTN Number	Station Type
A	2363	069 907 xxxxx 2363	SIP Endpoint
B	2364	069 907 xxxxx 2364	SIP Endpoint
Call-Through	2365	069 907 xxxxx 2365	Special FMC Number
Callback	2366	069 907 xxxxx 2366	Special FMC Number
SIM-Switch	2367	069 907 xxxxx 2367	Special FMC Number
X		069 xxxxx 6174	PSTN

Table 1: Extensions Used for Testing

Note that the extensions shown for the Special FMC Number entries do not have corresponding IP Office extensions, but are used for command codes which are sent to the MC Client, which are used for the purposes shown in the following table.

FMC Number	Usage	Description
2365	Call-Through	The call-through number is used by the MC-Client in GSM mode to dial the MC Controller. All users share the same Call-Through number. The Call-Through is mandatory for initiating calls from the Client whenever it is out of WIFI range.
2366	Callback	The callback is used to reduce mobile costs when roaming outside the home GSM network. The MC Client calls the callback number in order to trigger a callback. The callback is an optional feature.
2367	SIM Switch	The SIM Switch is used for switching between multiple GSM SIM cards. A user can activate his SIM card by calling this number. This is done by the MC Client after it has detected a new SIM card inserted into the mobile phone.

Table 2: FMC Number Usage

3. Validated Equipment and Software

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

Equipment	Software
Avaya IP Office 500	5.0 (18)
Avaya Voicemail Pro	4.2 (27)
Nokia E71	----------
Comdasys MC Client	Nokia MC Client 2.1
Comdasys Mobile Convergence Controller	9530.6

Table 3: Equipment and Software Validated

4. Avaya IP Office Configuration

All configuration steps for Avaya IP Office were performed using the IP Office Manager application. This application presents the administrator with a hierarchy of icons for configuring various components, as shown below.

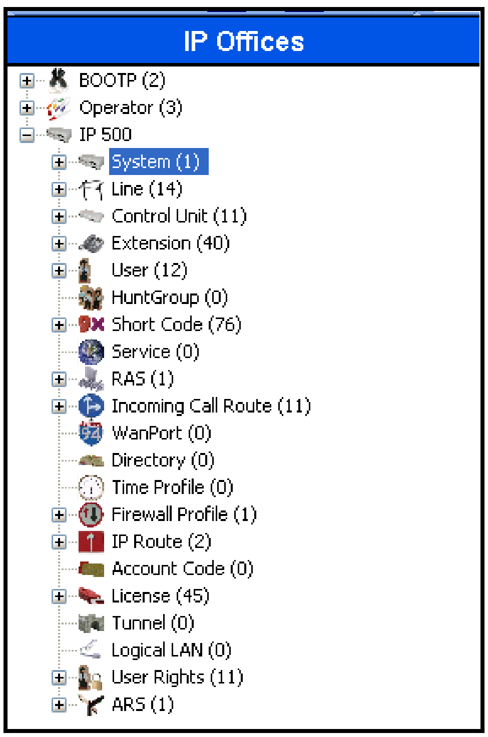

Figure 6: IP Office Manager Top Level Presentation

4.1. Licenses

The licenses which were used for testing Avaya IP Office with Comdasys Mobile Convergence Solution are shown below. Note that although Voicemail Pro was used for testing, another type of voicemail that is compatible with IP Office can also be used.

Figure 7: IP Office Licenses: SIP Trunk Channels

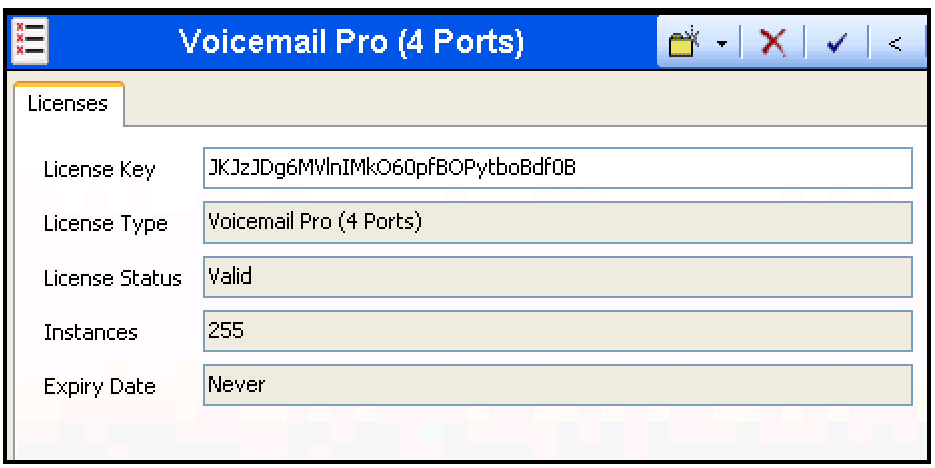

Figure 8: IP Office Licenses: Voicemail Pro

4.2. System

Select the "System" icon shown in **Figure 6** and enter the parameters shown in the following table.

Tab	Parameter	Usage
LAN1 LAN Settings	IP Address	Enter the IP address assigned to IP Office.
	IP Mask	Enter the network mask assigned to IP Office.
LAN1 VoIP	H323 Gatekeeper Enable	Check this box.
	SIP Trunks Enable	Check this box.
	SIP Registrar Enable	Check this box.
LAN1 SIP Registrar	UDP Port	Enter 5060. This value must also be entered as the "Foreign Port" value for the "IPO" entry in **Figure 11**.
Voicemail	Voicemail Type	Select the appropriate voicemail interface from the drop-down menu.
Telephony	Dial Delay Time	Enter the inter-digit dial delay time. A value of "5" seconds was used for the test.
	Dial Delay Count	Enter "0".
	Automatic Codec Preference	Select "G.711 ALAW 64K".

Table 4: IP Office System Parameters

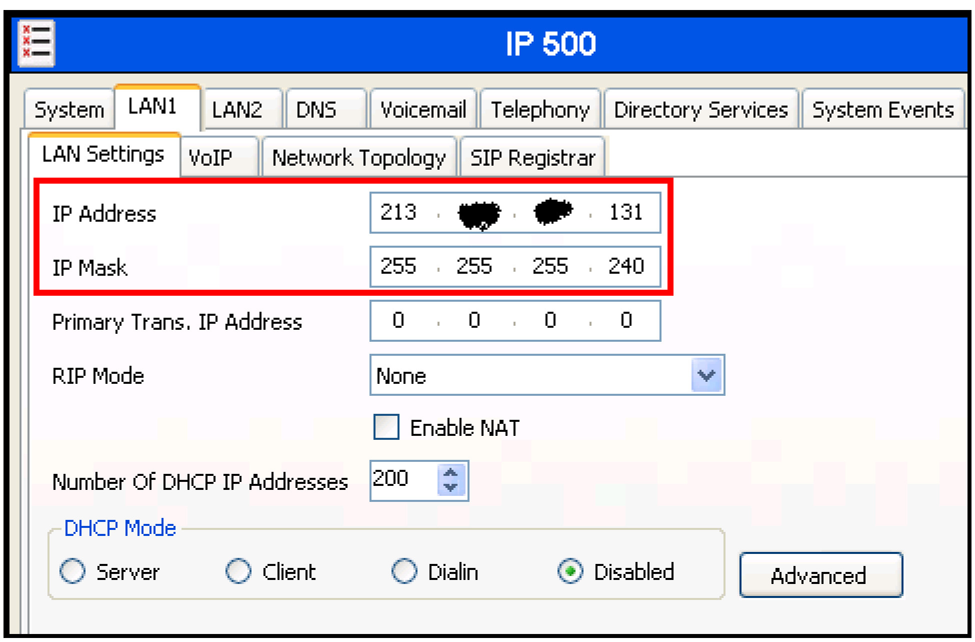

Figure 9: IP Office System: LAN1 Settings Tab

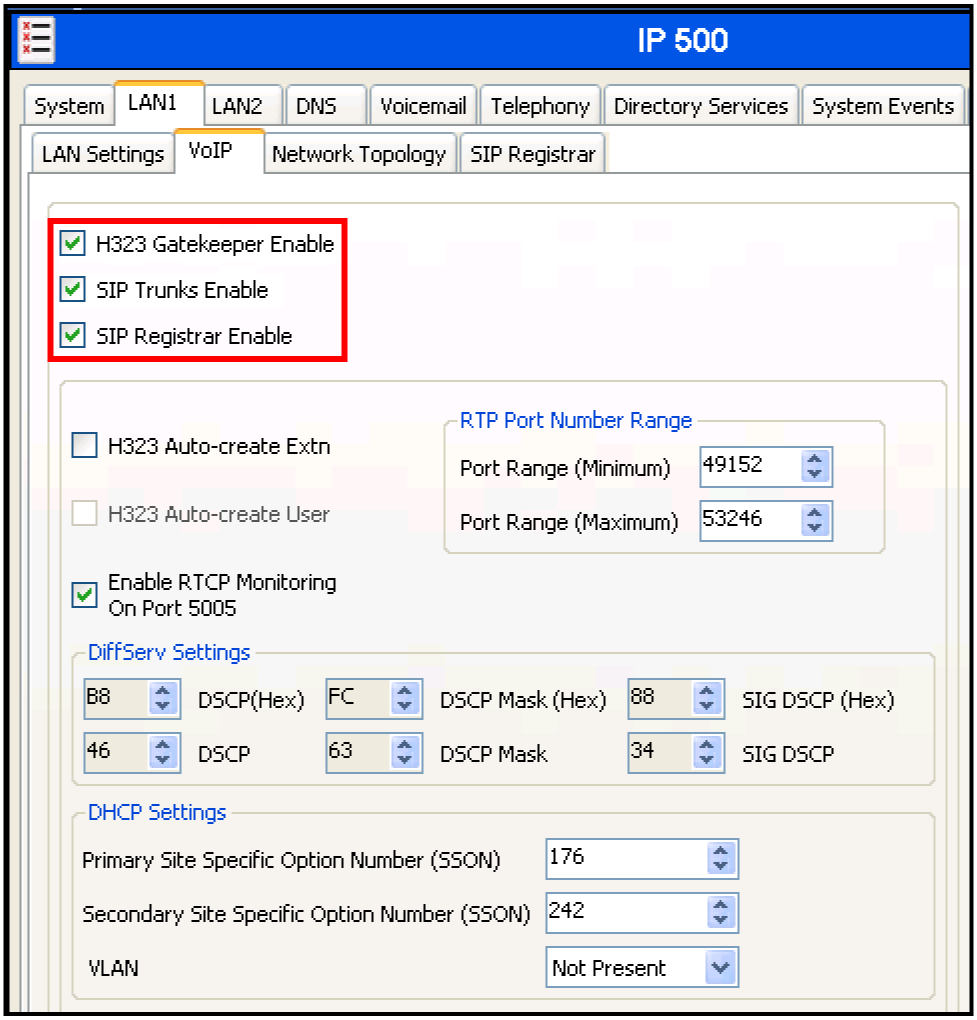

Figure 10: IP Office System: LAN1 VoIP Tab

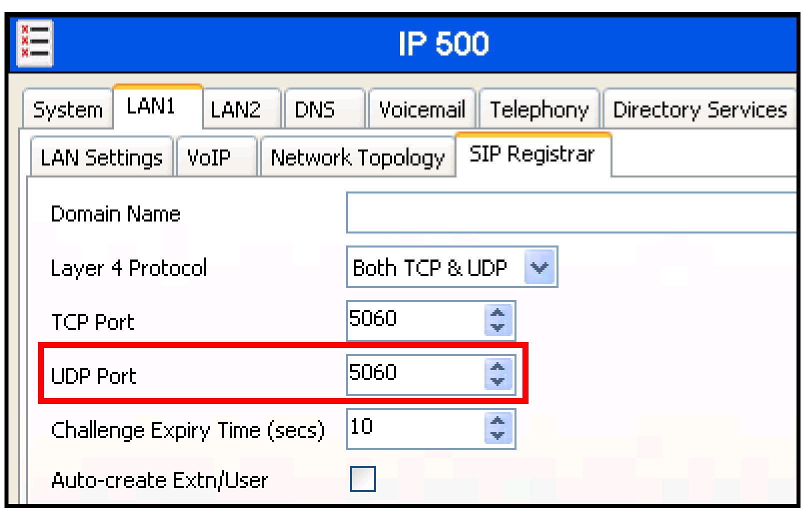

Figure 11: IP Office System: LAN1 SIP Registrar Tab

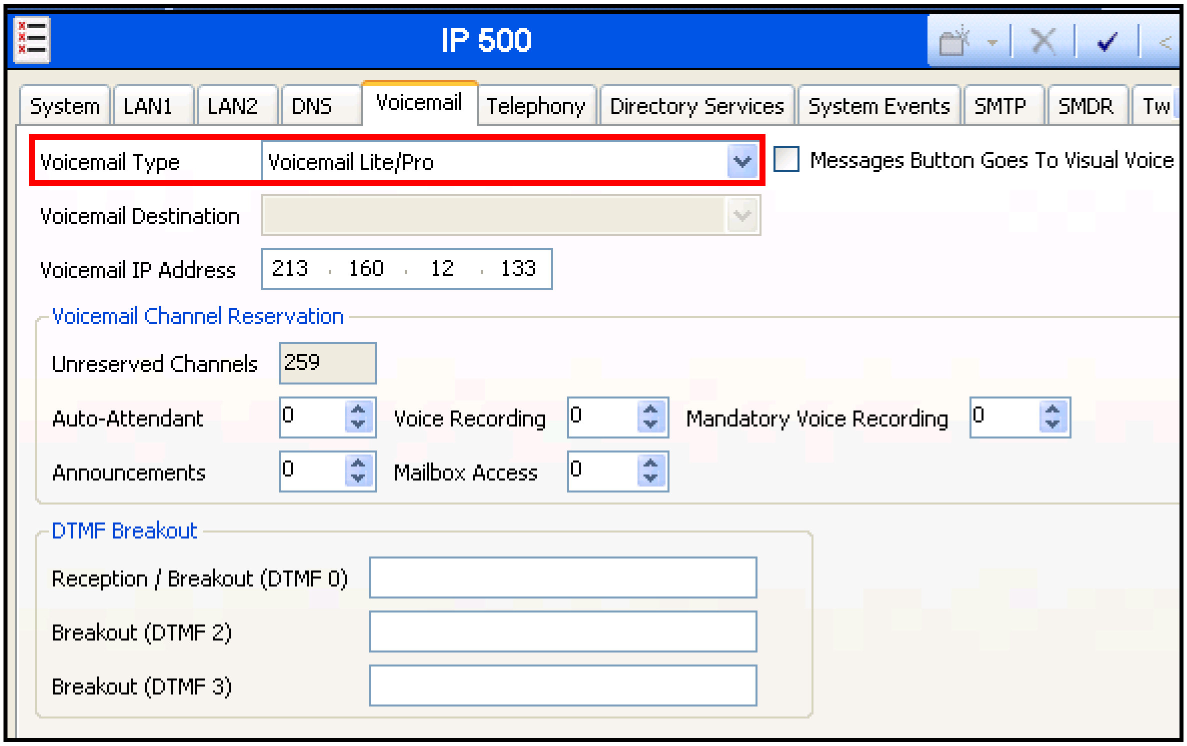

Figure 12: IP Office System: LAN1 Voicemail Tab

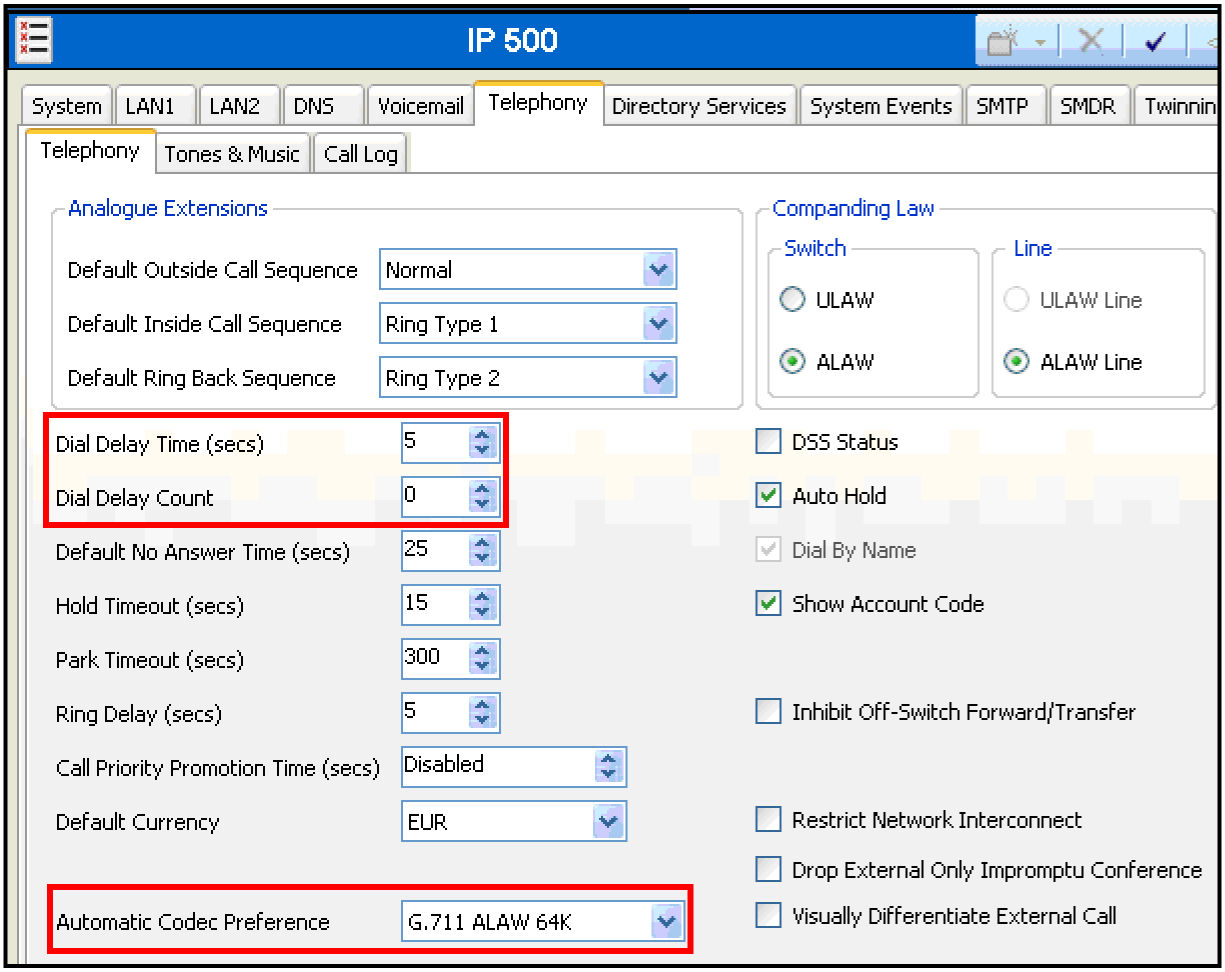

Figure 13: IP Office System: Telephony Tab

4.3. Default Gateway

Select the “IP-Route” icon shown in **Figure 6** and create a route with the parameters shown in the following table.

Parameter	Usage
IP Address	Enter “0.0.0.0”.
IP Mask	Enter “0.0.0.0”.
Gateway IP Address	Enter the address of the router which is used to attach IP Office to the Comdasys VoIP Network.
Destination number	Select “LAN1” from the drop-down list.

Table 5: IP Office Route: Default Gateway Parameters

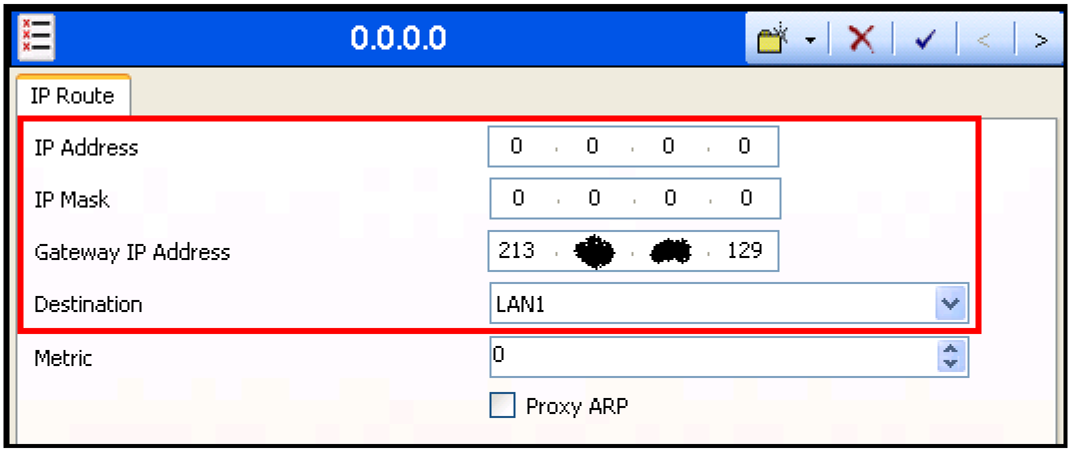

Figure 14: IP Office Route: Default Gateway

4.4. SIP Trunk

Select the "Line" icon shown in **Figure 6** and add a new SIP line using the parameters shown in the following table.

Tab	Parameter	Usage
SIP Line	ITSP Domain Name	Enter the IP address assigned to the Comdasys Mobile Convergence Controller SIP trunk.
	ITSP IP Address	Enter the IP address assigned to the Comdasys Mobile Convergence Controller SIP trunk.
	In Service	Check this box.
SIP URI	Local URI	Enter "*" to match all incoming SIP invitations.
	Contact	Enter "*" to match all incoming SIP invitations.
	Display Name	Enter "*" to match all incoming SIP invitations.
	Incoming Group ID	Enter the line number assigned to the SIP line.
	Outgoing Group ID	Enter the line number assigned to the SIP line.
VoIP	Compression Mode	Select "Automatic Select".

Table 6: IP Office SIP Line Parameters

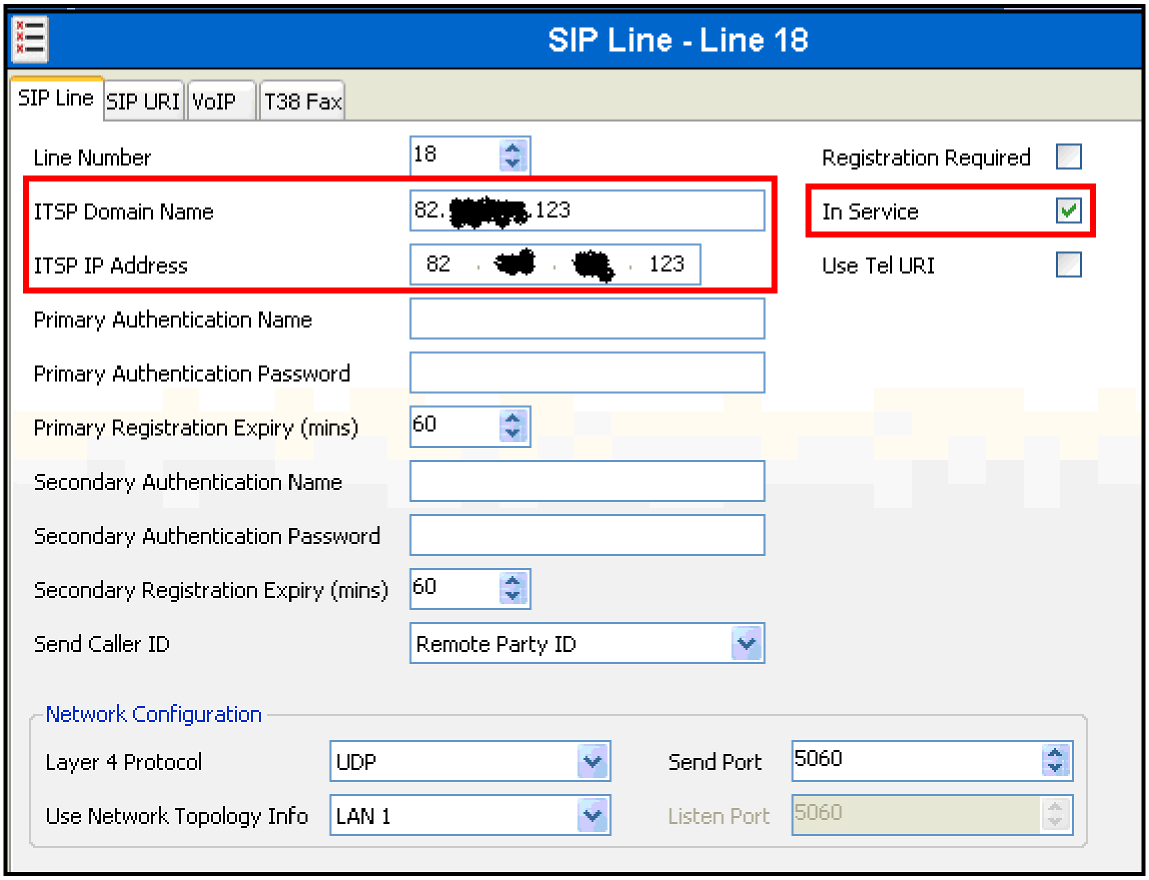

Figure 15: IP Office Line: SIP Line Tab

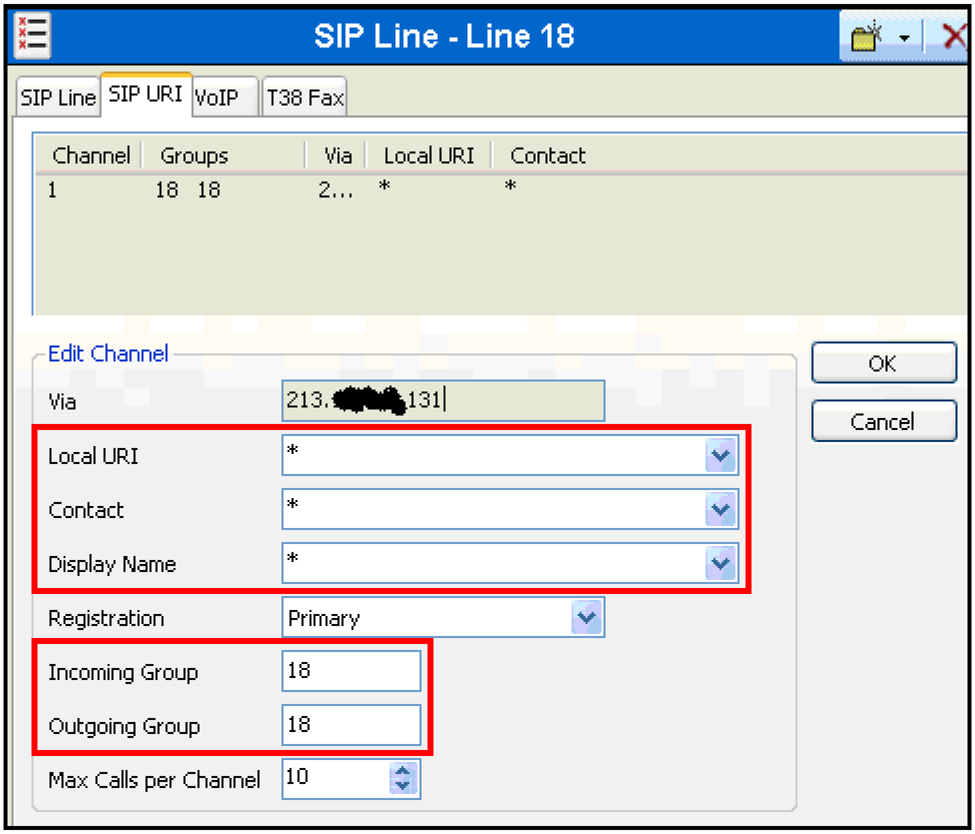

Figure 16: IP Office Line: SIP URI Line Tab

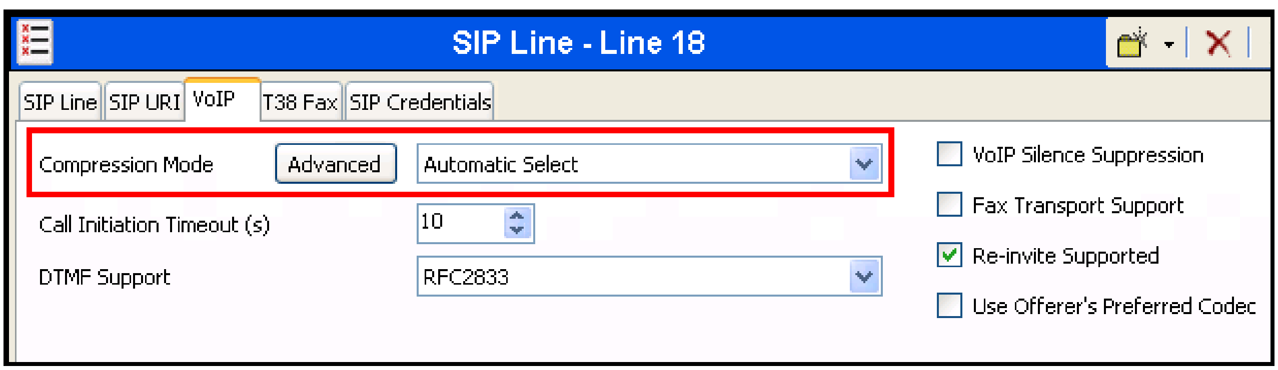

Figure 17: IP Office Line: VoIP Tab

4.5. PSTN Line

Select the "Line" icon shown in **Figure 6** and add a new line to the PSTN as shown in **Figure 1** using the parameters shown in the following table. The parameters shown here are for the E1 line which was used for testing. The configuration for this step will vary depending on the type of PSTN line that is used.

Parameter	Usage
Line SubType	Select "ETSI" from the drop-down menu for and E1 line.
Incoming Group ID	Enter an available group ID number.
Outgoing Group ID	Use the same value used for "Incoming Group ID".
Prefix	Enter the dial prefix used to dial local PSTN numbers.
National Prefix	Enter the dial prefix used to dial national PSTN numbers.
International Prefix	Enter the dial prefix used to dial international PSTN numbers.

Table 7: IP Office PSTN Line Parameters

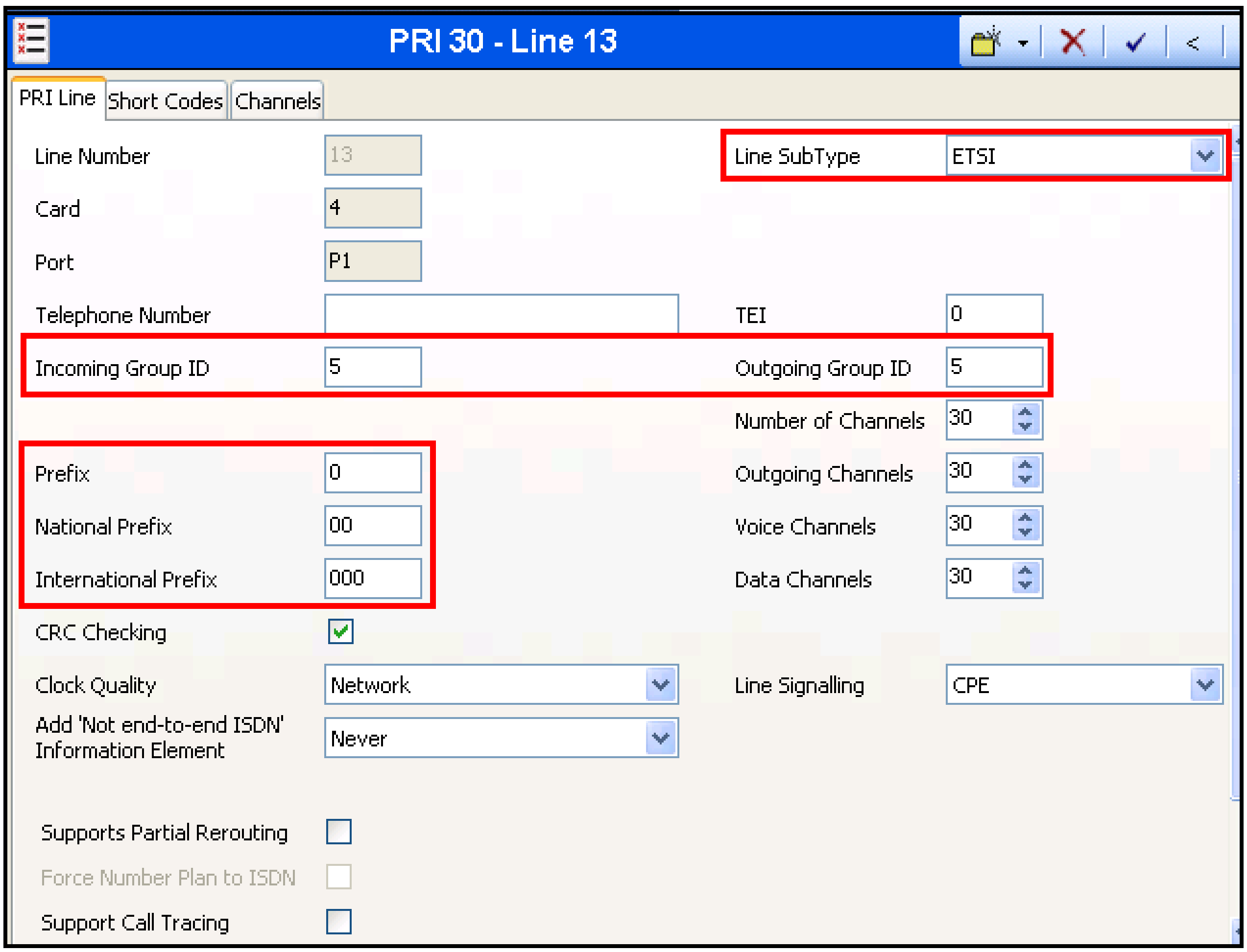

Figure 18: IP Office Line: PRI Line Tab

4.6. Mobile Endpoints

Select the "Extensions" icon shown in **Figure 6**, create an extension for a SIP telephone, and enter the extension in the "Base Extension" field. Repeat this for each extension shown in **Table 1.**

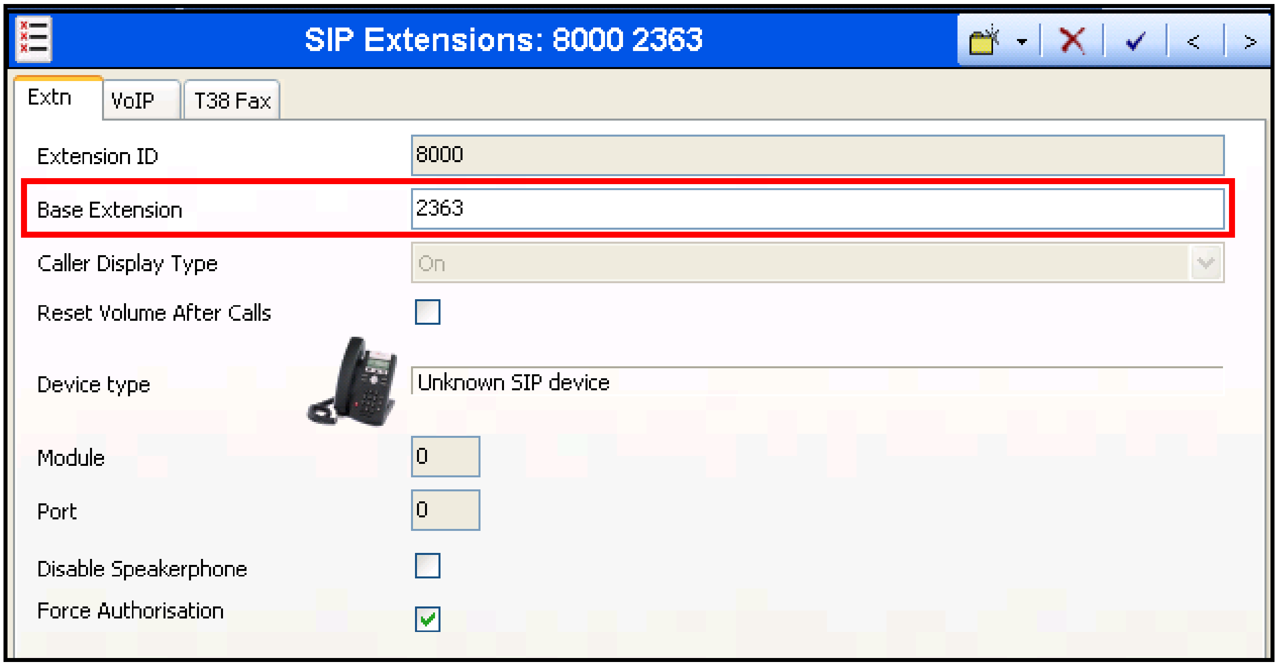

Figure 19: IP Office Local Telephone Extension: Extn Tab

Select the "Users" tab shown in **Figure 6** and add a new user for each local telephone shown in **Table 1**, using the parameters shown in the table below.

Tab	Parameter	Usage
User	Name	Enter an appropriate name to be assigned to the user.
	Extension	Enter the local extension to be assigned to the user.
Voicemail	Voicemail On	Check this box.
Telephony / Call Settings	Call Waiting On	Check this box.
Telephony / Supervisor Settings	Login Code	Enter the password to be used client authorization.
SIP	SIP Name	Enter the DID which is assigned to the user.
	SIP Display Name	Enter an appropriate name to be assigned to the user.
	Contact	Enter the DID which is assigned to the user.

Table 8: IP Office User Parameters

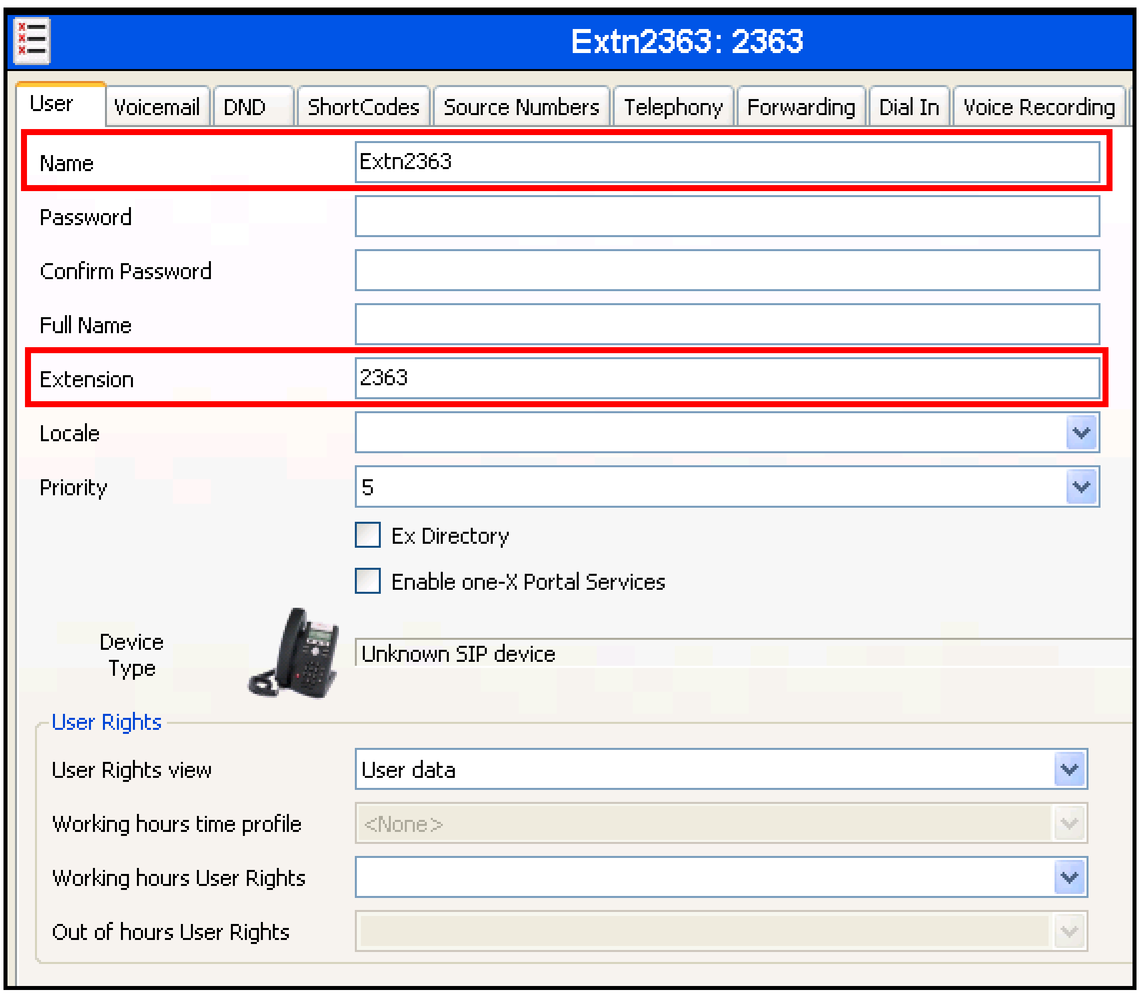

Figure 20: IP Office Local Telephone User: User Tab

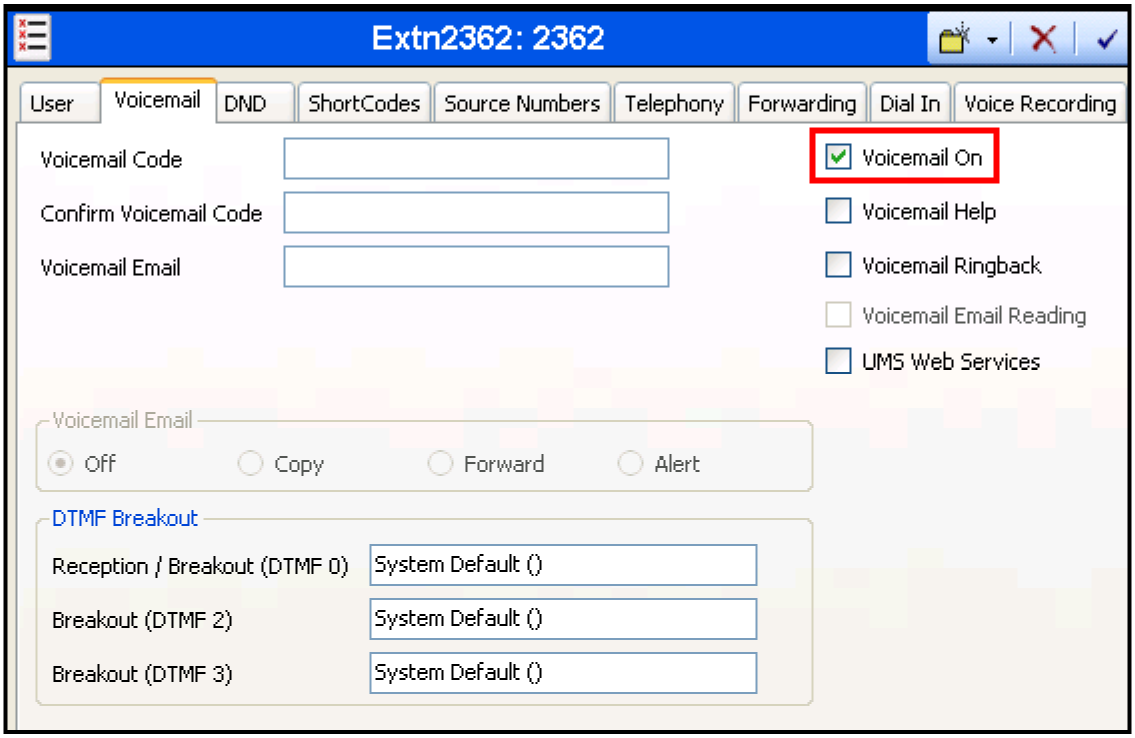

Figure 21: IP Office Local Telephone User: Voicemail Tab

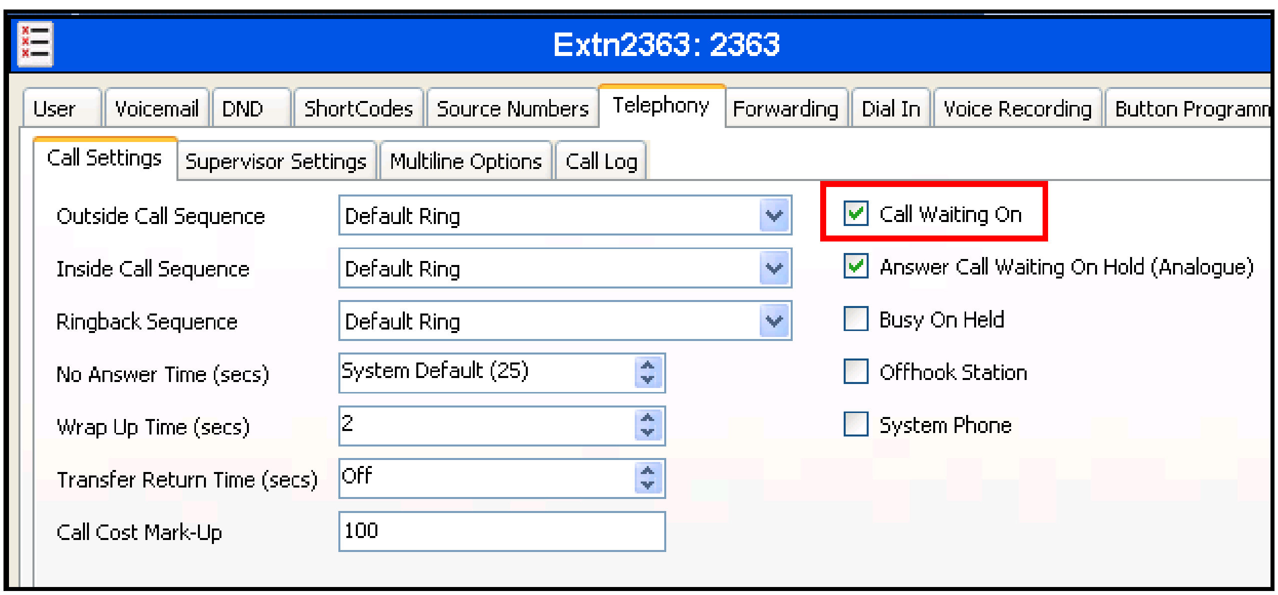

Figure 22: IP Office Local Telephone User: Telephony/Call Settings Tab

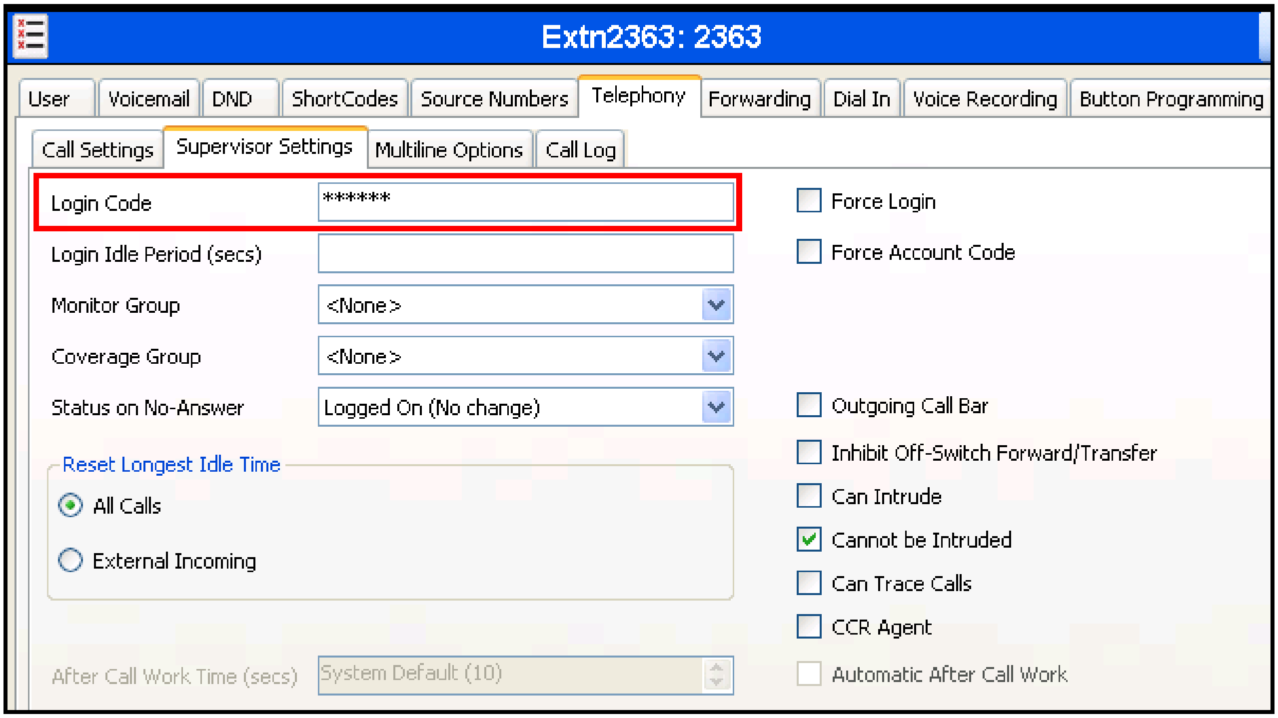

Figure 23: IP Office Local Telephone User: Telephony/Supervisor Settings Tab

Figure 24: IP Office Local Telephone User: SIP Tab

4.7. Outgoing Call Routing

4.7.1. Outgoing PSTN Call Routing

Create a shortcode to route outgoing calls from Avaya IP Office to the PSTN. Select the "Shortcode" icon shown in **Figure 6** and create a new shortcode with the values shown in the following table.

Parameter	Usage
Code	Enter 0N;
Feature	Select "Dial" from the drop-down menu.
Telephone Number	Enter NSi<trunk>E, where <trunk> is the prefix for the PRI trunk to the PSTN.
Line Group Id	Enter the line group number assigned to the PSTN Line configured in **Figure 18**.

Table 9: IP Office Outgoing Call Shortcode Parameters

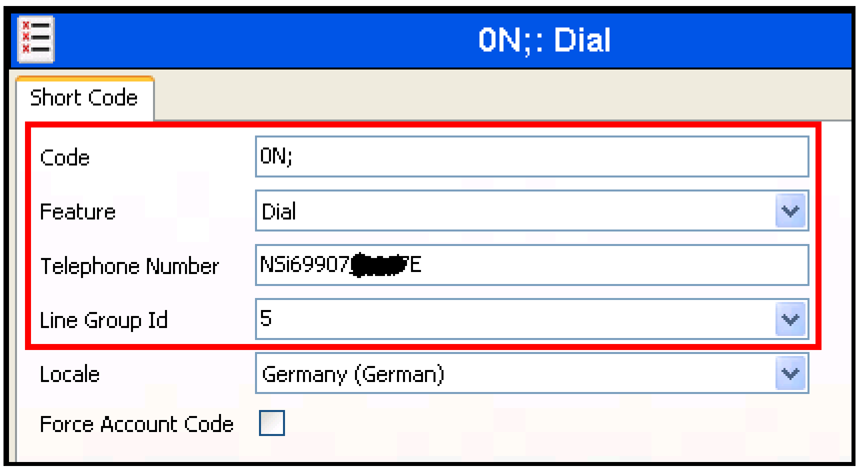

Figure 25: IP Office Outgoing Call Shortcode

4.7.2. Outgoing Special FMC Code Routing

Create shortcodes to route outgoing calls for the Special FMC Codes "Call Through", "Call Back" and "SIM Switch" codes shown in **Table 2**. Create a new shortcode with the values shown in the following table. Each of these codes has a corresponding Incoming Call Route, as shown in **Figure 31**.

Parameter	Usage
Code	Enter *< MC Code >*, where <MC Code> is the four digit Special FMC Code to be routed.
Feature	Select "Dial" from the drop-down menu.
Telephone Number	Enter the 4-digit Special FMC Code to be routed.
Line Group Id	Enter the line group number assigned to the SIP trunk in **Figure 16**.

Table 10: IP Office Outgoing Special FMC Number Shortcode Parameters

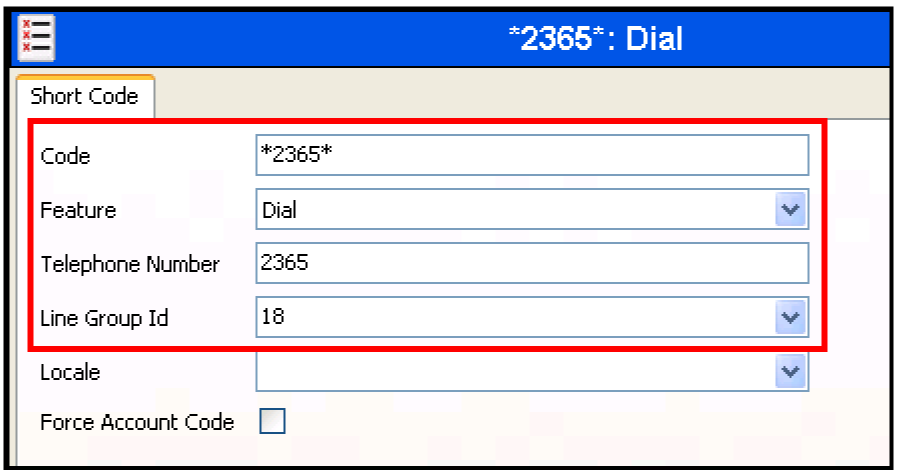

Figure 26: IP Office Special FMC Number Shortcode

4.8. Incoming Call Routing

4.8.1. Incoming PSTN Call Routing

Select the "Incoming Call Route" icon shown in **Figure 6** and create a new incoming call route with the values shown in the table below. This routes calls from the PSTN to the proper endpoint.

Tab	Parameter	Usage
Standard	Line Group Id	Enter the line group number assigned to the PSTN Line configured in **Figure 18**.
	Incoming Number	Enter the telephone number assigned to the local PSTN trunk followed by the sequence "xxxx" to serve as a placeholder for the local extensions.
Destinations	Destination	Enter "#" which will be replaced by the "xxxx" local extension which matches the "xxxx" in the previous step.

Table 11: IP Office PSTN Incoming Call Route Parameters

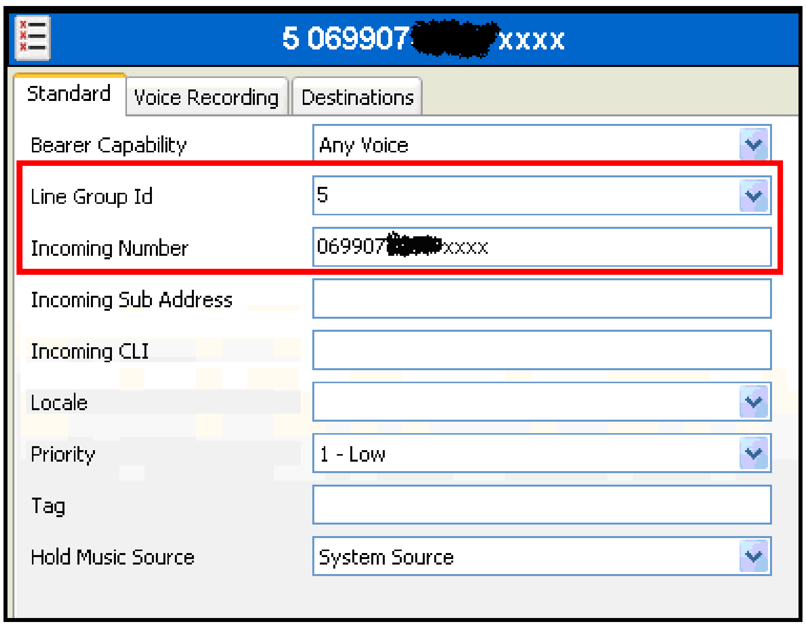

Figure 27: IP Office PSTN Incoming Call Route: Standard Tab

Figure 28: IP Office PSTN Incoming Call Route: Destinations Tab

4.8.2. Incoming SIP Call Routing

Select the "Incoming Call Route" icon shown in **Figure 6** and create a new incoming call route with the values shown in the following table. This routes calls to the PSTN using the SIP trunk.

Tab	Parameter	Usage
Standard	Line Group Id	Enter the line group number assigned to the SIP Line configured in **Figure 16**.
	Incoming Number	Leave this field blank.
Destinations	Destination	Enter "." which will be replaced by the "xxxx" local extension which matches the "xxxx" in the previous step.

Table 12: IP Office SIP Trunk Incoming Call Route Parameters

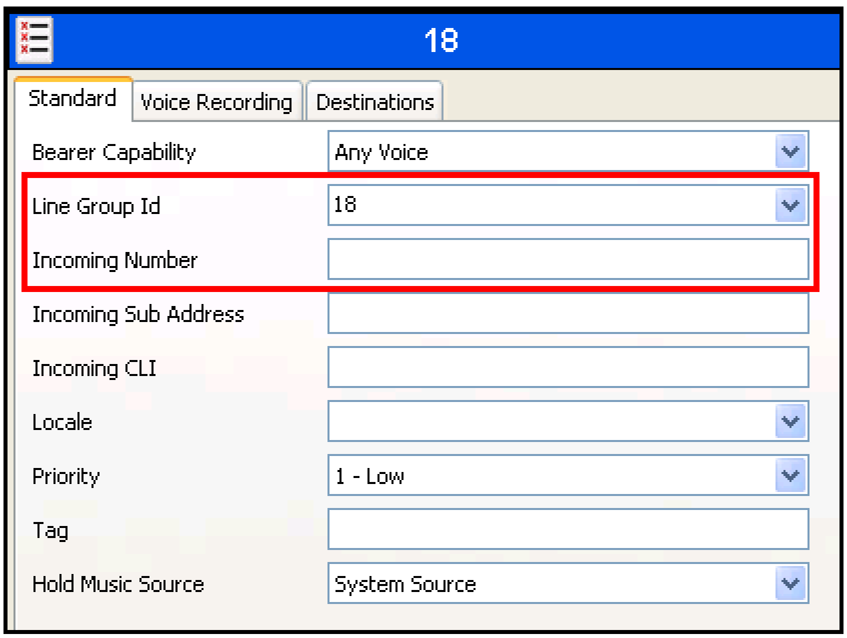

Figure 29: IP Office SIP Incoming Call Route: Standard Tab

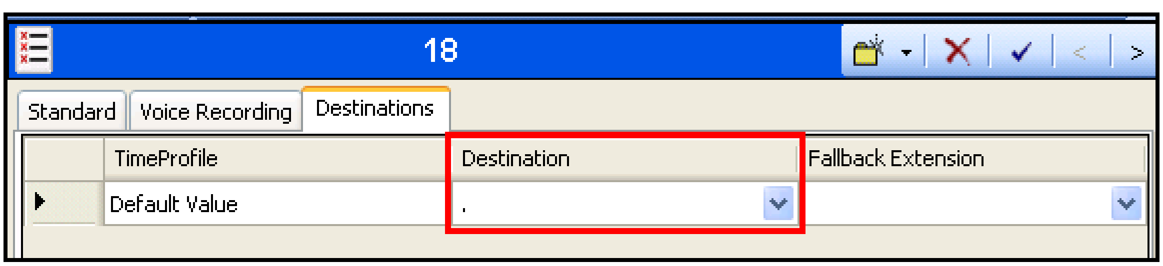

Figure 30: IP Office SIP Incoming Call Route: Destinations Tab

4.8.3. Incoming SIP Special FMC Number Routing

Select the “Incoming Call Route” icon shown in **Figure 6** and create a new incoming call route with the values shown in the table below for each Special FMC Number shown in **Table 1**. This routes Special FMC Number calls from the PSTN to the Comdasys Server via the SIP trunk.

Tab	Parameter	Usage
Standard	Line Group Id	Enter the line group number assigned to the PSTN trunk configured in **Figure 18**.
	Incoming Number	Enter the telephone number assigned to the local PSTN trunk followed by the sequence the <MC Code> extension shown in **Table 1**.
Destinations	Destination	Enter the shortcode *<MC Code>*, where the <MC Code> is one of the Special FMC Number entries contained in **Table 1**.

Table 13: IP Office Special FMC Number Call Route Parameters

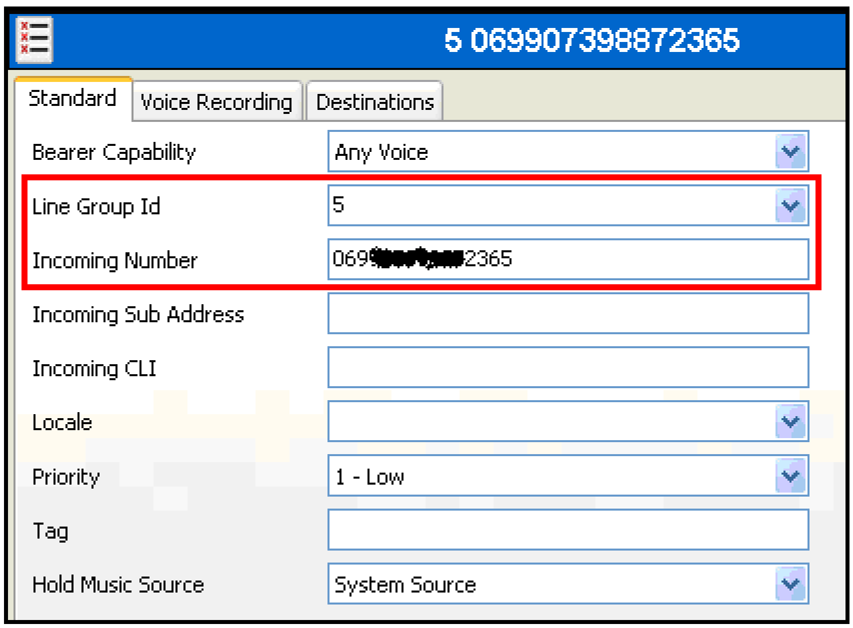

Figure 31: IP Office Special FMC Number Call Route: Standard Tab

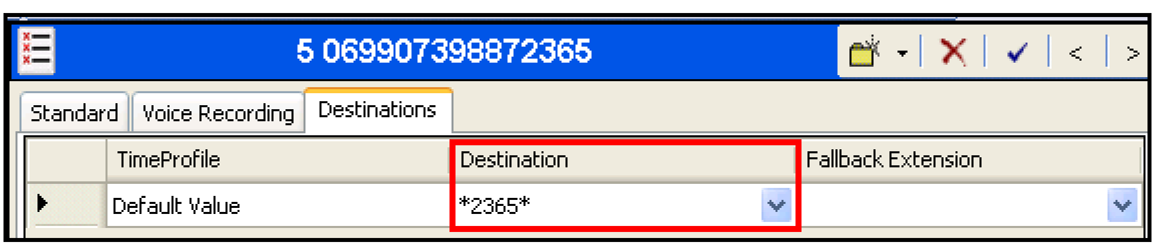

Figure 32: IP Office Special FMC Number Call Route: Destinations Tab

5. Comdasys Mobile Convergence Controller Configuration

Only one PBX or Avaya IP Office can be connected to the MC Controller if “disable separate profiles” registration mode is used. This is mandatory for the interworking between the Comdasys MC Controller and the Avaya IP Office. Contact the Comdasys Support if you plan to connect the MC Controller to more than one PBX at the same time.

The Mobile Convergence Controller has an integrated HTTP server, allowing configuration actions to be performed via a web browser: select the IP address of the Mobile Convergence Controller and perform the configuration steps shown in the remainder of this section. The following view illustrates the main configuration screen:

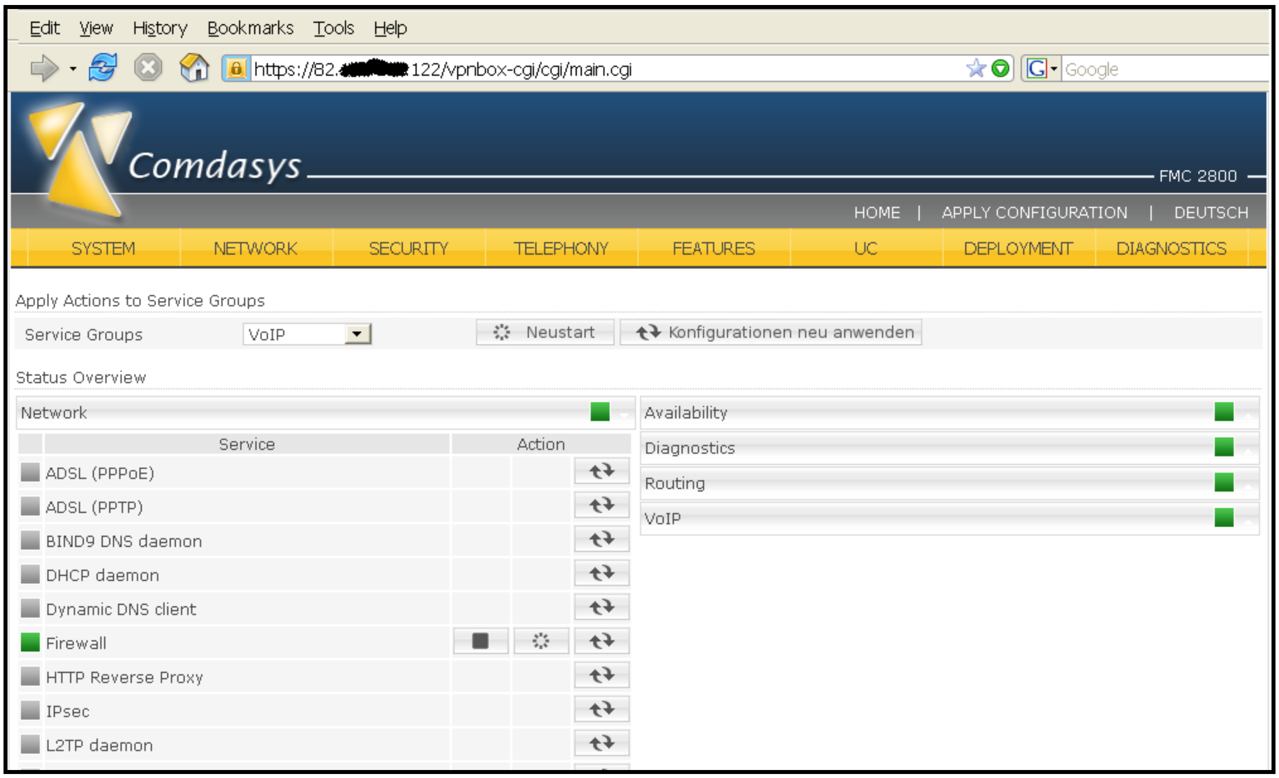

Figure 33: MC Controller Main View

5.1. Network Configuration

Two separate IP addresses are required on the LAN side in order to connect to the Avaya IP Office. The first address (typically LAN interface 1) is used for SIP subscriber registrations while the second address (LAN interface 2) is used for the SIP trunk. Only the LAN1 interface must be connected physically to the Network. The second interface IP should be in the same IP subnet and will be routed through the first physical (LAN 1) interface. No physical LAN 2 port connection is required. Click on the "Network" tab and set these configuration values as shown in the following table.

Interface	Parameter	Usage
LAN 1	IP Address	Set this to the IP address which SIP clients use to register with IP Office.
	Netmask	Set this value as required by the LAN to which the MC Controller is attached.
LAN 2	IP Address	Set this value to the IP address of the SIP trunk to IP Office. This should match the "ITSP IP Address" parameter in **Figure 15**.
	Netmask	Set this value as required by the LAN to which the MC Controller is attached.

Table 14: MC Controller LAN Configuration Parameters

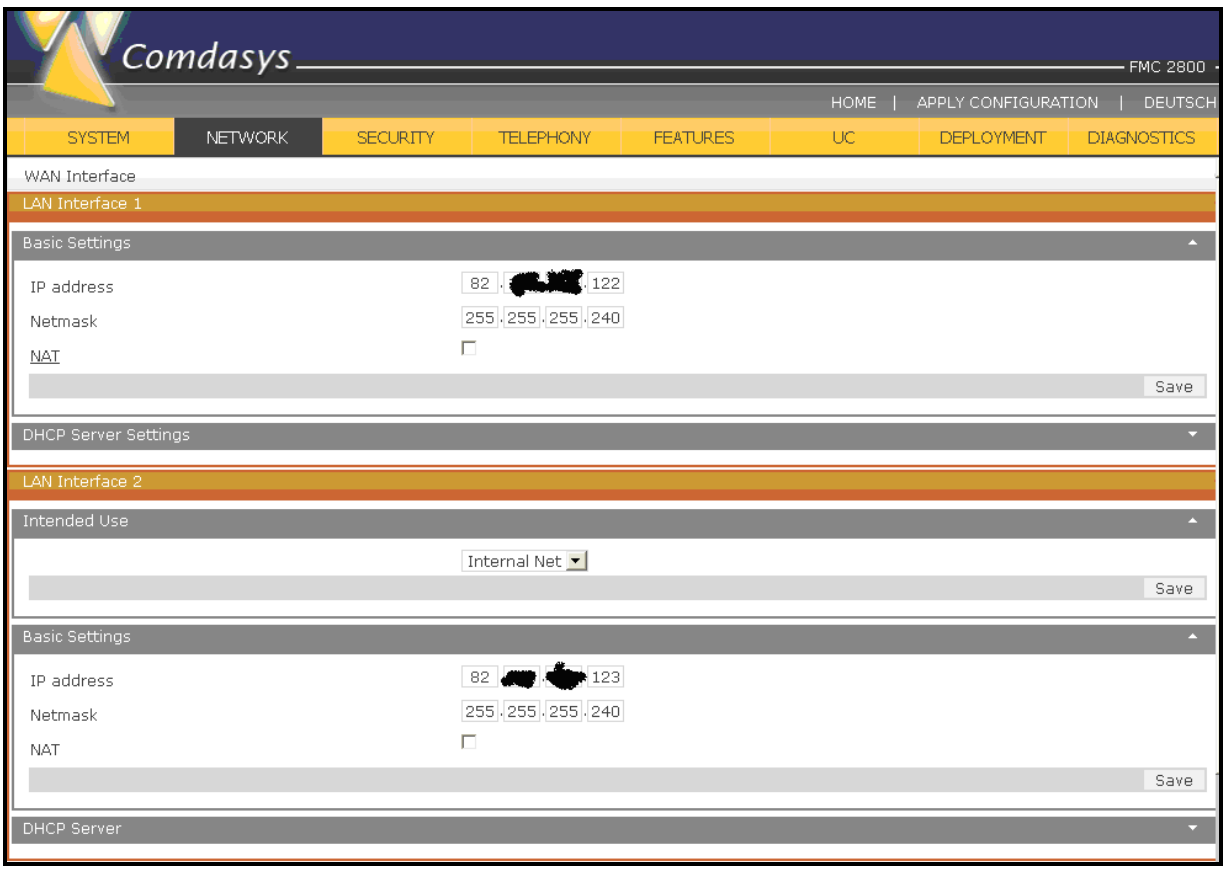

Figure 34: LAN Configuration

5.2. Telephony Global Settings

Set these configuration values as shown in the following table. Refer to the Comdasys FMC Mobile Convergence – Administrator Manual **[3]** for a detailed explanation of these settings.

Parameter	Usage
Enable Call-Through Early Media	Check this box.
Enable Client Early Media	Check this box.
Enable busy sound in WiFi	Check this box.
Disable Inband DTMF Detection	Check this box.

Table 15: MC Controller LAN Configuration Parameters

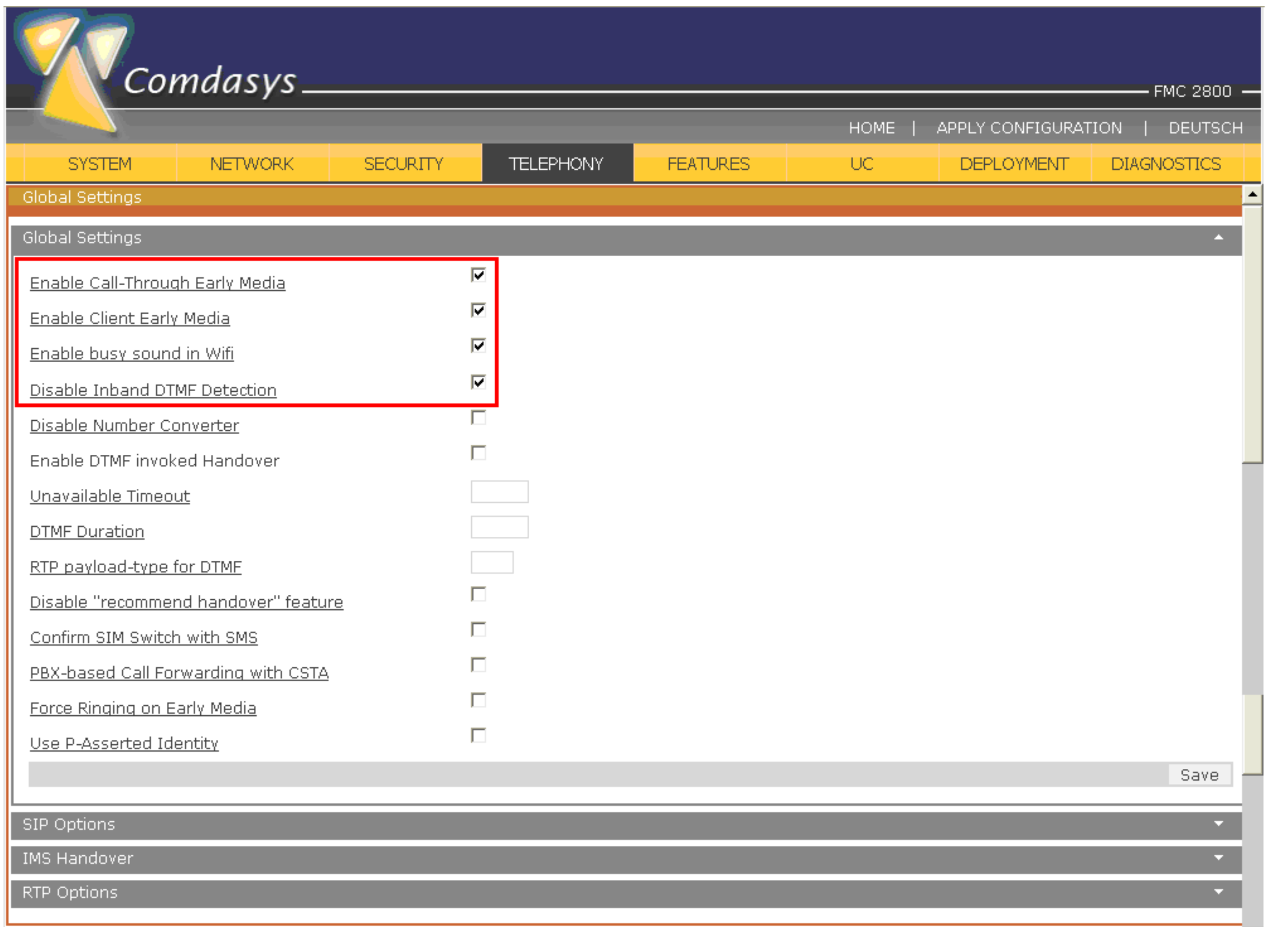

Figure 35: Global Settings

5.3. Telephony Settings

Click the "Telephony" tab to access the individual telephony settings screens.

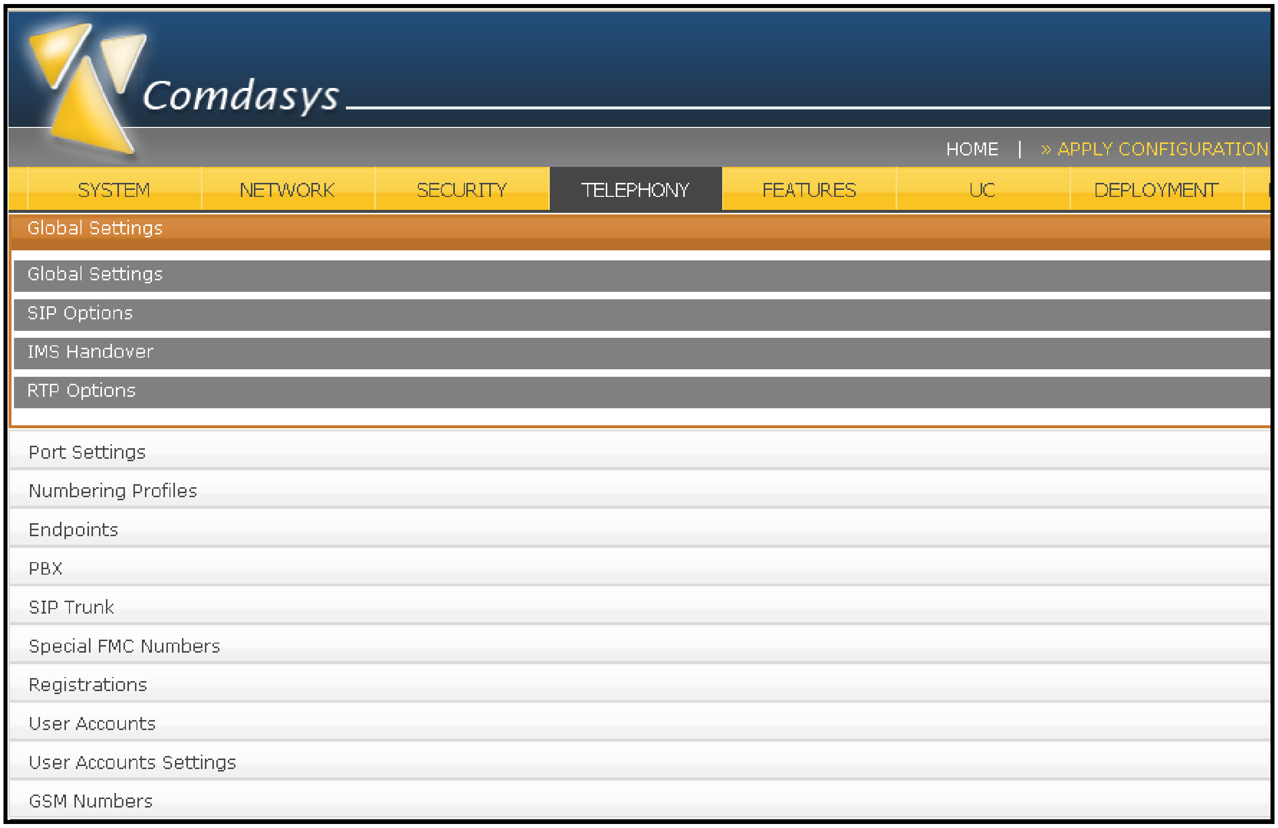

Figure 36: Telephony Tab

5.3.1. Telephony Port Settings

From the "Telephony" tab shown in **Figure 36**, select "Port Settings" and configure the port settings as shown below. These settings are required when working with the "disable separate profiles" registration.

When using one or more Special FMC Numbers on LAN 1 and source port 5060 the system will already use the local port 5060. Therefore, set up the SIP PBX Start Port 5059 here (because 5060 is already in use) to get a valid setup.

Parameter	Usage
SIP Client Port	This value must be less than the SIP PBX Start Port (this is an internal requirement).
SIP PBX Start Port	This value must be set to 5059 (default is 12000).
SIP Trunk Start Port	This value must be set to 5060.

Table 16: MC Controller Port Setting Parameters

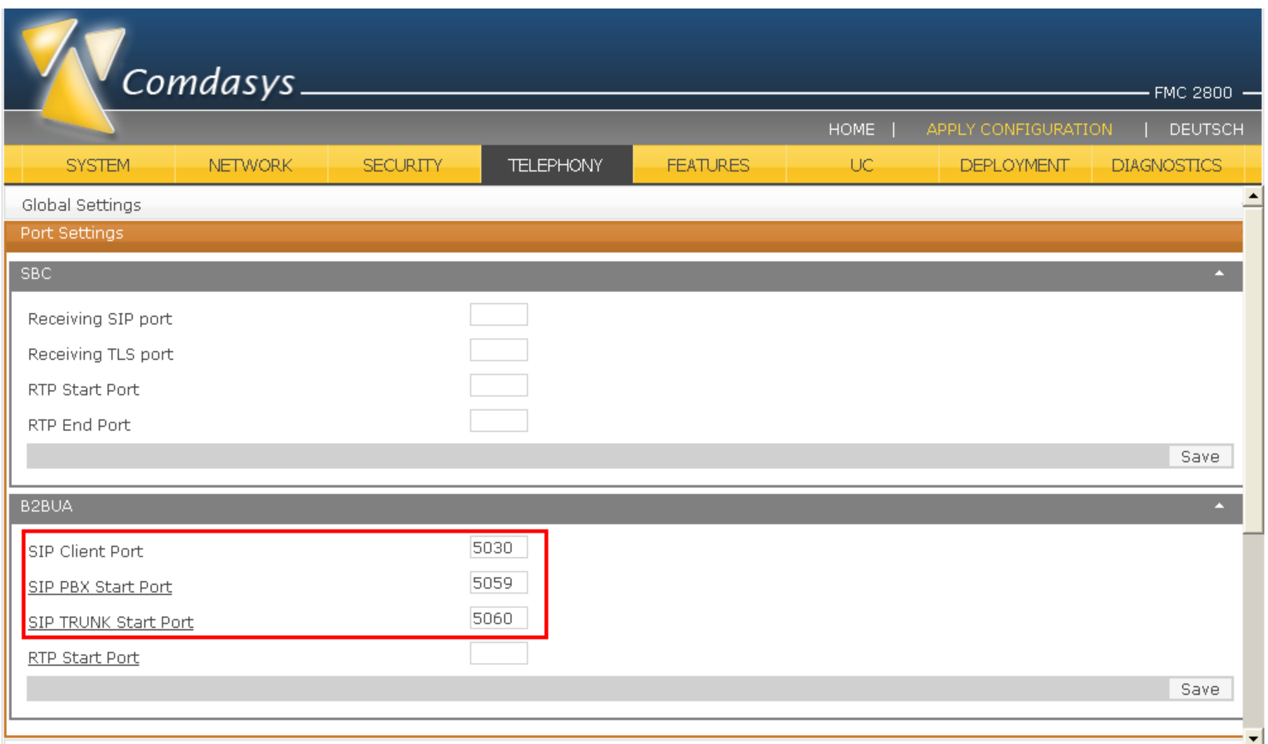

Figure 37: Port Settings

5.3.2. Numbering Profiles

From the "Telephony" tab shown in **Figure 36**, select "Numbering Profiles" and configure the numbering profiles as shown in the following table. Consult the MC Controller Manual **[3]** for information on adaptation to the local Country / PBX.

Parameter	Usage
Name	Enter a descriptive name to be used to identify this profile.
Country Code	Enter the local country code, i.e. "49" for Germany.
Country Prefix	Enter the prefix "00" which is used to dial international numbers.
Area Code	Enter the local area code, i.e. "69" for Frankfurt.
Area Prefix	Enter the prefix "0" which is used to dial national numbers.
Outgoing Prefix	Enter a "0" here to if you need to dial another zero for getting an external line to the PSTN.
Internal Length	Enter the length of the extension numbers contained in **Table 1**.

Table 17: MC Controller Numbering Profiles Parameters

Figure 38: Numbering Profiles

5.3.3. Endpoints

From the "Telephony" tab shown in **Figure 36**, select "Endpoints" and configure the endpoints as shown in the following table.

Configure two endpoints. The first endpoint is used for the SIP subscriber registration and should run on the "LAN 1" interface. Set the authentication realm to "ipoffice" unless this was not changed on the IP Office side. The second endpoint is used for routing calls from and to the mobile handset when used in GSM mode. For an installation where the mobile integration outside of WIFI coverage is not needed or not used, the configuration of the second endpoint as well as the Trunk and the Special FMC Numbers configuration can be skipped.

Endpoint Name	Parameter	Usage
IP Office	Hostname/IP	Enter the IP address of IP office as configured in **Figure 9.**
	Local Interface	Enter "LAN 1". This should correspond to the "IP Address" parameter in **Figure 9**.
	Foreign Port	Enter 5060. This should correspond to the "UDP Port" parameter in **Figure 11**.
	Realm	Enter "ipoffice".
	Preferred Codec	Enter "G.711 alaw / 20ms". This should be compatible with the "Compression Mode" parameters in **Figure 17** and the "Automatic Codec Preference" parameter in **Figure 13**.
IP Office-trunk	Hostname/IP	Enter the IP address of IP office as configured in **Figure 9.**
	Local Interface	Enter "LAN 2". This should correspond to the IP Address for the LAN interface selected by the "Use Network Topology Info" parameter shown in **Figure 15**.
	Foreign Port	Enter 5060. This should correspond to the "Listen Port" shown in **Figure 15**.
	Realm	Enter "ipoffice".
	Preferred Codec	Enter "G.711 alaw / 20ms". This should be compatible with the "Compression Mode" parameters in **Figure 17** and the "Automatic Codec Preference" parameter in **Figure 13**.

Table 18: MC Controller Endpoints Parameters

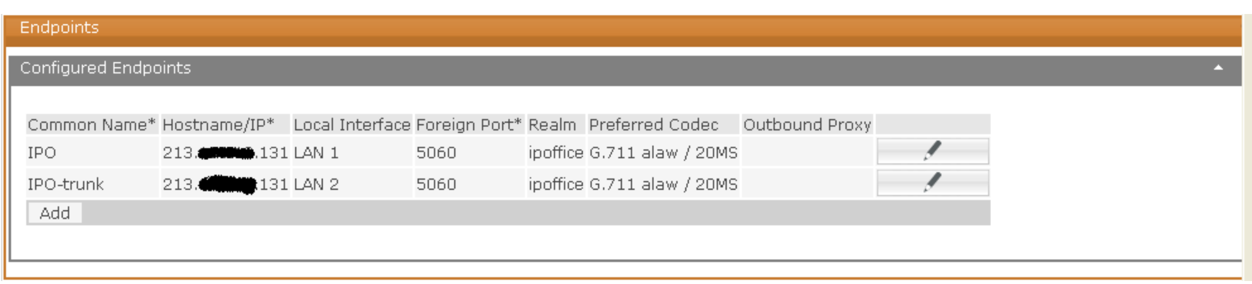

Figure 39: Endpoints

5.3.4. SIP Trunk

From the "Telephony" tab shown in **Figure 36**, select "SIP Trunk" and configure the SIP trunk as shown in the following table. A SIP trunk is required between the MC Controller and the IP Office in order to support the full GSM integration.

Figure 40: SIP Trunk

5.3.5. PBX

From the "Telephony" tab shown in **Figure 36**, select "PBX" and configure a PBX using the parameters shown in the following table.

Parameter	Usage
Common Name	Enter an appropriate name to identify the IP Office.
Endpoint	Enter the endpoint created for LAN 1 in **Figure 39**.
From Converter Profile	Select the profile created in **Figure 38**.
To Converter Profile	Select the profile created in **Figure 38**.
SIP Trunk	Select the trunk created in **Figure 40**.
Country	Select the country in which the system is located.
DTMF Type	Select RFC2833. This should correspond to the "DTMF Support" parameter selected in **Figure 17**.
Encode URI	Enter "Disabled".
Mode	Select the registration mode "Disable Separate Profile". This setting is mandatory for Avaya IP Office version 5.0 and 6.0 but might change in future versions.
Call Log Sync	Enter "Disabled".

Table 19: MC Controller PBX Parameters

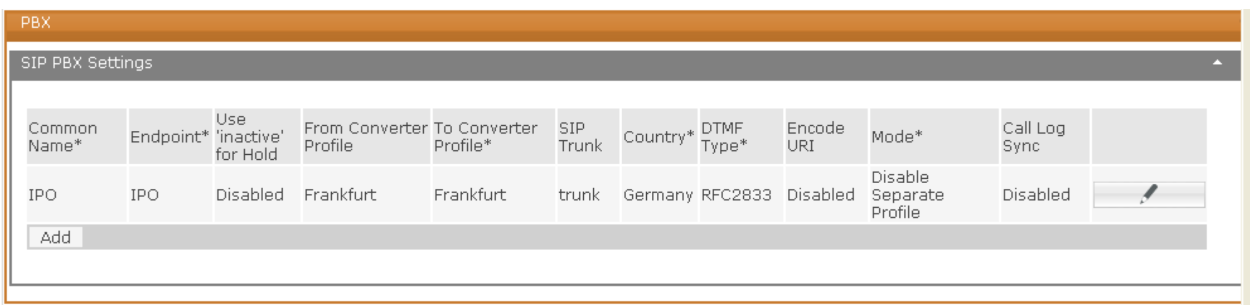

Figure 41: PBX

5.3.6. Special FMC Numbers

From the "Telephony" tab shown in **Figure 36**, select "Special FMC Number" and configure the FMC Numbers shown in **Table 2** using the parameters shown in the following table. Special MC Numbers are used to signal the MC Client to perform specific actions as described in **Table 2**. Configure each of the Special FMC Numbers shown in **Table 2** as shown in Error! Reference source not found..

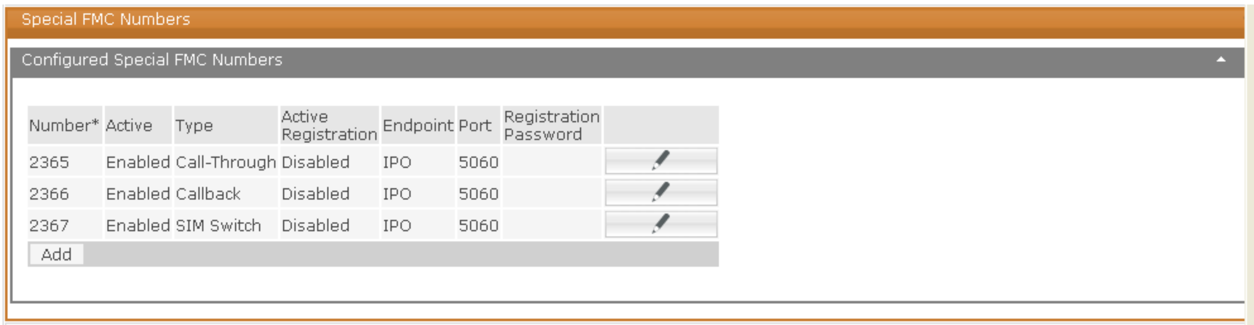

Figure 42: Special FMC Numbers

5.3.7. Registrations

From the "Telephony" tab shown in **Figure 36**, select "Registrations" and configure a registration for each local IP Office extensions listed in **Table 1**, using the parameters shown in the table below.

Parameter	Usage
PBX Username	Enter the IP Office extension.
PBX Password	Enter the User "Login Code" configured in **Figure 23**.
PBX Number	Enter the IP Office extension.
PBX	Select the PBX created in Error! Reference source not found..

Table 20: MC Controller Registrations Parameters

Figure 43: Registrations

5.3.8. User Accounts

From the "Telephony" tab shown in **Figure 36**, select "User Accounts" and configure a User Account for each local IP Office extensions listed in **Table 1**, using the parameters shown in the table below.

Parameter	Usage
SIP Number	A unique number must be assigned which cannot be the same as the number of the registration used by Avaya IP Office. As a general rule, it is recommended to simply add a "1" in front of the extension.
SIP User Password	Enter the User "Login Code" configured in **Figure 23**.
GSM Number	Enter the GSM number of the endpoint shown in **Table 1**.
Registrations	Enter the IP Office extension of the endpoint followed by "@IPO".
Active User	Select "Enabled".

Table 21: MC Controller User Accounts Parameters

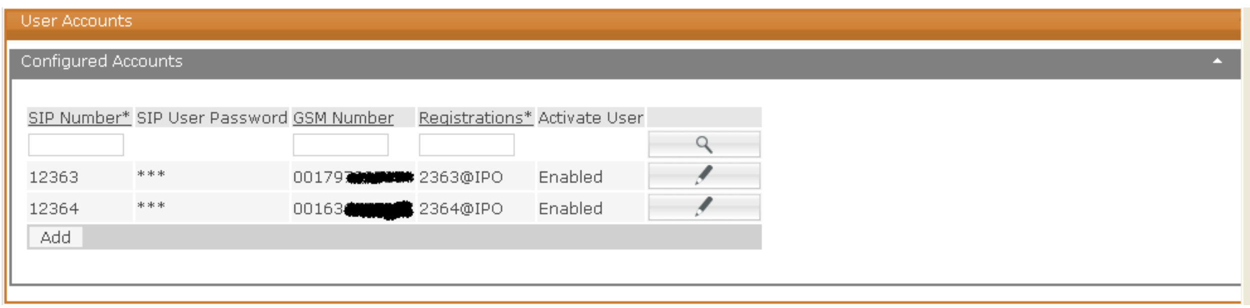

Figure 44: User Accounts

5.3.9. User Account Settings

From the "Telephony" tab shown in **Figure 36**, select "User Account Settings" and configure the User Account Settings for each local IP Office extensions listed in **Table 1**, using the parameters shown in the table below.

Parameter	Usage
SIP Number	This is the number which was created in Error! Reference source not found..
Static Roaming	Set this value to "Enabled".
Use DMC	Set this value to "Disenabled".
Call Waiting	Set this value to "Enabled".
Activate MWI	Set this value to "Enabled".
Call Reverse	Set this value to "Disenabled".
Security	Set this value to "Disenabled".
Activate DND	Set this value to "Disenabled".

Table 22: MC Controller User Account Settings Parameters

Figure 45: User Account Settings

5.3.10. GSM Numbers

From the "Telephony" tab shown in **Figure 36**, select "GSM Numbers" and configure the GSM number for each local IP Office extensions listed in **Table 1**, using the parameters shown in the table below. Each user is allowed to own more than one GSM number. However, only one GSM number can be active at the same time. The activation is done by selecting the right number in the "User Account" setup or by initiating a SIM switch call.

Figure 46: GSM Numbers

6. Comdasys MC Client Configuration

Installation of the Client is done via PC-Suite or deployment. Consult the Client manual off the specific phone platform (Symbian, Blackberry, Android or iPhone) about the Configuration items. From the MC Client, navigate to Options -> Setup and enter the setting shown in the following table:

Parameter	Usage
Registrar	Set this to the Registrar IP address of the MC Controller which is configured as “Basic Settings” “IP Address” in Error! Reference source not found..
Port	Set to the Default port 5062 of the MC Controller.
Username	Set it to the FMC user account “SIP Number” configured in Error! Reference source not found..
Password	Set this to the FMC ”SIP User Password” configured in Error! Reference source not found..
Call-through	Set the Call-through number contained in **Table 1** using international format.
Callback Number	Set the Callback number contained in **Table 1** using international format.

Table 23: MC Client Configuration Parameters

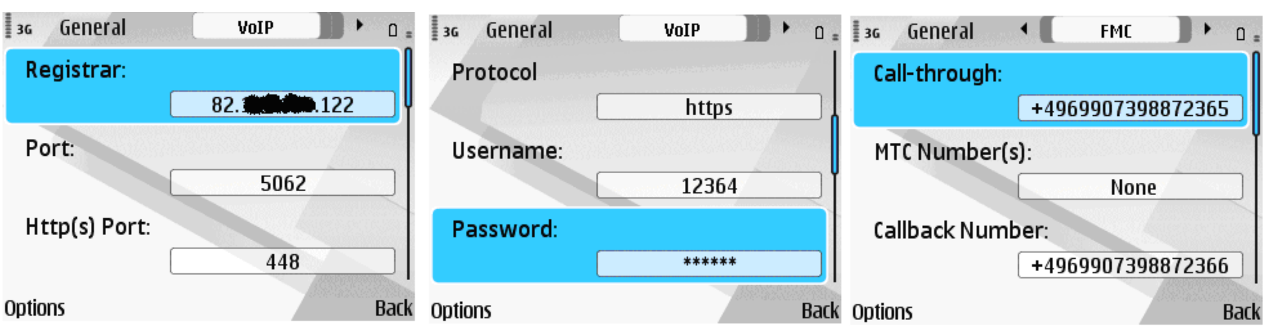

Figure 47: CM Client Configuration Screens

7. General Test Approach and Test Results

All tests were performed manually. Only functional testing was performed: no performance testing was done.

The following issues were encountered during testing:

- When the IP Office receives a "Temporarily Moved"-SIP status message from the MC Controller when it attempts to forward a call, it does not redirect the call to the endpoint indicated by the SIP "Contact" field, as this SIP feature which can be used to perform call forwarding is not supported by IP Office. The call is instead forwarded by IP Office to the voicemail account.
- When a call from a client to the PSTN via the IP Office is rejected by the called party, the call is not released by IP Office.

8. Verification Steps

The correct configuration of the system can be verified by performing the following steps:

- Use the FMC Status Page to check the "Active Endpoint Registration" status.

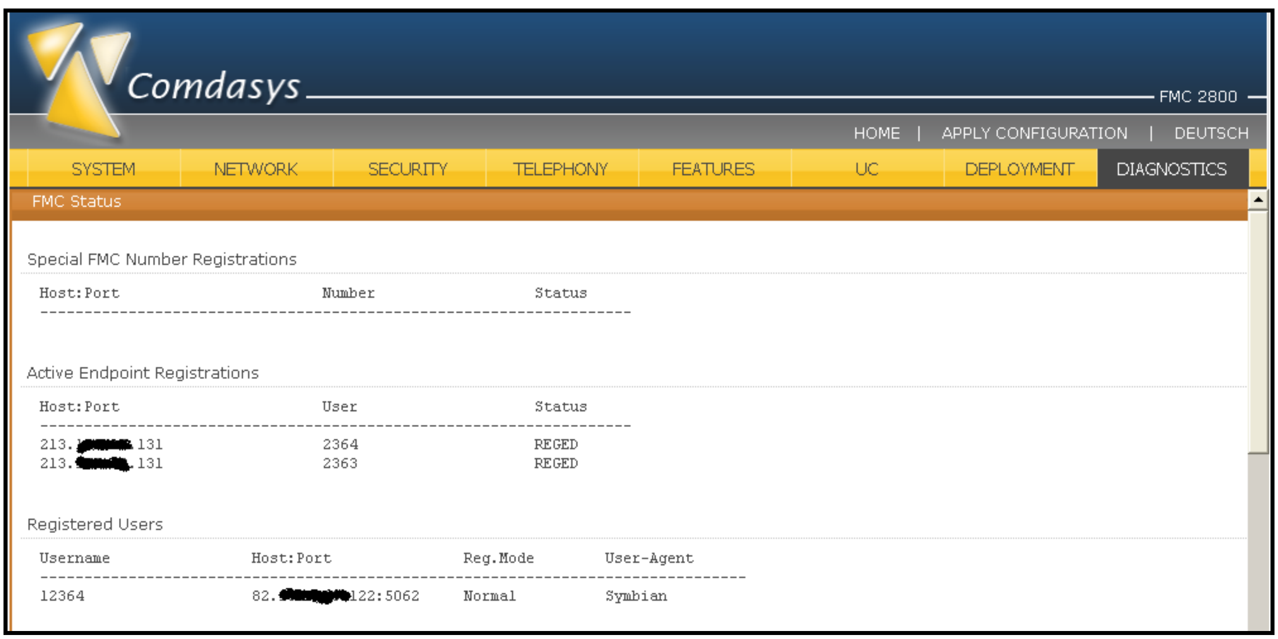

Figure 48: MC Controller Status

- Use the IP Office System Status program to verify that there are no alarms for the SIP trunk.

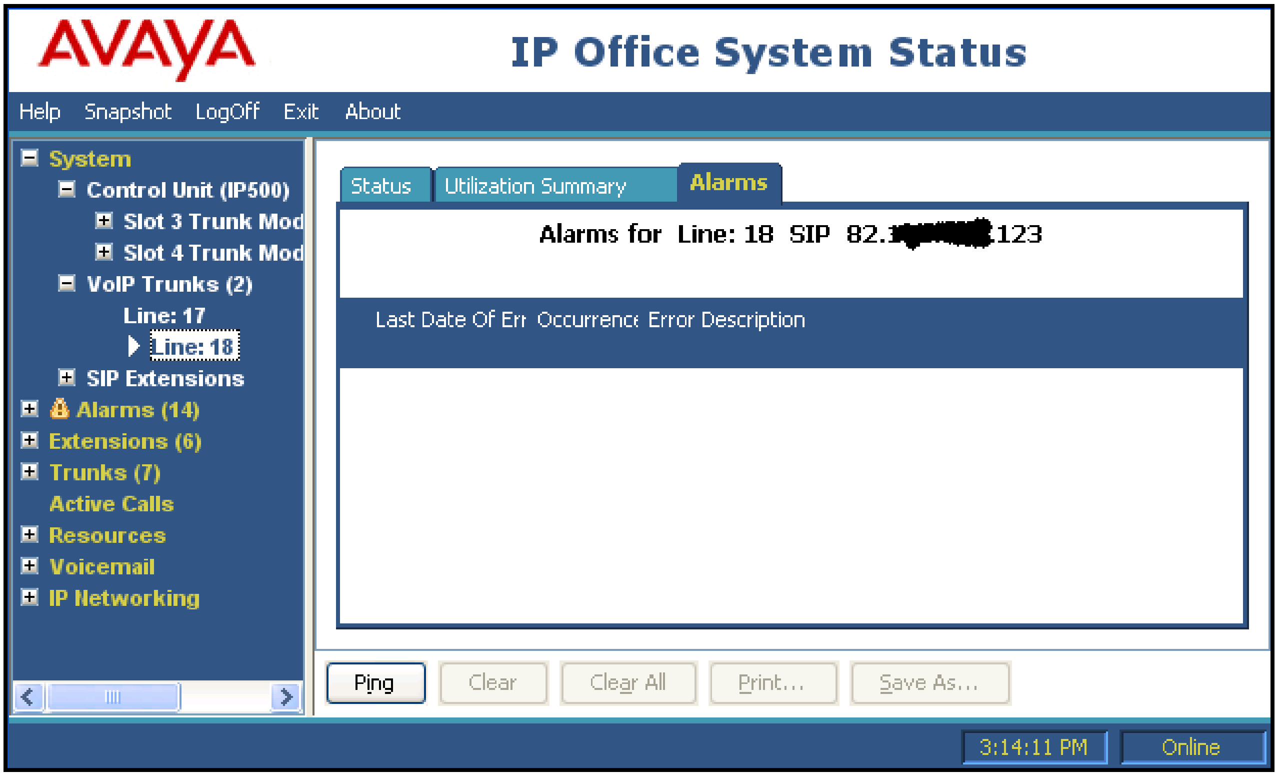

Figure 49: IP Office System Status SIP Trunk Alarms

- Use the IP Office System Status program to verify that all of the configured SIP trunk channels are in the “Idle” state.

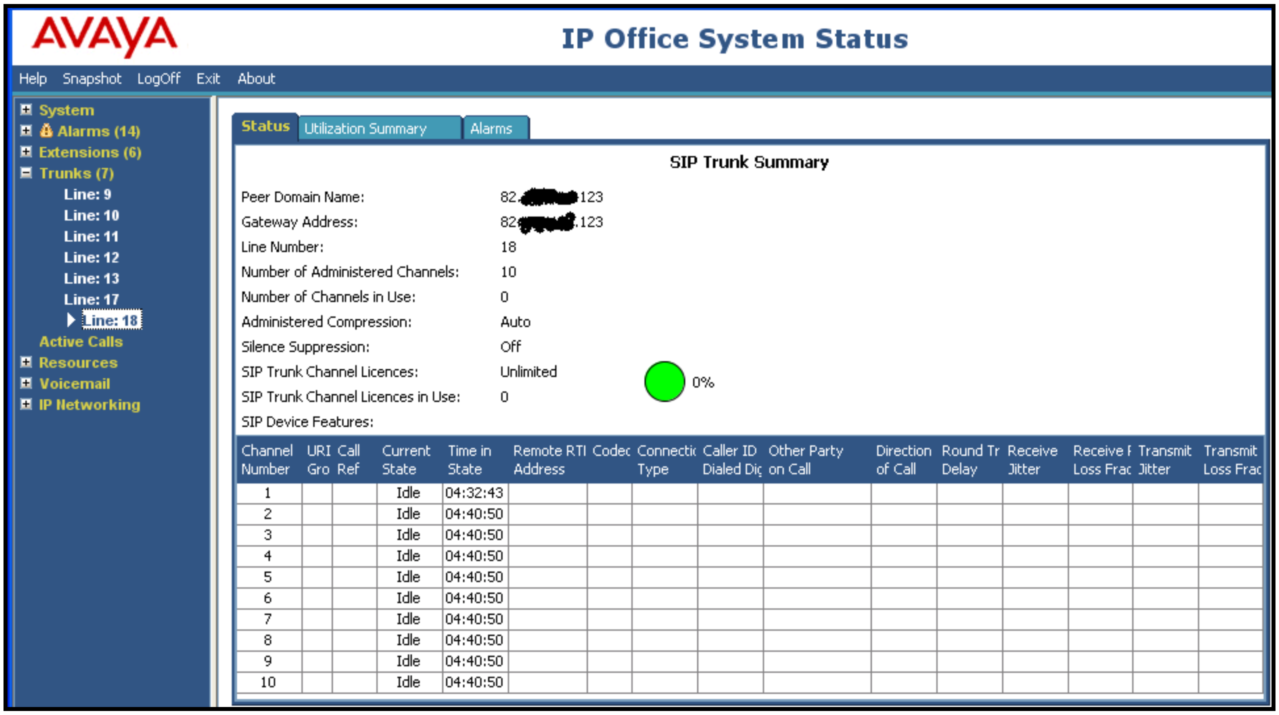

Figure 50: IP Office System Status SIP Channel Status

- Use the IP Office System Status program to verify that all of the SIP endpoints listed in **Table 1** have logged in.

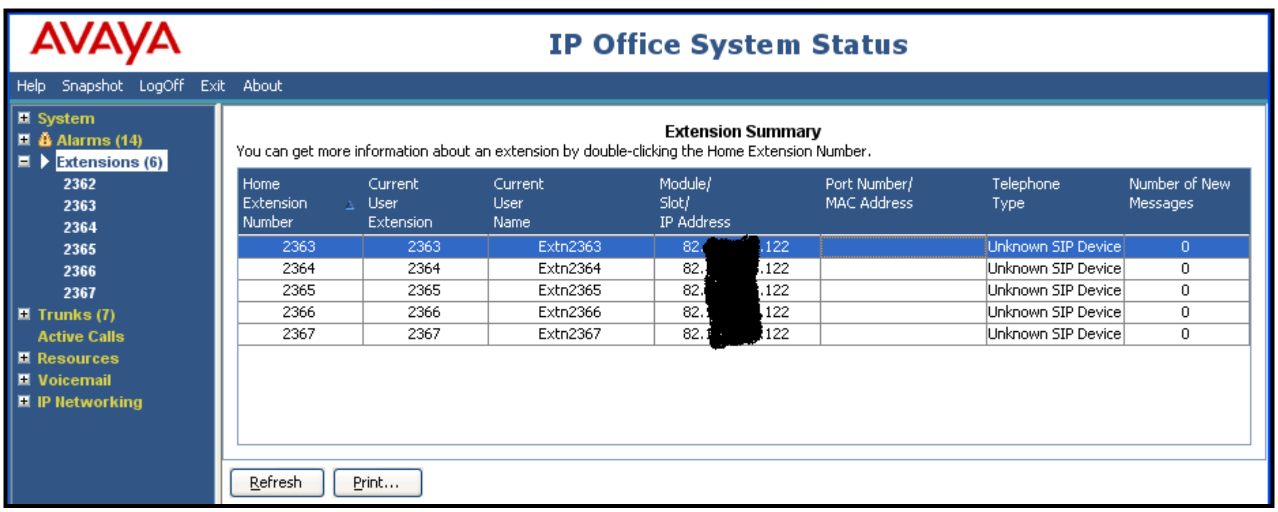

Figure 51: IP Office System Status Extension Summary

After restarting the client, it will connect to MC Controller. The Green status: "Access point ready" indicates the client is successfully registered with the MC Controller.

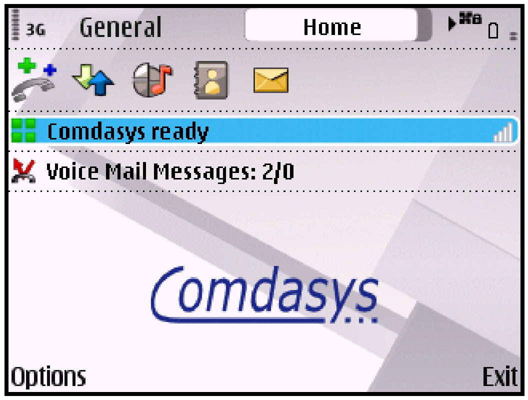

Figure 52: MC Client Screen

9. Conclusion

These Application Notes contain instructions for the configuration of a connection between the IP Office and the Comdasys MC Controller. All test cases passed with exceptions noted in **Section 7**.

10. Additional References

This section references documentation which is relevant to these Application Notes. The Avaya product documentation is available at http://support.avaya.com. Comdasys documentation can be obtained from http://ftp.comdasys.com/pub/documentation/.

[1] *IP Office Installation*, August 2009, Document Number 15-601042.
[2] *IP Office 5.0 Manager,* August 2009, Document Number 15-601011
[3] *Comdasys FMC Mobile Convergence – Administrator Manual*, March 2010, Document Version 1.5
[4] *Comdasys MC Client Symbian Manual, Version 2.0*, October 2009

Several Internet Engineering Task Force (IETF) standards track RFC documents were referenced within these Application Notes. The RFC documents may be obtained at: http://www.rfc-editor.org/rfcsearch.html.

[5] RFC 3261 - *SIP (Session Initiation Protocol)*, June 2002, Proposed Standard

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.